A Recurrent Variational Autoencoder for Speech Enhancement

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2020 IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP)











Introduction

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 also want the method to be speaker independent.

Speech enhancement as a source separation problem



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

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

- ▷ $s_{fn} \in \mathbb{C}$ 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.

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):

structured sparsity penalties

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

▷ non-negative matrix factorization (NMF)

(Benaroya et al. 2003; Févotte et al. 2009; Ozerov et al. 2012)

▷ deep generative neural networks

(Bando et al. 2018)

Deep generative speech model for speech enhancement

It was recently proposed to model the speech variance by a generative neural network (variational autoencoder) (Bando et al. 2018).



- single-microphone semi-supervised speech enhancement (Bando et al. 2018; Leglaive et al. 2018; Leglaive et al. 2019b; Pariente et al. 2019).
- multi-microphone semi-supervised speech enhancement (Sekiguchi et al. 2018; Leglaive et al. 2019a; Fontaine et al. 2019; Sekiguchi et al. 2019).

Previous works only considered a feed-forward and fully-connected generative neural network, thus neglecting speech temporal dynamic.

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In this work,

- $\triangleright\,$ we propose a recurrent VAE speech model trained on clean speech signals;
- ▷ at test time, it is combined with an NMF noise model;
- we derive a variational expectation-maximization algorithm where the pre-trained encoder of the VAE is fine-tuned from the noisy mixture signal;
- ▷ experiments show that the temporal dynamic induced over the estimated speech signal improves the speech enhancement performance.

Deep generative speech model

▷ $\mathbf{s} = \mathbf{s}_{0:N-1} = {\mathbf{s}_n \in \mathbb{C}^F}_{n=0}^{N-1}$ is a sequence of *N* STFT speech time frames. ▷ $\mathbf{z} = \mathbf{z}_{0:N-1} = {\mathbf{z}_n \in \mathbb{R}^L}_{n=0}^{N-1}$ is a corresponding sequence of *N* latent random vectors. ▷ s = s_{0:N-1} = {s_n ∈ C^F}^{N-1}_{n=0} is a sequence of N STFT speech time frames.
 ▷ z = z_{0:N-1} = {z_n ∈ ℝ^L}^{N-1}_{n=0} is a corresponding sequence of N latent random vectors.

Deep generative speech model

Independently for all time frames, in its most general form, we have:

$$\mathbf{s}_{n} \mid \mathbf{z} \sim \mathcal{N}_{c}\left(\mathbf{0}, \operatorname{diag}\left\{\mathbf{v}_{\mathbf{s},n}(\mathbf{z})\right\}\right), \quad \text{with} \quad \mathbf{z}_{n} \stackrel{\mathrm{i.i.d}}{\sim} \mathcal{N}\left(\mathbf{0}, \mathbf{I}\right),$$
 (4)

and where $\mathbf{v}_{s,n}(\mathbf{z})$ is provided by a decoder/generative neural network.

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Deep generative speech model Independently for all time frames, in its most general form, we have: $\mathbf{s}_n \mid \mathbf{z} \sim \mathcal{N}_c \left(\mathbf{0}, \text{diag} \left\{ \mathbf{v}_{\mathbf{s},n}(\mathbf{z}) \right\} \right), \text{ with } \mathbf{z}_n \overset{\text{i.i.d}}{\sim} \mathcal{N} \left(\mathbf{0}, \mathbf{I} \right),$ (4)

and where $\mathbf{v}_{s,n}(\mathbf{z})$ is provided by a decoder/generative neural network.

Multiple choices can be made to define this neural network, leading to different probabilistic graphical models.

Feed-forward fully-connected neural network (FFNN)

Variance model

Probabilistic graphical model



The speech STFT time frames are not only conditionally independent, but also marginally independent: $p(\mathbf{s}; \theta_{dec}) = \prod_{n=0}^{N-1} p(\mathbf{s}_n; \theta_{dec}).$

Recurrent neural network (RNN)

Variance model

Probabilistic graphical model



The speech STFT time frames are not marginally independent anymore.

Variance model

Probabilistic graphical model



The speech STFT time frames are not marginally independent anymore.



Training dataset

 $\{\mathbf{s}^{(i)} \in \mathbb{C}^{F \times N}\}_{i=1}^{I}$: i.i.d sequences of N STFT speech time frames.

Maximum marginal likelihood

 $\max_{\boldsymbol{\theta}_{\mathsf{dec}}} \ln p(\mathbf{s}; \boldsymbol{\theta}_{\mathsf{dec}})$

Training dataset

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Intractability issue

$$p(\mathbf{s}; oldsymbol{ heta}_{\mathsf{dec}}) = \int p(\mathbf{s}, \mathbf{z}; oldsymbol{ heta}_{\mathsf{dec}}) d\mathbf{z}$$

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Solution

Variational inference (Jordan et al. 1999) + neural networks

= variational autoencoder (VAE) (Kingma and Welling 2014)

Variational lower bound



Variational lower bound



$$\label{eq:problem #1} \begin{tabular}{c} \hline & \mathsf{Problem \ \#1} \\ & & \\ &$$





 $q(\mathbf{z}|\mathbf{s}; \theta_{enc})$ is an approximation of the intractable posterior $p(\mathbf{z}|\mathbf{s}; \theta_{dec})$, and it is defined by an encoder/recognition network (Kingma and Welling 2014).

Looking at posterior dependencies



Looking at posterior dependencies



Looking at posterior dependencies



Inference model and encoder network



Inference model and encoder network



With this inference model defined, the variational lower bound is completely specified and it can be optimized using gradient-ascent based algorithms. We used around 25 hours of clean speech data, from the Wall Street Journal (WSJ0) dataset.

Semi-supervised speech enhancement

Models for semi-supervised speech enhancement

Pre-trained deep generative speech model

$$s_{fn} \mid \mathbf{z} \sim \mathcal{N}_{c}(\mathbf{0}, v_{s, fn}(\mathbf{z})), \qquad \mathbf{z}_{n} \stackrel{\text{i.i.d}}{\sim} \mathcal{N}(\mathbf{0}, \mathbf{I}),$$
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where $v_{s,fn}$ is the decoder neural network (FFNN, RNN or BRNN) whose parameters were learned during the training phase.

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Pre-trained deep generative speech model

NMF-based noise model

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

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

(7)

Models for semi-supervised speech enhancement

Pre-trained deep generative speech model –

$$s_{fn} \mid \mathbf{z} \sim \mathcal{N}_{c}\left(0, v_{s, fn}(\mathbf{z})\right), \qquad \mathbf{z}_{n} \stackrel{\text{i.i.d}}{\sim} \mathcal{N}(\mathbf{0}, \mathbf{I}),$$
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Likelihood

$$x_{fn} \mid \mathbf{z} \sim \mathcal{N}_c \left(0, g_n v_{s, fn}(\mathbf{z}) + (\mathbf{W}_b \mathbf{H}_b)_{f, n} \right), \tag{8}$$

where $g_n \in \mathbb{R}_+$ is a gain parameter (Leglaive et al. 2018).

Speech estimation with Wiener-like filtering



Speech estimation with Wiener-like filtering

$\hat{s}_{fn} = \mathbb{E}_{p(s_{fn}|\times_{fn};\phi)}[s_{fn}] = \mathbb{E}_{p(\mathbf{z}|\mathbf{x};\phi)} \left[\frac{\sqrt{g_n} v_{s,fn}(\mathbf{z})}{g_n v_{s,fn}(\mathbf{z}) + (\mathbf{W}_b \mathbf{H}_b)_{f,n}} \right] x_{fn}.$ (9)

Two problems:

1. We need to estimate the remaining unknown model parameters:

$$\phi = \{g_0, ..., g_{N-1}, \mathbf{W}_b, \mathbf{H}_b\},\$$

but the marginal likelihood $p(\mathbf{x}; \phi)$ is intractable.

2. We need to find an approximation to the intractable posterior $p(\mathbf{z}|\mathbf{x}; \phi)$.

Variational lower bound at test time

$$\mathcal{L}_{\mathbf{x}}(\boldsymbol{\theta}_{\mathsf{enc}}, \boldsymbol{\phi}) = \mathbb{E}_{q(\mathbf{z}|\mathbf{x}; \boldsymbol{\theta}_{\mathsf{enc}})} \left[\ln p(\mathbf{x}|\mathbf{z}; \boldsymbol{\phi}) \right] - D_{\mathsf{KL}} \left(q(\mathbf{z}|\mathbf{x}; \boldsymbol{\theta}_{\mathsf{enc}}) \parallel p(\mathbf{z}) \right), \quad (10)$$

where $q(\mathbf{z}|\mathbf{x}; \theta_{enc})$ corresponds to the pre-trained inference model from the training phase, but now the encoder takes noisy speech as input.

Alternate maximization with respect to θ_{enc} (E-Step) and ϕ (M-Step).

Temporal dynamic

$$\hat{s}_{fn} = \mathbb{E}_{\rho(\mathbf{z}|\mathbf{x};\phi)} \left[\frac{\sqrt{g_n} v_{s,fn}(\mathbf{z})}{g_n v_{s,fn}(\mathbf{z}) + (\mathbf{W}_b \mathbf{H}_b)_{f,n}} \right] x_{fn}$$

$$\approx \mathbb{E}_{q(\mathbf{z}|\mathbf{x};\theta_{enc})} \left[\frac{\sqrt{g_n} v_{s,fn}(\mathbf{z})}{g_n v_{s,fn}(\mathbf{z}) + (\mathbf{W}_b \mathbf{H}_b)_{f,n}} \right] x_{fn}. \tag{11}$$

The expectation is intractable, so it is approximated by an empirical average using samples drawn from:

$$q(\mathbf{z}|\mathbf{x};\boldsymbol{\theta}_{enc}) = \prod_{n=0}^{N-1} q(\mathbf{z}_n|\mathbf{z}_{0:n-1},\mathbf{x};\boldsymbol{\theta}_{enc}).$$
(12)

For the RNN and BRNN generative models, this sampling is done recursively. There is a temporal dynamic that is propagated from the latent vectors to the estimated speech signal, through the expectation in (11).

Monte Carlo E-Step

For the FFNN generative model only, a Markov chain Monte Carlo method was used to sample from the intractable posterior $p(\mathbf{z}|\mathbf{x}; \phi)$ (Bando et al. 2018; Leglaive et al. 2018).

"Point-estimate" E-Step

In (Kameoka et al. 2019), it was proposed to only rely on a "point estimate" of the latent variables, based on the maximum a posteriori (MAP):

$$\mathbf{z}^{\star} = \arg \max_{\mathbf{z}} \{ p(\mathbf{z}|\mathbf{x}; \phi) \propto p(\mathbf{x}|\mathbf{z}; \phi) p(\mathbf{z}) \},$$

which can be obtained with gradient-based optimization techniques.

Experiments

Experimental setting

Dataset:

- ▷ About 1.5 hours of noisy speech @ 16 kHz using the WSJ0 (unseen speakers) and QUT-NOISE datasets.
- ▷ Noise types: {"café", "home", "street", "car"}.
- $\triangleright\,$ Signal-to-noise ratios (SNRs): {-5, 0, 5} dB.

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Performance measures (higher is better):

- ▷ scale-invariant signal-to-distortion ratio (SI-SDR) in dB
- ▷ perceptual evaluation of speech quality (PESQ) (between -0.5 and 4.5)
- ▷ extended short-time objective intelligibility (ESTOI) (between 0 and 1)

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Methods:

- ▷ Monte Carlo EM
 ▷ "Point-estimate" EM
 ▶ PEEM {FFNN, RNN, BRNN}
- Proposed variational EM

VEM - {FFNN, RNN, BRNN}

Results



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- \triangleright The RNN model outperforms the FFNN model.
- $\triangleright~$ The VEM algorithm outperforms the PEEM algorithm.

Results



- ▷ For the FFNN generative model, the MCEM algorithm gives the best results.
- > The RNN model outperforms the FFNN model.
- ▷ The VEM algorithm outperforms the PEEM algorithm.
- > The BRNN model does not perform significantly better than the RNN model.

- ▷ We combined a recurrent VAE with an NMF noise model for semi-supervised speech enhancement.
- ▷ The inference model (encoder network) should be carefully designed in order to preserve posterior temporal dependencies between the latent variables.
- ▷ The temporal dynamic induced over the estimated speech signal is beneficial in terms of speech enhancement results.

Audio examples and code:

https://sleglaive.github.io/demo-icassp2020.html

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