Semi-Supervised Speech Enhancement with Variational Autoencoders

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Outline

- 1. Bayesian and Deep Learning
- 2. Speech Enhancement with Non-negative Matrix Factorization
- 3. Speech Enhancement with Variational Autoencoders
 - Deep Generative Speech Modeling
 - Speech Enhancement
 - Experiments
 - Extensions
- 4. Conclusion

Bayesian and Deep Learning



Ill-posed inverse problem: requires external information.

Bayesian methodology vs deep learning

Bayesian methodology

External information:	Prior	p(latent)
	Likelihood	<i>p</i> (obs. latent)
Problem solving:	Posterior	p(latent obs.)
Advantages:	Flexible, explanatory	
Drawbacks:	Performance, strong hypotheses	

Bayesian methodology

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Discriminative deep learning approach

External information:Training dataProblem solving:Observations neural networkAdvantages:State-of-the-art, fast at test timeDrawbacks:Poorly flexible once trained, poorly explanatory

Bayesian methodology

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Discriminative deep learning approach

External information:Training dataProblem solving:Observations $\frac{neural}{network}$ latent variables of interestAdvantages:State-of-the-art, fast at test timeDrawbacks:Poorly flexible once trained, poorly explanatory

How to exploit the best of both worlds?

Learn the prior directly from data using deep generative models.

Deep-learning-based generative models (with latent variables)



- ▷ Variational autoencoders (Kingma and Welling 2014)
- ▷ Generative adversarial networks (Goodfellow et al. 2014)

Application: semi-supervised speech enhancement



Semi-supervised approach (Smaragdis et al. 2007):

- ◊ Training from clean speech signals only.
- $\diamond\,$ Free of generalization issues regarding the noisy recording environment.

We want the method to be speaker independent.

Speech Enhancement with Non-negative Matrix Factorization

Speech enhancement as a source separation problem



In the short-term Fourier transform (STFT) domain, we observe:

$$x_{fn} = s_{fn} + b_{fn}, \tag{1}$$

- \triangleright $s_{fn} \in \mathbb{C}$ is the clean speech signal.
- \triangleright $b_{fn} \in \mathbb{C}$ is the noise signal.
- ▷ $(f, n) \in \mathbb{B} = \{0, ..., F 1\} \times \{0, ..., N 1\}.$
- \triangleright f is the frequency index and n the time-frame index.

▷ Non-stationary Gaussian model (Pham and Garat 1997; Cardoso 2001) Independently for all $(f, n) \in \mathbb{B}$:

$$s_{fn} \sim \mathcal{N}_c(0, v_{s,fn}) \qquad \perp \qquad b_{fn} \sim \mathcal{N}_c(0, v_{b,fn}).$$
 (2)

Consequently, we also have:

$$x_{fn} \sim \mathcal{N}_c \left(0, v_{s,fn} + v_{b,fn} \right). \tag{3}$$

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- ▷ Spectro-temporal variance modeling (Vincent et al. 2010; Vincent et al. 2014)
 - persistence with structured sparsity penalties;

(Févotte et al. 2006; Kowalski and Torrésani 2009)

- redundancy with non-negative matrix factorization;
 (Benaroya et al. 2003; Févotte et al. 2009; Ozerov et al. 2012)
- ▷ more complex structures with deep neural networks.

(Bando et al. 2018; Leglaive et al. 2018)

Non-negative matrix factorization (NMF) (Lee and Seung 1999)

Low-rank matrix factorization technique with non-negativity constraints.

Learning the parts of objects by non-negative matrix factorization

Daniel D. Lee* & H. Sebastian Seung*†

* Bell Laboratories, Lucent Technologies, Murray Hill, New Jersey 07974, USA † Department of Brain and Cognitive Sciences, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139, USA

Is perception of the whole based on perception of its parts? There is psychological1 and physiological2.3 evidence for parts-based representations in the brain, and certain computational theories of object recognition rely on such representations45. But little is known about how brains or computers might learn the parts of objects. Here we demonstrate an algorithm for non-negative matrix factorization that is able to learn parts of faces and semantic features of text. This is in contrast to other methods, such as principal components analysis and vector quantization, that learn holistic, not parts-based, representations. Non-negative matrix factorization is distinguished from the other methods by its use of non-negativity constraints. These constraints lead to a parts-based representation because they allow only additive, not subtractive, combinations. When non-negative matrix factorization is implemented as a neural network, parts-based representations emerge by virtue of two properties: the firing rates of neurons are never negative and synaptic strengths do not change sign.



Principal component analysis (PCA) is also a low-rank matrix factorization, with different constraints.

NMF for face images (Lee and Seung 1999)

A face can be represented a linear combination of basis images.

with NMF: localized features representing intuitive notions of parts of faces.
 with PCA: eigenfaces.



Reproduced from (Lee and Seung, 1999)

A spectrogram can be represented a linear combination of spectral templates.



> Non-stationary Gaussian model:

$$s_{fn} \sim \mathcal{N}_c(0, v_{s,fn})$$
 and $b_{fn} \sim \mathcal{N}_c(0, v_{b,fn}).$ (4)

> Variance model based on non-negative matrix factorization (NMF):

$$v_{j,fn} = (\mathbf{W}_j \mathbf{H}_j)_{f,n}, \qquad j \in \{s, b\},$$
(5)

▶ **W**_j ∈ ℝ^{F×K_j}₊ is a dictionary matrix of spectral templates;
 ▶ **H**_j ∈ ℝ^{K_j×N}₊ is the activation matrix;

 \triangleright K_j is the rank of the factorization (usually $K_j(F + N) \ll FN$).



Semi-supervised NMF-based speech enhancement

(Smaragdis et al. 2007; Mysore and Smaragdis 2011)

 $\widehat{\mathbf{S}_{fn}} = \mathbb{E}_{p(s_{fn}|x_{fn})}[s_{fn}] = \frac{(\mathsf{W}_s\mathsf{H}_s)_{f,n}}{(\mathsf{W}_s\mathsf{H}_s + \mathsf{W}_b\mathsf{H}_b)_{f,n}} x_{fn}.$ (6)

Semi-supervised NMF-based speech enhancement

(Smaragdis et al. 2007; Mysore and Smaragdis 2011)

Speech enhancement with Wiener filtering

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 (6)

Training: learn W_s from a dataset of clean speech signals

$$\min_{\mathbf{W}_{s}\geq 0}\sum_{(f,n)\in\mathbb{B}}d_{\mathrm{IS}}\Big(|s_{fn}|^{2},\,(\mathbf{W}_{s}\mathbf{H}_{s})_{f,n}\Big),\tag{7}$$

 \triangleright $d_{IS}(\cdot, \cdot)$ is the Itakura-Saito (IS) divergence.

- ▷ equivalent to maximizing the likelihood of $\mathbf{s} = \{s_{fn}\}_{(f,n)\in\mathbb{B}}$ (Févotte et al. 2009).
- ▷ majorize-minimize algorithm (Févotte and Idier 2011).

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Test: estimate $\mathbf{H}_{s}, \mathbf{W}_{b}, \mathbf{H}_{b}$ from the noisy mixture signal $\min_{\mathbf{H}_{s}, \mathbf{W}_{b}, \mathbf{H}_{b} \geq 0} \sum_{(f, n) \in \mathbb{B}} d_{\mathrm{IS}} \left(|x_{fn}|^{2}, \left(\mathbf{W}_{s} \mathbf{H}_{s} + \mathbf{W}_{b} \mathbf{H}_{b} \right)_{f, n} \right).$ (8)

Research problem: from NMF to neural networks



In this work, we explore the use of neural networks in order to overcome the limitations of this variance model.

Speech Enhancement with Variational Autoencoders

Speech Enhancement with Variational Autoencoders

Deep Generative Speech Modeling

Deep generative speech model (Bando et al. 2018)

Independently for all $(f, n) \in \mathbb{B}$,

$$\mathbf{s}_{fn} \mid \mathbf{z}_n \sim \mathcal{N}_c\left(\mathbf{0}, \sigma_f^2(\mathbf{z}_n)\right), \quad \text{with } \mathbf{z}_n \sim \mathcal{N}(\mathbf{0}, \mathbf{I}_L), \quad (9)$$

and $\sigma_f^2 : \mathbb{R}^L \mapsto \mathbb{R}_+$ corresponds to a neural network of parameters θ_s .



How to learn the parameters θ_s of this generative neural network?

Learning the model parameters with variational autoencoders

- ▷ **Training dataset** of STFT speech time frames: $\mathbf{s} = {\{\mathbf{s}_n \in \mathbb{C}^F\}}_{n=0}^{N-1}$.
- ▷ Associated latent variables: $z = {z_n \in \mathbb{R}^L}_{n=0}^{N-1}$.
- $\triangleright \text{ Difficulty: Intractable marginal likelihood } p(\mathbf{s}; \boldsymbol{\theta}_s) = \int p(\mathbf{s} | \mathbf{z}; \boldsymbol{\theta}_s) p(\mathbf{z}) d\mathbf{z}.$
- ▷ **Solution**: Variational autoencoder (VAE) (Kingma and Welling 2014).

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- ▷ Solution: Variational autoencoder (VAE) (Kingma and Welling 2014).

Maximize a lower bound of $\ln p(\mathbf{s}; \theta_s)$, which can be recast as:

$$\min_{\boldsymbol{\theta}_{s}} \sum_{(f,n)\in\mathbb{B}} \mathbb{E}_{q(\mathbf{z}_{n}|\mathbf{s}_{n};\boldsymbol{\phi})} \Big[d_{lS} \left(|s_{fn}|^{2}; \sigma_{f}^{2}(\mathbf{z}_{n}) \right) \Big],$$
(10)

where $q(\mathbf{z}_n|\mathbf{s}_n; \phi)$ is an approximation of the intractable posterior $p(\mathbf{z}_n|\mathbf{s}_n; \theta_s)$ and is defined by an "encoder network" of parameters ϕ .

The dependency of $\sigma_f^2(\cdot)$ on θ_s is not made explicit to avoid cluttered notations.

For any variational distribution $q(\mathbf{z}|\mathbf{s}; \phi)$, we have:

$$\ln p(\mathbf{s}; \boldsymbol{\theta}_s) = \mathcal{L}(\boldsymbol{\phi}, \boldsymbol{\theta}_s) + D_{\mathsf{KL}} \big(\boldsymbol{q}(\mathbf{z}|\mathbf{s}; \boldsymbol{\phi}) \parallel p(\mathbf{z}|\mathbf{s}; \boldsymbol{\theta}_s) \big), \tag{11}$$

where $D_{\mathsf{KL}}(q \parallel p) = \mathbb{E}_q[\ln q - \ln p] \ge 0.$



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Problem #1	
$\max_{oldsymbol{ heta}_s} \mathcal{L}(\phi, oldsymbol{ heta}_s)$	
where $\mathcal{L}(\phi, oldsymbol{ heta}_s) \leq \ln p(\mathbf{s}; oldsymbol{ heta}_s).$	

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To define the objective function, we need to define $q(\mathbf{z}|\mathbf{s}; \boldsymbol{\phi})$.

The encoder network

 $q(\mathbf{z}|\mathbf{s}; \phi)$ is defined independently for all time frames $n \in \{0, ..., N-1\}$ and all latent dimensions $l \in \{0, ..., L-1\}$ by:

$$(\mathbf{z}_n)_I \mid \mathbf{s}_n \sim \mathcal{N}\Big(\tilde{\mu}_I(\mathbf{s}_n), \tilde{\sigma}_I^2(\mathbf{s}_n) \Big),$$
(13)

where $\tilde{\mu}_I : \mathbb{C}^F \mapsto \mathbb{R}$ and $\tilde{\sigma}_I^2 : \mathbb{C}^F \mapsto \mathbb{R}_+$ correspond to a neural network of parameters ϕ .



Variational free energy: full expression

$$\mathcal{L}(\boldsymbol{\theta}_{s},\boldsymbol{\phi}) \stackrel{c}{=} -\sum_{f=0}^{F-1} \sum_{n=0}^{N-1} \mathbb{E}_{\boldsymbol{q}(\boldsymbol{z}_{n}|\boldsymbol{s}_{n};\boldsymbol{\phi})} \left[\boldsymbol{d}_{\mathsf{lS}} \left(|\boldsymbol{s}_{fn}|^{2} ; \sigma_{f}^{2}(\boldsymbol{z}_{n}) \right) \right] \\ + \frac{1}{2} \sum_{l=1}^{L} \sum_{n=0}^{N-1} \left[\ln \tilde{\sigma}_{l}^{2}(\boldsymbol{s}_{n}) - \tilde{\mu}_{l}^{2}(\boldsymbol{s}_{n}) - \tilde{\sigma}_{l}^{2}(\boldsymbol{s}_{n}) \right].$$
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(14)

▷ Intractable expectation replaced by a sample average:

$$\mathbb{E}_{q(\mathbf{z}_{n}|\mathbf{s}_{n};\boldsymbol{\phi})}\left[d_{\mathsf{IS}}\left(\left|s_{fn}\right|^{2};\sigma_{f}^{2}(\mathbf{z}_{n})\right)\right] \approx \frac{1}{R}\sum_{r=1}^{R}\left[d_{\mathsf{IS}}\left(\left|s_{fn}\right|^{2};\sigma_{f}^{2}\left(\tilde{\mathbf{z}}_{n}^{(r)}\right)\right)\right],\quad(15)$$

where $\{\tilde{\mathbf{z}}_{n}^{(r)}\}_{r=1}^{R}$ are i.i.d. realizations drawn¹ from $q(\mathbf{z}_{n}|\mathbf{s}_{n};\phi)$.

¹using the so-called "reparametrization trick" (Kingma and Welling 2014).

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 \triangleright In practice R = 1:

$$\mathbb{E}_{q(\mathbf{z}_{n}|\mathbf{s}_{n};\boldsymbol{\phi})}\left[d_{\mathsf{IS}}\left(\left|s_{fn}\right|^{2};\sigma_{f}^{2}(\mathbf{z}_{n})\right)\right]\approx d_{\mathsf{IS}}\left(\left|s_{fn}\right|^{2};\sigma_{f}^{2}\left(\tilde{\mathbf{z}}_{n}\right)\right).$$
(16)

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Training procedure step by step (0)

$$\mathcal{L}(\boldsymbol{\theta}_{s},\boldsymbol{\phi}) \stackrel{c}{=} -\sum_{f=0}^{F-1} \sum_{n=0}^{N-1} \left[d_{\mathsf{IS}}\left(\left| \boldsymbol{s}_{fn} \right|^{2}; \sigma_{f}^{2}\left(\tilde{\boldsymbol{z}}_{n} \right) \right) \right] + \frac{1}{2} \sum_{l=1}^{L} \sum_{n=0}^{N-1} \left[\ln \tilde{\sigma}_{l}^{2}\left(\boldsymbol{s}_{n} \right) - \tilde{\mu}_{l}^{2}\left(\boldsymbol{s}_{n} \right) - \tilde{\sigma}_{l}^{2}\left(\boldsymbol{s}_{n} \right) \right].$$
(17)



Training procedure step by step (1)

$$\mathcal{L}\left(\boldsymbol{\theta}_{s},\boldsymbol{\phi}\right) \stackrel{c}{=} -\sum_{f=0}^{F-1} \sum_{n=0}^{N-1} \left[d_{\mathsf{IS}}\left(\left| \boldsymbol{s}_{fn} \right|^{2}; \sigma_{f}^{2}\left(\tilde{\boldsymbol{z}}_{n} \right) \right) \right] \\ + \frac{1}{2} \sum_{l=1}^{L} \sum_{n=0}^{N-1} \left[\ln \tilde{\sigma}_{l}^{2}\left(\boldsymbol{s}_{n} \right) - \tilde{\mu}_{l}^{2}\left(\boldsymbol{s}_{n} \right) - \tilde{\sigma}_{l}^{2}\left(\boldsymbol{s}_{n} \right) \right].$$
(18)



Training procedure step by step (2)

$$\mathcal{L}(\boldsymbol{\theta}_{s},\boldsymbol{\phi}) \stackrel{c}{=} -\sum_{f=0}^{F-1} \sum_{n=0}^{N-1} \left[d_{lS} \left(|\boldsymbol{s}_{fn}|^{2}; \sigma_{f}^{2}(\tilde{\mathbf{z}}_{n}) \right) \right] \\ + \frac{1}{2} \sum_{l=1}^{L} \sum_{n=0}^{N-1} \left[\ln \tilde{\sigma}_{l}^{2}(\mathbf{s}_{n}) - \tilde{\mu}_{l}^{2}(\mathbf{s}_{n}) - \tilde{\sigma}_{l}^{2}(\mathbf{s}_{n}) \right].$$
(19)

Training procedure step by step (3)

frequency (kHz) 4

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time (s)

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Training procedure step by step (4)

$$\mathcal{L}(\boldsymbol{\theta}_{s},\boldsymbol{\phi}) \stackrel{c}{=} -\sum_{f=0}^{F-1} \sum_{n=0}^{N-1} \left[d_{lS} \left(|\boldsymbol{s}_{fn}|^{2} ; \sigma_{f}^{2} (\tilde{\mathbf{z}}_{n}) \right) \right] \\ + \frac{1}{2} \sum_{l=1}^{L} \sum_{n=0}^{N-1} \left[\ln \tilde{\sigma}_{l}^{2} (\mathbf{s}_{n}) - \tilde{\mu}_{l}^{2} (\mathbf{s}_{n}) - \tilde{\sigma}_{l}^{2} (\mathbf{s}_{n}) \right].$$
(21)

Iterative optimization with a gradient-ascent-based algorithm.

Summary

NMF-based model

 $v_{s,fn} = (\mathbf{W}_s)_{f,:}^\top \times (\mathbf{H}_s)_{:,n}$

- ▷ linear function of $(\mathbf{H}_s)_{:,n} \in \mathbb{R}_+^{K_s}$.
- $\triangleright \#$ trainable parameters = $F \times K_s$.
- ▷ IS divergence minimization.
- ▷ Interpretability.



VAE-based model

$$v_{s,fn} = \sigma_f^2(\mathbf{z}_n)$$

- ▷ non-linear function of $\mathbf{z}_n \in \mathbb{R}^L$.
- \triangleright # trainable parameters is free.
- ▷ IS divergence minimization.
- Lack of (direct) interpretability.



Speech Enhancement with Variational Autoencoders

Speech Enhancement

Models for semi-supervised speech enhancement

Supervised speech model

$$\mathbf{s}_{fn} \mid \mathbf{z}_n \sim \mathcal{N}_c\left(\mathbf{0}, \sigma_f^2(\mathbf{z}_n)\right), \qquad \mathbf{z}_n \sim \mathcal{N}(\mathbf{0}, \mathbf{I}),$$
 (22)

where $\sigma_f^2(\cdot)$ corresponds to the decoder network of parameters θ_s .

Unsupervised noise model

$$b_{fn} \sim \mathcal{N}_c \left(0, (\mathbf{W}_b \mathbf{H}_b)_{f,n} \right),$$
 (23)

where $\mathbf{W}_b \in \mathbb{R}_+^{F \times K_b}$ and $\mathbf{H}_b \in \mathbb{R}_+^{K_b \times N}$.

Likelihood

$$x_{fn} \mid \mathbf{z}_n \sim \mathcal{N}_c \left(\mathbf{0}, \sigma_f^2(\mathbf{z}_n) + (\mathbf{W}_b \mathbf{H}_b)_{f,n} \right).$$
(24)

S. L., L. Girin, R. Horaud "A variance modeling framework based on variational autoencoders for speech enhancement", IEEE MLSP, 2018.

Semi-supervised VAE-based speech enhancement

Speech enhancement with Wiener-like filtering $\hat{s}_{fn} = \mathbb{E}_{\rho(s_{fn}|x_{fn};\theta)}[s_{fn}] = \mathbb{E}_{\rho(\mathbf{z}_n|\mathbf{x}_n;\theta)} \left[\frac{\sigma_f^2(\mathbf{z}_n)}{\sigma_f^2(\mathbf{z}_n) + (\mathbf{W}_b\mathbf{H}_b)_{f,n}} \right] x_{fn}, \quad (25)$ where the expectation is intractable: Markov chain Monte Carlo (MCMC).

27

Semi-supervised VAE-based speech enhancement

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$$\hat{s}_{fn} = \mathbb{E}_{\rho(s_{fn}|x_{fn};\theta)}[s_{fn}] = \mathbb{E}_{\rho(\mathbf{z}_n|\mathbf{x}_n;\theta)} \left[\frac{\sigma_f^2(\mathbf{z}_n)}{\sigma_f^2(\mathbf{z}_n) + (\mathbf{W}_b\mathbf{H}_b)_{f,n}} \right] x_{fn}, \quad (25)$$

where the expectation is intractable: Markov chain Monte Carlo (MCMC).

— **Training**: learn $\sigma_f^2(\cdot)$ from a dataset of clean speech signals — Introduce an encoder network and maximize a lower bound of $p(\mathbf{s}; \boldsymbol{\theta}_s)$.

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where the expectation is intractable: Markov chain Monte Carlo (MCMC).

— **Training**: learn $\sigma_f^2(\cdot)$ from a dataset of clean speech signals — Introduce an encoder network and maximize a lower bound of $p(\mathbf{s}; \boldsymbol{\theta}_s)$.

Test: estimate \mathbf{W}_b , \mathbf{H}_b from the noisy mixture signal We would like to maximize w.r.t $\mathbf{W}_b \in \mathbb{R}^{F \times K_b}_+, \mathbf{H}_b \in \mathbb{R}^{K_b \times N}_+$:

$$p(\mathbf{x};\boldsymbol{\theta}) = \int p(\mathbf{x}|\mathbf{z};\boldsymbol{\theta})p(\mathbf{z})d\mathbf{z}.$$
 (26)

We develop a Monte Carlo EM algorithm (see paper for further details).

S. L., L. Girin, R. Horaud "A variance modeling framework based on variational autoencoders for speech enhancement", IEEE MLSP, 2018.

Speech Enhancement with Variational Autoencoders

Experiments

- ▷ Clean speech signals: TIMIT database (Garofolo et al. 1993).
- ▷ Noise signals: DEMAND database (domestic environment, nature, office, indoor public spaces, street and transportation).

▷ **Training**:

- ▷ training set of TIMIT database;
- $\triangleright~\sim$ 4 hours of speech;
- ▷ 462 speakers.

▷ **Test**:

- ▷ 168 noisy mixtures at 0 dB signal-to-noise ratio;
- ▷ Different speakers and sentences than in the training set.

- 1. Semi-supervised NMF baseline.
- 2. Fully-supervised deep-learning-based method (Xu et al. 2015):
 - ▷ Deep neural network for mapping noisy speech log-power spectrograms to clean speech log-power spectrograms.
 - ▷ From (Xu et al. 2015):

"to improve the generalization capability we include more than 100 different noise types in designing the training set"

▷ Here, we use different noise datasets for training and testing (with overlapping noise types).

Experimental results

Median value indicated above each boxplot.



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Median value indicated above each boxplot.





Song: "Sunrise" by Shannon Hurley, from the MGT Music Audio Signal Separation (MASS) dataset.



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Speech Enhancement with Variational Autoencoders

Extensions

Alpha-stable noise model

Example noise signal recorded within an accelerating subway.

▷ Gaussian NMF-based noise model:



▷ Alpha-stable noise model:



S. L., U. Şimşekli, A. Liutkus, L. Girin, R. Horaud "Speech enhancement with variational autoencoders and alpha-stable distributions", IEEE ICASSP, 2019.

Multi-microphone recording setup

- A fully-supervised model would need to be retrained. We might even need to collect new data.
- Our semi-supervised approach can be easily adapted to this new configuration.



Multichannel speech modelLet $\mathbf{s}_{fn} \in \mathbb{C}^I$ be the multichannel speech signal, we have: $\mathbf{s}_{fn} \mid \mathbf{z}_n \sim \mathcal{N}_c \left(\mathbf{0}, \sigma_f^2(\mathbf{z}_n) \times \mathbf{R}_{\mathbf{s}, f} \right), \quad \mathbf{z}_n \sim \mathcal{N}(\mathbf{0}, \mathbf{I}),$ (27) $\sigma_f^2(\cdot)$ is learned during the training stage.> $\mathbf{R}_{\mathbf{s}, f}$ is the spatial covariance matrix and is estimated at test time.

S. L., L. Girin, R. Horaud "Semi-supervised multichannel speech enhancement with variational autoencoders and non-negative matrix factorization", IEEE ICASSP, 2019.



Song: "Ana" by Vieux Farka Toure, from the MGT Music Audio Signal Separation (MASS) dataset.



Song: "Ana" by Vieux Farka Toure, from the MGT Music Audio Signal Separation (MASS) dataset.



Song: "Ana" by Vieux Farka Toure, from the MGT Music Audio Signal Separation (MASS) dataset.



Song: "Ana" by Vieux Farka Toure, from the MGT Music Audio Signal Separation (MASS) dataset.

The speech generative process is conditioned on visual information of the lip region, which is invariant to the acoustic noise.



Mostafa Sadeghi, S. L., Xavier Alameda-Pineda, Laurent Girin, Radu Horaud "Audio-visual speech enhancement using conditional variational auto-encoder", submitted to IEEE Transactions on Audio, Speech and Language Processing, 2019.

Generate a sequence of speech STFT time frames from a sequence of latent vectors.







(c) bidirectional recurrent NN

Recurrent models induce a temporal dynamic over the reconstructed speech, with Wiener filtering.

S. L., X. Alameda-Pineda, L. Girin, R. Horaud "A recurrent variational autoencoder for speech enhancement ", submitted to IEEE ICASSP, 2020.

Conclusion

We combined the learning capabilities of neural networks with the flexibility of probabilistic models for speech enhancement.

- Variational autoencoders are more expressive than NMF models due to their non-linear nature and due to the freedom in the number of trainable parameters.
- Semi-supervised approaches are flexible and can easily adapt to different situations at test time, in terms of noise and number of microphones.

Some challenges that we would like to address:

- ▷ to account for phase information;
- ▷ to develop deep generative spatial models of multi-microphone signals;
- b to encode multi-level and multi-time-scale properties of speech signals in the deep generative process;
- \triangleright to develop more efficient statistical inference algorithms.

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Thank you for your attention

Audio examples and code: https://sleglaive.github.io

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