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Single-Microphone Speech Enhancement and Separation Using Deep Learning

Kolbæk, Morten

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Single-Microphone Speech Enhancement and Separation Using Deep Learning

November 30, 2018

Morten Kolbæk

PhD Fellow Department of Electronic Systems Aalborg University Denmark





Supervisors: Prof. Jesper Jensen, AAU, Oticon Prof. Zheng-Hua Tan, AAU

Stay Abroad: Dr. Dong Yu, Tencent AI Lab / Microsoft Research

Oticon Fonden







Agenda



Introduction:

- Cocktail Party Problem
- Speech Enhancement and Separation
- Deep Learning

Scientific Contributions:

- Generalization of Deep Learning based Speech Enhancement
 - Human Receivers Speech Intelligibility
 - Machine Receivers Speaker Verification
- On STOI Optimal Deep Learning based Speech Enhancement
- Permutation Invariant Training for Deep Learning based Speech Separation
- Summary and Conclusion



Part I

Introduction

The Cocktail Party Problem



- Speech Enhancement and Separation
- Deep Learning

The Cocktail Party Problem



How do we recognize what one person is saying when others are speaking at the same time (the "**cocktail party problem**")? On what logical basis could one **design a machine** ("filter") for carrying out such an operation?

– Colin Cherry, 1953.



















The Cocktail Party Problem



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The Cocktail Party Problem



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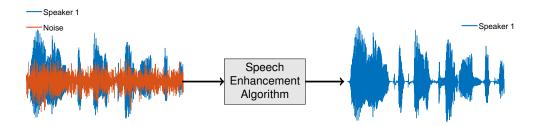
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Speech Enhancement and Separation

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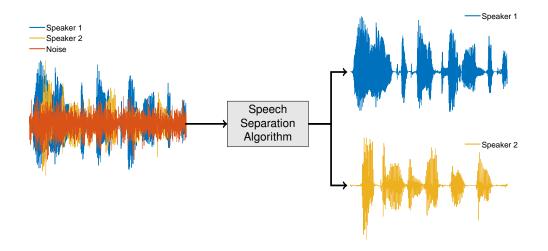
- Cocktail Party Problem
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- Deep Learning

Single-Microphone Speech Enhancement First Task of the Thesis



NEW

Single-Microphone Speech Separation



NEW

Speech Enhancement and Separation

THOME WORDLING

Why Is Solving the Cocktail Party Problem Important?

Human Receivers

- Potential: Hundreds of millions of people worldwide have a hearing loss.
- Challenge: Hearing impaired often struggle in "cocktail party" situations.
- Solution: Algorithms that can enhance the speech signal of interest.
- Application: Hearing Assistive Devices e.g. hearing aids or cochlear implants.

Machine Receivers

- Potential: Millions of people vocally interact with smartphones.
- Challenge: These devices operate in complex acoustic environments.
- Solution: Noise-robust human-machine interface.
- Application: Social robots or digital assistants e.g. Google Asst., Siri, etc.

Speech Enhancement and Separation



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Whats new? - A paradigm shift!

Classical Paradigm

- Derive the solution using specific mathematical models that approximate speech and noise.
- Simplifying assumptions for mathematical tractability.
- Generally not data-driven.
- Good performance when assumptions are valid (sometimes they are not).

Deep Learning Paradigm

Learn the solution using general mathematical models that have "observed" speech and noise.

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- No explicit assumptions.
- Data-driven.
- State-of-the-art performance given enough data and computational resources.

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Deep Learning



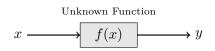
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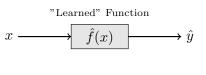
Deep Learning What is it?



• Deep Learning: Subfield of Machine Learning.

Machine Learning: Use data to "learn" or approximate unknown functions f(x) that can be used to make predictions.



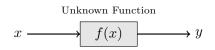


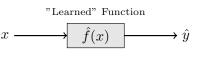
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Deep Learning What is it? – Classical Regression Example

Estimate Happiness from income

- Hypothesis: Happiness is associated with income.
- Data: Perceived happiness and income from people.
- Candidate Models:

■ 7-params. (Big Capacity) $\hat{f}_1(x) = ax^6 + bx^5 + cx^4$ $+ dx^3 + ex^2 + fx + g$

4-params. (Small Capacity) $\hat{f}_2(x) = ax^3 + bx^2 + cx + d$

• **Goal:** Find parameters of $\hat{f}_1(x)$ and $\hat{f}_2(x)$ that best explain the observations.

Subjective Happiness Scale



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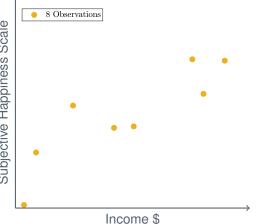
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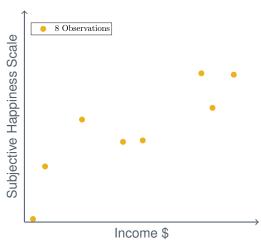
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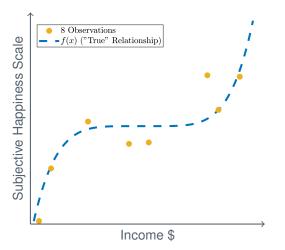


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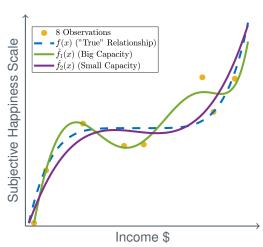
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T-params. (Big Capacity) $\hat{f}_1(x) = -0.2x^6 + 2.5x^5 - 8.1x^4 + 10.3x^3 - 5.4x^2 + 1.2x + 0.3$

■ 4-params. (Small Capacity) $\hat{f}_2(x) = -22.2x^3 + 2.6x^2 + 3.8x - 0.6$

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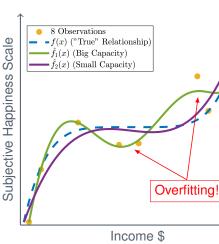
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Scale



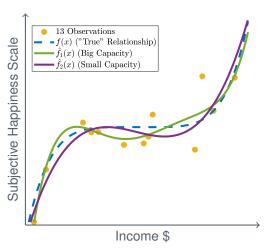
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T-params. (Big Capacity) $\hat{f}_1(x) = 1.1x^6 - 6.5x^5 + 15.1x^4$ $- 18.0x^3 + 11.0x^2 - 2.7x + 0.6$

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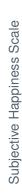
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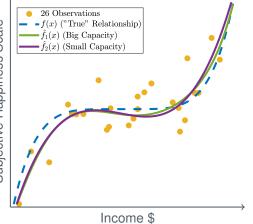
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 \blacksquare 4-params. (Small Capacity) $\hat{f}_2(x) = -9.2x^3 + 2.9x^2 + 1.1x - \textbf{0.2}$

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Estimate Happiness from income

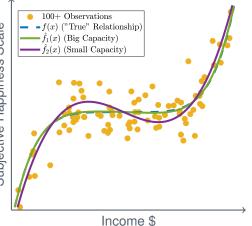
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Scale Subjective Happiness





Estimate Happiness from income

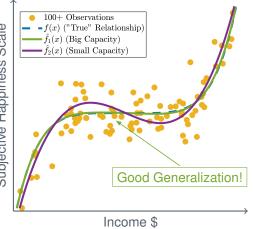
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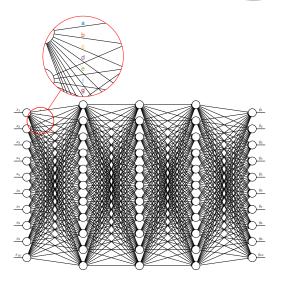


Deep Learning

"Regression" using Deep Neural Networks.

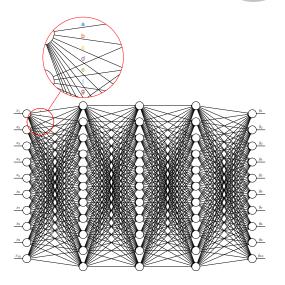
Deep Neural Network

- Non-linear function with potentially MANY (millions) parameters.
- If big enough, they can approximate any function.
- With enough data, they can learn complex mappings.



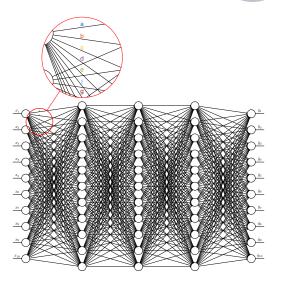
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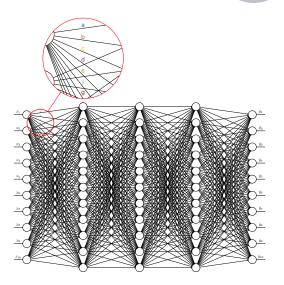
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Google facebook.

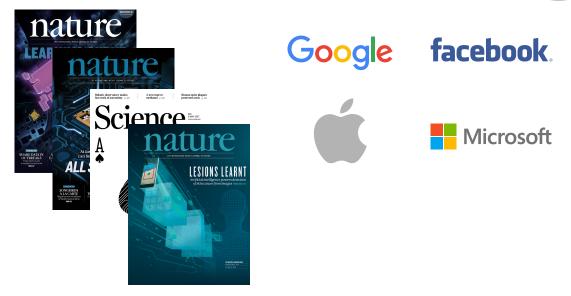






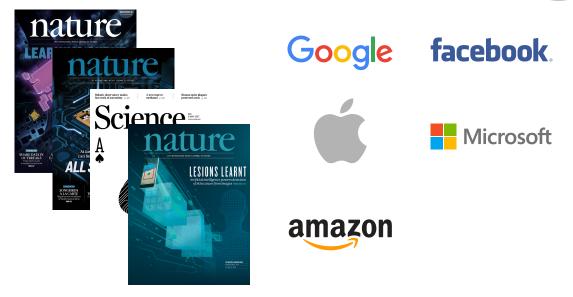






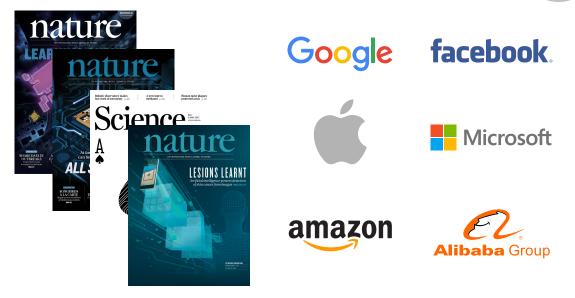












Deep Learning What Can It Do?







Part II

Scientific Contributions

Generalization of DNN based Speech Enhancement Human Receivers - Speech Intelligibility

Generalization of Deep Learning based Speech Enhancement

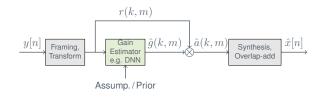
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Promising Results

- Recent studies show that speech enhancement algorithms based on deep learning outperform classical techniques.
- DNNs trained and tested in "narrow" conditions.

Research Gap

Unknown how these algorithms perform in general "broader" conditions and in conditions with a mismatch between training and test.



- y[n]: Noisy speech (time-domain)
- r(k,m): Noisy speech (transform-domain)
- $\hat{g}(k,m)$: Estimated gain
- $\hat{a}(k,m)$: Enhanced speech (transform-domain)
 - $\hat{x}[n]$: Enhanced speech (time-domain)



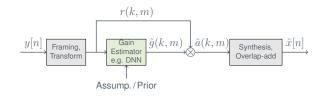
Generalization of DNN based Speech Enhancement Human Receivers - Motivation and Research Gap

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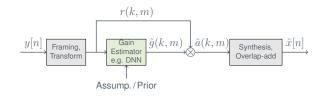
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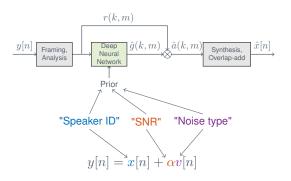
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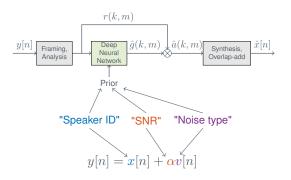


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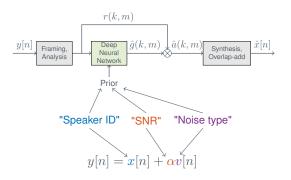
- We studied generalizability capability of deep neural network-based speech enhancement algorithms for additive-noise corrupted speech [1].
- Specifically, our goal was to study the generalization error w.r.t. three dimensions
 - Speaker Identity
 - Signal-to-Noise Ratio
 - Noise type
- We trained multiple DNNs with various priors.
- Generalization was evaluated using PESQ and STOI, which are speech quality and intelligibility estimators, respectively.



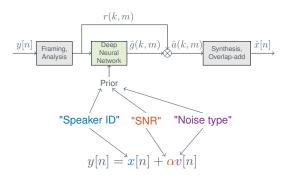
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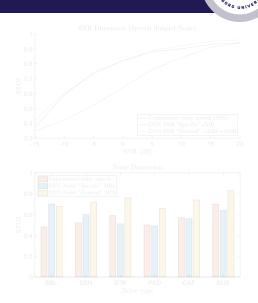


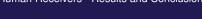
^[1] M. Kolbæk, et al., IEEE TASLP, 2017

Generalization of DNN based Speech Enhancement Human Receivers - Results and Conclusion

Results and Conclusion

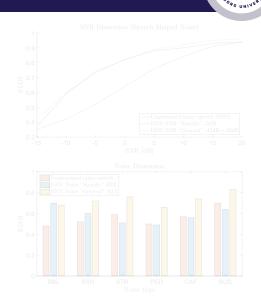
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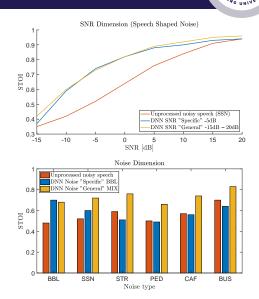
Results and Conclusion

- Performance (PESQ and STOI) is generally reduced when a "narrow" system is tested in a more general scenario.
- Performance is comparable or exceeding performance of a classical technique.
- Matching the noise type is the most critical, whereas matching the speaker and SNR is less critical.
- Listening tests show small improvement in speech intelligibility relative to previously published results.
- Both PESQ and informal listening tests indicate that DNN systems improve speech quality.



Results and Conclusion

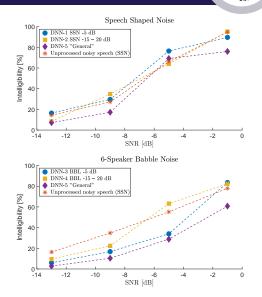
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Generalization of DNN based Speech Enhancement

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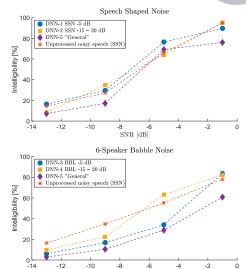
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Generalization of DNN based Speech Enhancement Human Receivers - Results and Conclusion

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SNR [dB]



Generalization of DNN based Speech Enhancement Human Receivers - Speech Intelligibility

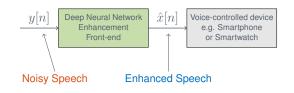
• Generalization of Deep Learning based Speech Enhancement

- Human Receivers Speech Intelligibility
- Machine Receivers Speaker Verification
- On STOI Optimal Deep Learning based Speech Enhancement
- Permutation Invariant Training for Deep Learning based Speech Separation
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Motivation

- Digital devices with voice-user interfaces struggle in "cocktail-party" conditions.
- Such devices can benefit from denoising front-ends.
- A State-of-the-art noise-robust speaker verification system relies on speaker dependent non-negative matrix factorization (Thomsen *et al.* 2016).

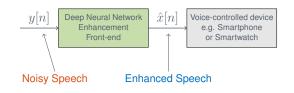
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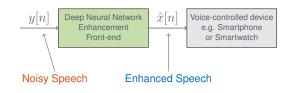
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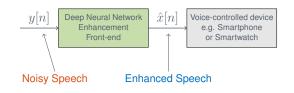
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Generalization of DNN based Speech Enhancement Machine Receivers - Contribution

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- We designed a DNN based speech enhancement front-end for a speaker verification system [2].
- Goal was to study the generalization error w.r.t. three dimensions:
 - Speaker Identity
 - Signal-to-Noise Ratio
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- Generalization was evaluated using equal error rates and the results were compared to existing enhancement techniques.





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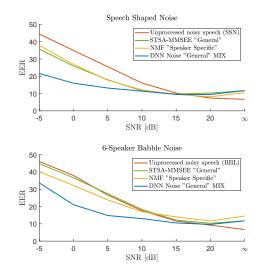
Generalization of DNN based Speech Enhancement

Results

- Male-speaker "general" DNN-based speech enhancement front-end generally leads to lower EER compared to classical techniques.
- Even NMF which is "narrow", i.e. speaker, text, and noise type dependent.

Conclusion

- DNN based speech enhancement front-end improves state-of-the-art noise-robust speaker verification.
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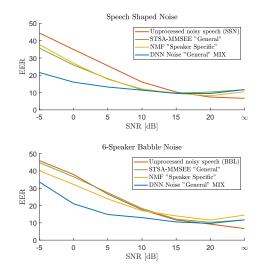
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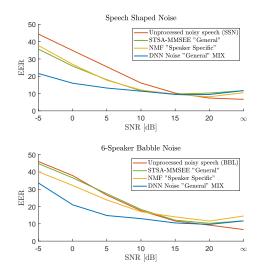
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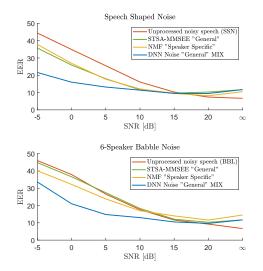


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Morten Kolbæk | Single-Microphone Speech Enhancement and Separation Using Deep Learning

On STOI Optimal DNN based Speech Enhancement

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On STOI Optimal DNN based Speech Enhancement Motivation, Research Gap, and Contribution

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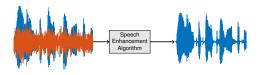
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- Can we use a function with a stronger link to SI? e.g. the STOI SI estimator.

Research Gap

No DNN-based speech enhancement algorithm exists that maximize STOI.

Contribution

• We propose such an algorithm [3,4].



Mean-Square Error:

$$J_{MSE} = \frac{1}{K} \sum_{k=1}^{K} (a(k,m) - \hat{a}(k,m))^2$$

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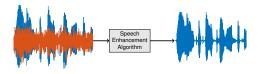
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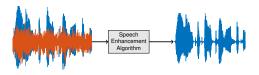
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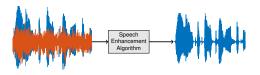
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On STOI Optimal DNN based Speech Enhancement Motivation, Research Gap, and Contribution

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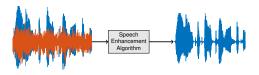
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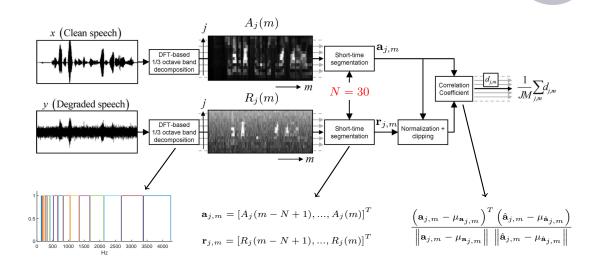
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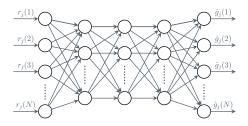
On STOI Optimal DNN based Speech Enhancement Short-Time Objective Intelligibility (STOI) - Architecture

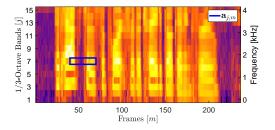


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On STOI Optimal DNN based Speech Enhancement Proposed STOI-based Approach







$$\hat{\mathbf{a}}_{j,m} = \hat{\mathbf{g}}_{j,m} \circ \mathbf{r}_{j,m}$$

- $\hat{\mathbf{g}}_{j,m}$: Estimated Gains
- $\mathbf{r}_{j,m}$: Noisy Speech 1/3-Octave band
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$$\mathcal{L}_{ELC} = \frac{\left(\mathbf{a}_{j,m} - \mu_{\mathbf{a}_{j,m}}\right)^{T} \left(\hat{\mathbf{a}}_{j,m} - \mu_{\hat{\mathbf{a}}_{j,m}}\right)}{\left\| \mathbf{a}_{j,m} - \mu_{\mathbf{a}_{j,m}} \right\| \left\| \hat{\mathbf{a}}_{j,m} - \mu_{\hat{\mathbf{a}}_{j,m}} \right\|}$$

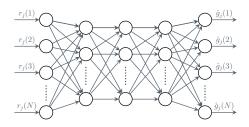
$$\mathcal{L}_{EMSE} = \frac{1}{N} \left\| \mathbf{a}_{j,m} - \hat{\mathbf{a}}_{j,m} \right\|^2$$

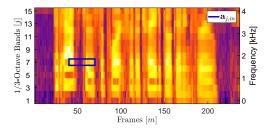
- \mathcal{L} : Loss for sample m in band j ELC: Envelope Linear Correlation
- EMSE : Envelope Mean-Square Error



On STOI Optimal DNN based Speech Enhancement Proposed STOI-based Approach

► **STOI**-based Speech Enhancement Model





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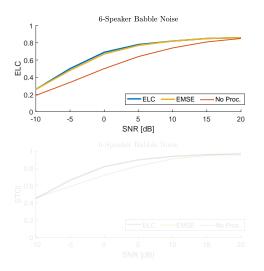
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Experimental Results

- DNNs designed to maximize approximate-STOI, improves ELC at various SNRs (and noise types).
- Similar conclusions can be drawn for DNNs that minimize EMSE.
- Same conclusions hold when the same DNNs are evaluated using STOI.
- Apparently, nothing to gain in terms of STOI, when maximizing ELC compared to minimizing MSE.

New Hypothesis

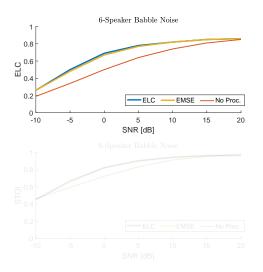




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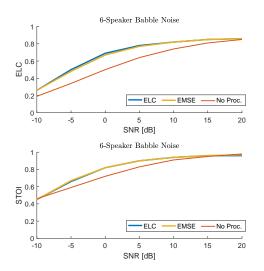




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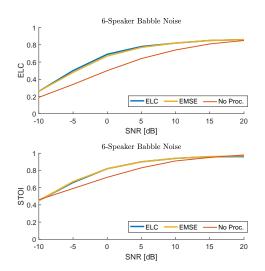


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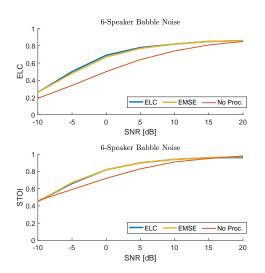
Are the solutions in fact the same?



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Method

► Using Bayesian statistics we derive the maximum mean ELC (MMELC) estimator.

Result

We show, under certain general conditions, that the MMELC estimator is asymptotically (in N) equivalent to the classical STSA-MMSE estimator.

- The STSA-MMSE estimator leads to the same approximate-STOI value as the MMELC estimator.
- For practical DNN based speech enhancement algorithms this is valid already at N > 15.
- No reason to optimize for ELC if the goal is to perform optimally w.r.t. STOL STSA-MSE is near optimal.

$$\begin{split} \hat{\underline{a}}_{MMELC} &= \arg\max_{\underline{\hat{a}}} \int \mathcal{L}_{ELC}\left(\underline{a},\underline{\hat{a}}\right) f_{\underline{A}|\underline{R}}\left(\underline{a}|\underline{r}\right) \, d\underline{a} \\ &= \frac{\mathbb{E}_{\underline{A}|\underline{r}}\left[\underline{e}(\underline{A}|\underline{r})\right]}{\|\mathbb{E}_{\underline{A}|\underline{r}}\left[\underline{e}(\underline{A}|\underline{r})\right]\|} \end{split}$$

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$$\begin{split} \hat{\underline{a}}_{MMELC} &= \arg\max_{\underline{\hat{a}}} \int \mathcal{L}_{ELC}\left(\underline{a},\underline{\hat{a}}\right) f_{\underline{A}|\underline{R}}\left(\underline{a}|\underline{r}\right) \, d\underline{a} \\ &= \frac{\mathbb{E}_{\underline{A}|\underline{r}}\left[\underline{e}(\underline{A}|\underline{r})\right]}{\|\mathbb{E}_{\underline{A}|\underline{r}}\left[\underline{e}(\underline{A}|\underline{r})\right]\|} \end{split}$$

$$\begin{split} \hat{\underline{a}}_{MMSE} &= \arg\min_{\underline{\hat{a}}} \int \left(\underline{a} - \underline{\hat{a}}\right)^2 f_{\underline{A}|\underline{R}}\left(\underline{a}|\underline{r}\right) \, d\underline{a} \\ &= \mathbb{E}_{A|r}\left[\underline{A}|\underline{r}\right] \end{split}$$

$$\lim_{N \to \infty} \hat{\underline{a}}_{MMELC} = \hat{\underline{a}}_{MMSE} - \mu_{\hat{\underline{a}}_{MMSE}}$$

Permutation Invariant Training for Speech Separation

Generalization of Deep Learning based Speech Enhancement

- Human Receivers Speech Intelligibility
- Machine Receivers Speaker Verification
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- Permutation Invariant Training for Deep Learning based Speech Separation
- Summary and Conclusion

Permutation Invariant Training for Speech Separation Motivation, Research Gap, and Contribution

Motivation

- Speech separation algorithms are useful for various applications.
- E.g. "Cocktail party" situations.
- Existing solutions are complicated or limited.

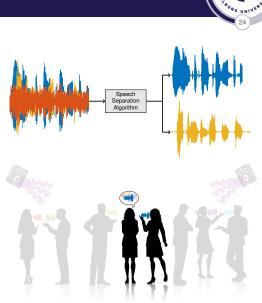
Research Gap

No DNN-only solution exists for speaker independent multi-talker speech separation.

Contribution

▶ We propose such algorithms [5,6,7].

[5] D. Yu, *et al.*, *IEEE ICASSP*, 2017
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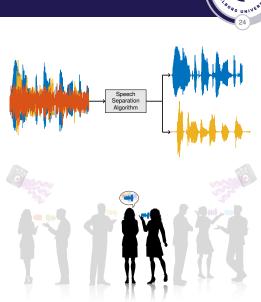
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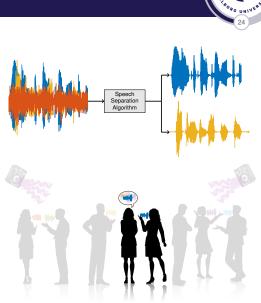
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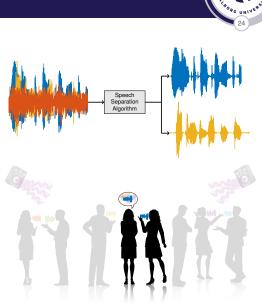
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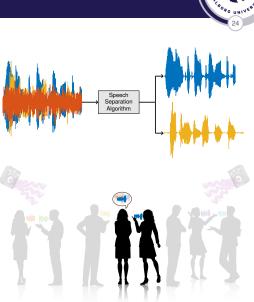
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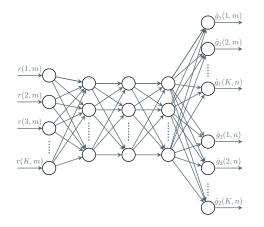
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Permutation Invariant Training for Speech Separation

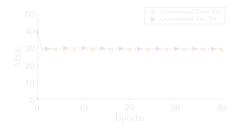
• 2-Speaker Separation Model (S = 2)



MSE Cost Function

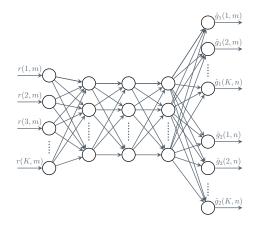
$$J_{MSE} = \frac{1}{SK} \sum_{s=1}^{S} \sum_{k=1}^{K} (a_s(k,m) - \hat{g}_s(k,m)r(k,m))^2$$
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Training Progress for Speaker "Independent" Data



Permutation Invariant Training for Speech Separation

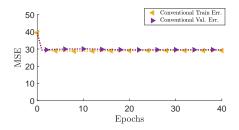
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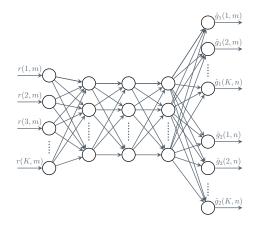
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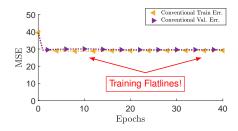
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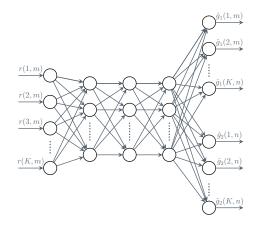
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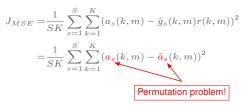
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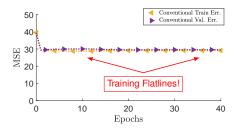
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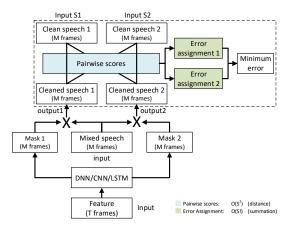
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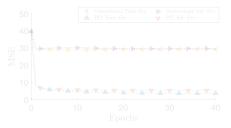
2-Speaker Frame-level PIT Technique



PIT MSE Cost Function

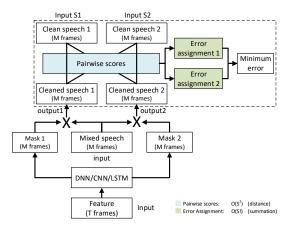
$$J_{PIT} = \min_{\theta \in \mathcal{P}} \frac{1}{SK} \sum_{s=1}^{S} \sum_{k=1}^{K} (a_s(k, m) - \hat{a}_{\theta(s)}(k, m))^2$$

PIT Training Progress (SGD)





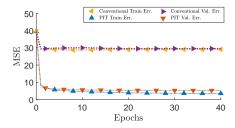
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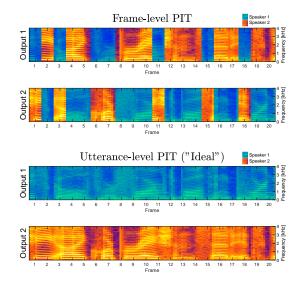
► PIT Training Progress (SGD)



Solution: Train with permutation corresponding to minimum utterance-level error (*for all m*).

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$$J_{uPIT} = \frac{1}{SK} \sum_{s=1}^{S} \sum_{k=1}^{K} (a_s(k,m) - \hat{a}_{\theta^*(s)}(k,m))^2$$

- Utterance-level PIT minimizes the utterance-level error, hence reducing context switch.
- **Note:** No extra computations during inference.

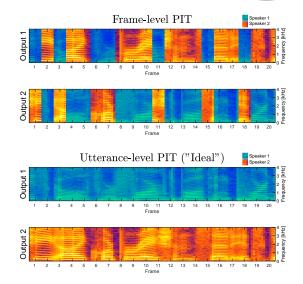


Permutation Invariant Training for Speech Separation Utterance-level Permutation Invariant Training

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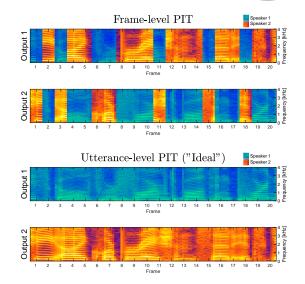


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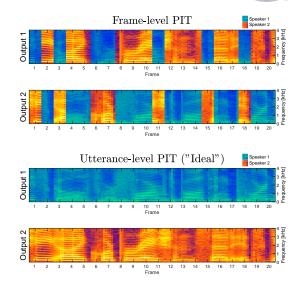


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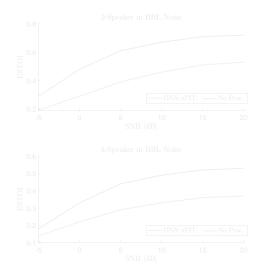
Permutation Invariant Training for Speech Separation Results and Conclusion

Result

- State-of-the-art on 2-talker and 3-talker speaker-independent speech separation tasks.
- DNNs trained with uPIT works well for speech separation and enhancement jointly.
- More interestingly, works well without prior knowledge about the number of speakers.

Conclusion

uPIT is a DNN training technique that enable DNN-only algorithms for speaker-independent multi-talker speech separation and enhancement.



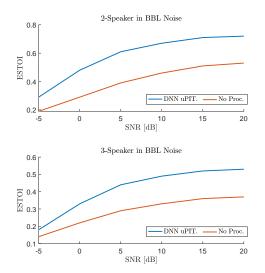


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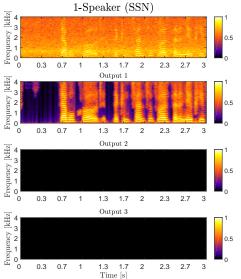
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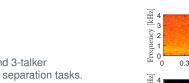


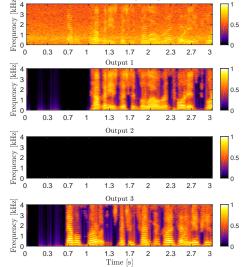
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2-Speakers (SSN)

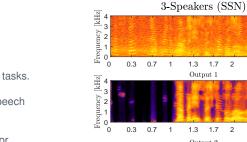
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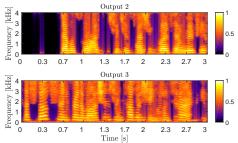
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1.7 2 2.3

1.7

2

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05

2.7

2.3 2.7 3

3





He cites double-quote the law of large numbers

Summary and Conclusion

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Academic Output: 3 Journal papers and 4 Conference papers

- M. Kolbæk, Z. H. Tan, and J. Jensen, "Speech Intelligibility Potential of General and Specialized Deep Neural Network Based Speech Enhancement Systems," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 25, no. 1, pp. 153–167, 2017.
- [2] M. Kolbœk, Z. H. Tan, and J. Jensen, "Speech Enhancement using Long Short-Term Memory based Recurrent Neural Networks for Noise Robust Speaker Verification," in *Proc. SLT*, 2016, pp. 305–311.
- [3] M. Kolbæk, Z.-H. Tan, and J. Jensen, "Monaural Speech Enhancement using Deep Neural Networks by Maximizing a Short-Time Objective Intelligibility Measure," in *Proc. ICASSP*, 2018, pp. 5059 – 5063.
- [4] M. Kolbæk, Z. H. Tan, and J. Jensen, "On the Relationship Between Short-Time Objective Intelligibility and Short-Time Spectral-Amplitude Mean-Square Error for Speech Enhancement," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 27, no. 2, pp. 283–295, 2018.
- [5] D. Yu, M. Kolbæk, Z. H. Tan, and J. Jensen, "Permutation Invariant Training of Deep Models for Speaker-independent Multi-talker Speech Separation," in *Proc. ICASSP*, 2017, pp. 241–245.
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- [7] M. Kolbæk, D. Yu, Z. H. Tan, and J. Jensen, "Joint separation and denoising of noisy multi-talker speech using recurrent neural networks and permutation invariant training," in *Proc. MLSP*, 2017, pp. 1–6.

Concluding Remarks

Generalizability [1, 2]

- Matching the noise type is the most critical, whereas matching the speaker and SNR is less critical if a modest amount of speakers are included in the training set.
- A male-speaker "general" DNN based speech enhancement front-end achieves state-of-the-art performance on a speaker verification task.

• Optimality [3, 4]

- The STSA-MMSE estimator is asymptotically equivalent to the MMELC estimator.
- The STSA-MSE cost function leads to enhanced speech signals which are essentially optimal in terms of STOI. In other words, there is no benefit from optimizing for STOI.

- A training criterion that enable DNNs to work well on single-microphone speaker-independent multi-talker speech separation and enhancement.
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 - Matching the noise type is the most critical, whereas matching the speaker and SNR is less critical if a modest amount of speakers are included in the training set.
 - A male-speaker "general" DNN based speech enhancement front-end achieves state-of-the-art performance on a speaker verification task.

► Optimality [3, 4]

- ► The STSA-MMSE estimator is asymptotically equivalent to the MMELC estimator.
- The STSA-MSE cost function leads to enhanced speech signals which are essentially optimal in terms of STOI. In other words, there is no benefit from optimizing for STOI.

- A training criterion that enable DNNs to work well on single-microphone speaker-independent multi-talker speech separation and enhancement.
- Simple solution to the label permutation problem.
- Achieves state-of-the-art performance.

Summary and Conclusion Not there yet, but a small step closer.



Thank you.



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