

## ICASSP2021 Tutorial: T-9

# Distant conversational speech recognition and analysis: Recent advances, and trends towards end-to-end optimization

Keisuke Kinoshita (NTT Corporation)

Yusuke Fujita (LINE Corporation)

Naoyuki Kanda (Microsoft)

Shinji Watanabe (Carnegie Mellon University)



[https://github.com/ICASSP2021-tutorial9/Distant\\_conversational\\_ASR\\_and\\_analysis/blob/main/ICASSP2021\\_tutorial\\_T9\\_slides.pdf](https://github.com/ICASSP2021-tutorial9/Distant_conversational_ASR_and_analysis/blob/main/ICASSP2021_tutorial_T9_slides.pdf)

The latest version of our tutorial slide is available at

# Tutorial presenters

# Tutorial Presenters (1/4)



## Keisuke Kinoshita

*Distinguished Researcher, NTT Corporation*

received the M. Eng. degree and Ph.D degree from Sophia University in Tokyo in 2003 and 2010, respectively. After joining NTT in 2003, he has been working on various topics related to distant conversational speech recognition such as 1ch/multi-channel speech enhancement, speaker diarization and robust speech recognition. He published more than 100 technical papers in refereed journals and conference proceedings, and also contributed to 5 book chapters. He was a Chief coordinator of the REVERB challenge 2014, and a member of IEEE AASP TC since 2018. He is honored to receive the 2006 IEICE Paper Award, the 2009 ASJ Outstanding Technical Development Prize, the 2011 ASJ Awaya Prize, the 2012 Japan Audio Society Award, the 2015 IEEE-ASRU Best Paper Award Honorable Mention.

## Tutorial Presenters (2/4)



### Yusuke Fujita

*Senior Research Engineer, LINE Corporation*

received his B.S. and M.S. degrees in computer science from Waseda University, Tokyo, Japan, in 2003 and 2005, respectively. He was a Senior Researcher at Hitachi Ltd. in Tokyo, Japan from 2005 to 2021, and was a Visiting Scholar at Johns Hopkins University, MD, USA (2018-2020). His research interests include speech recognition, speech separation, and speaker diarization. He has been working on end-to-end speaker diarization and distant speech recognition through the CHiME-5 challenge session chair, the CHiME-6 Scientific Committee, and the participation of CHiME-3,4 and CHiME-5 challenges.

## Tutorial Presenters (3/4)



Naoyuki Kanda

*Principal Researcher, Microsoft Corp.*

received a B.S. in Engineering, M.S. in Informatics, and Ph.D. in Informatics from Kyoto University, Japan, in 2004, 2006, and 2014, respectively. He was a Senior Researcher at Hitachi Ltd. in Tokyo, Japan from 2006 to 2019, and held appointments as a Research Expert (2014-2016) and Cooperative Visiting Researcher (2016-2017) at the National Institute of Information and Communications Technology (NICT) in Kyoto, Japan. He has been actively working on the joint modeling of speech separation, speech recognition, and speaker identification / diarization. His algorithms and systems for the distant conversational speech recognition won the second prize at the CHiME-5 speech recognition competition in 2018.

## Tutorial Presenters (4/4)



### Shinji Watanabe

*Associate Professor, Carnegie Mellon University*

received his B.S., M.S., and Ph.D. degrees from Waseda University, Tokyo, Japan. He was a research scientist at NTT Communication Science Laboratories, Kyoto, Japan, from 2001 to 2011, and a Senior Principal Research Scientist at Mitsubishi Electric Research Laboratories (MERL), Cambridge, MA USA from 2012 to 2017. He has been actively working on the distant conversational speech recognition area through the CHiME speech separation and recognition challenge organization, a tutorial speaker about "Recent Advances in Distant Speech Recognition" in Interspeech'16, and a number of participations of the related project including the first place at the DIHARD challenge 2018.

# Notations & Abbreviations

# Abbreviations

AHC	Agglomerative Hierarchical Clustering	MVDR	Minimum Variance Distortionless Response
ASR	Automatic Speech Recognition	NIN	Network In Network
BLSTM	Bi-directional LSTM neural network	NN	Neural Network
BCE	Binary Cross Entropy	PIT	Permutation Invariant Training
BF	Beamformer	PLDA	Probabilistic Linear Discriminant Analysis
CNN	Convolutional Neural Network	PSD	Power Spectral Density
CE	Cross Entropy	ReLU	Rectified Linear Unit
CTC	Connectionist Temporal Classification	RIR	Room Impulse Response
DEV	Development set	RNN	Recurrent Neural Network
DER	Diarization Error Rate	RNN-T	RNN-Tranceducer
DNN	Deep Neural Network	RTF	Relative Transfer Function
E2E	End-to-End	RTTM	Rich Transcription Time Marked
EEND	End-to-End Neural Diarization	SAD	Speaker Activity Detection
EVAL	Evaluation set	SC	Speaker Counting
FIFO	First-In, First-Out	SD	Speaker Diarization
GMM	Gaussian Mixture Model	SOT	Serialized Output Training
GRU	Gated Recurrent Unit	Spk	Speaker
GSS	Guided Source Separation	STFT	Short-Time Fourier Transformation
HMM	Hidden Markov Model	SWER	Speaker-Attributed Word Error Rate
LM	Language Model	TF	Time-Frequency
LSTM	Long-Short Term Memory neural network	TDNN	Time-Delay Neural Network
LLR	Log Likelihood Ratio	TS-VAD	Target-Speaker Voice Activity Detection
ML	Maximum Likelihood	UIS-RNN	Unbounded Interleaved-State RNN
MMI	Maximum Mututal Information	VAD	Voice Activity Detection
MMSE	Minimum Mean Squared Error	WER	Word Error Rate
MSE	Mean Squared Error	WPE	Weighted Prediction Error
		WSJ	Wall Street Journal

## Abbreviations

# Notations (1/2)

Mathematical expressions and operations

$a$	A scalar variable.
$\mathbf{a}$	A column vector.
$\mathbf{A}$	A matrix.
$C$	A constant.
$p(x)$	Probability density function
$\ \cdot\ _2$	Eucrdean norm of a vector
$\mathbb{R}$ and $\mathbb{C}$	A set of real scalars, and a set of complex scalars.
$\mathbb{R}^M$ and $\mathbb{R}^{M \times M}$	A set of $M$ dimentional real vectors, and a set of $M \times M$ dimentional real matrices. $\mathbb{C}^M$ and $\mathbb{C}^{M \times M}$ are defined similarly.

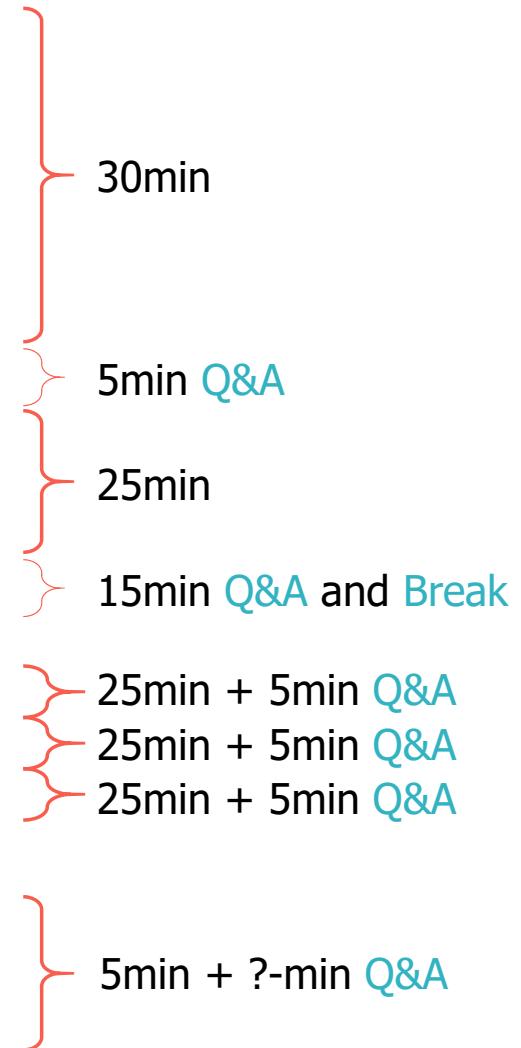
# Notations (2/2)

Indices	
$\tilde{t}, t, f, m$	Indices of time samples, time frames, frequency bins, microphones.
$k$	Index for speakers.
$i$	Index for spoken words or sub-words.
$\tilde{T}, T, F, M$	Total numbers of time samples, time frames, frequency bins, microphones.
$K$	Total numbers of speakers.
$I_k$	Total numbers of spoken words or sub-words for the $k$ -th speaker.
Symbols for Short Time Fourier Transformation (STFT) domain signals	
$s_{t,f,k} \in \mathbb{C}$	A clean speech signal for the $k$ -th speaker.
$x_{m,t,f,k} \in \mathbb{C}$	A microphone image of the $k$ -th speaker at the $m$ -th microphone, i.e., noiseless reverberant signal for the $k$ -th speaker at the $m$ -th microphone.
$y_{m,t,f} \in \mathbb{C}$	An observed signal at the $m$ -th microphone, i.e., noisy reverberant speech-mixture.
$n_{m,t,f} \in \mathbb{C}$	Diffuse noise.
Symbols for other domains	
$d_{t,k}$	Diarization label for the $k$ -th speaker at the time frame $t$ . A label for diarization systems.
$r_{n,k}$	The $n$ -th spoken word or sub-word of the $k$ -th speaker. A label for ASR systems.
$\mathbf{h}_t$	An embedding vector at the time frame $t$ .
$\mathbf{h}_k^{(\text{spk})}$	A speaker embedding vector for the $k$ -th speaker.
$s_k[\tilde{t}]$	A time-domain clean speech signal for the $k$ -th speaker. The other time-domain symbols are defined similarly.
$o$	Observed signal in a feature domain, e.g., Log-Mel feature domain.

# Agenda

## Part 1: Introduction

- 1.1. What is distant conversational ASR and analysis?
- 1.2. Why is it difficult?
- 1.3. Why is it important?
- 1.4. Its research history
- 1.5. Typical systems for distant conversational ASR and analysis



## Part 2: Current state-of-the-art systems

- 2.1. Descriptions of the techniques
- 2.2. Reproducible baselines

## Part 3: A new research trend: Jointly optimal systems

- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$  ( $x$ : other functionalities)
- 3.3. ASR +  $x$

## Part 4: Summary and discussion

- 4.1. Strengths of each approach
- 4.2. Challenges and fundamental difficulties

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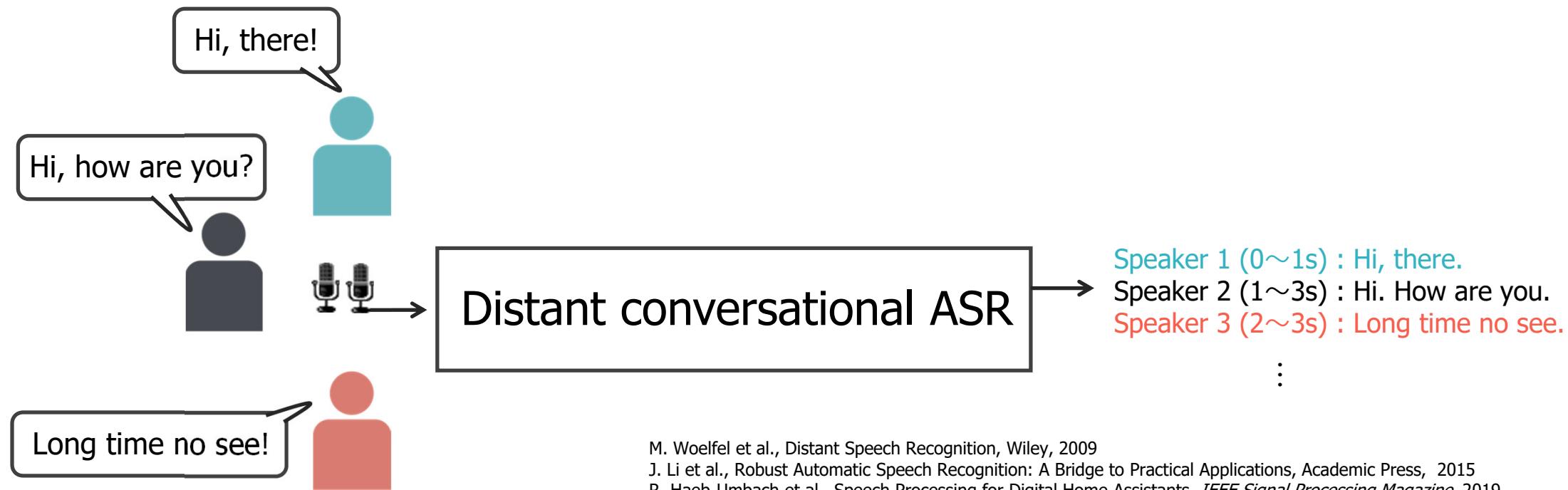
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# What is distant conversational ASR and analysis?

- A task to obtain “**who spoke what and when**” information from **natural conversation**
  - For “who part”, systems are not often required to output a name for the speakers, i.e., only a generic id which is unique within a meeting/conversation
- Sometimes, referred to as meeting recognition and meeting analysis



# What is distant conversational ASR and analysis?

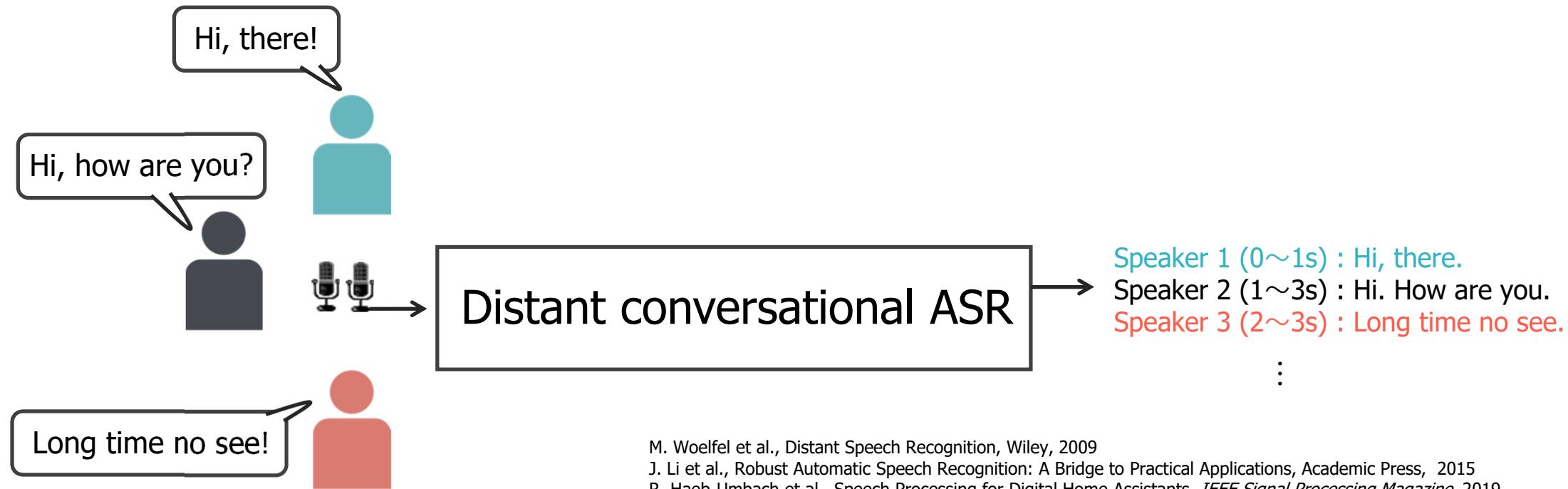
## Real-time Meeting Analysis System

NTT Communication Science Laboratories 2010

T. Hori *et al.*, "Low-Latency Real-Time Meeting Recognition and Understanding Using Distant Microphones and Omni-Directional Camera," in IEEE TASLP, 2012

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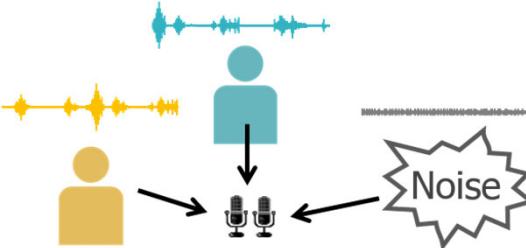
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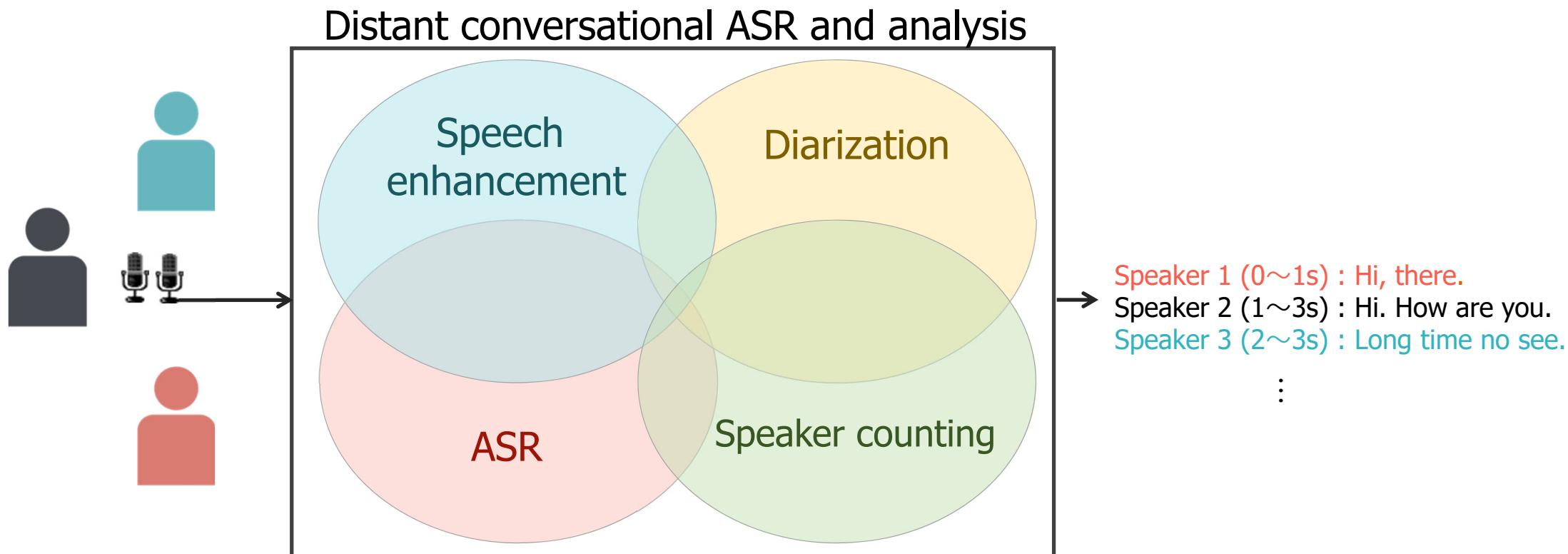
# Why is distant conversational ASR and analysis difficult?

## 1. Difficulties originated from the scenario

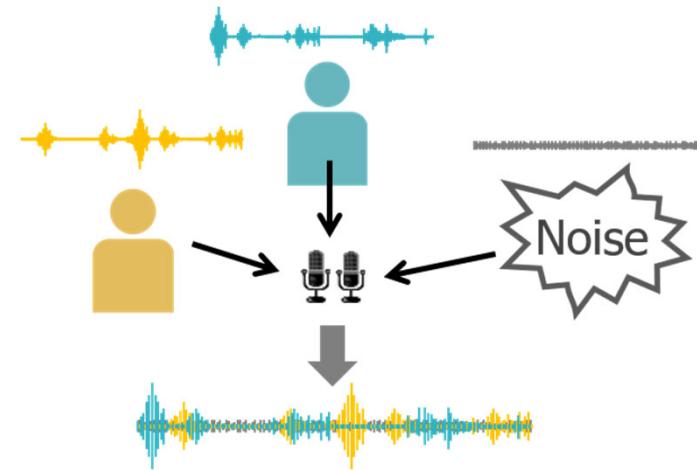
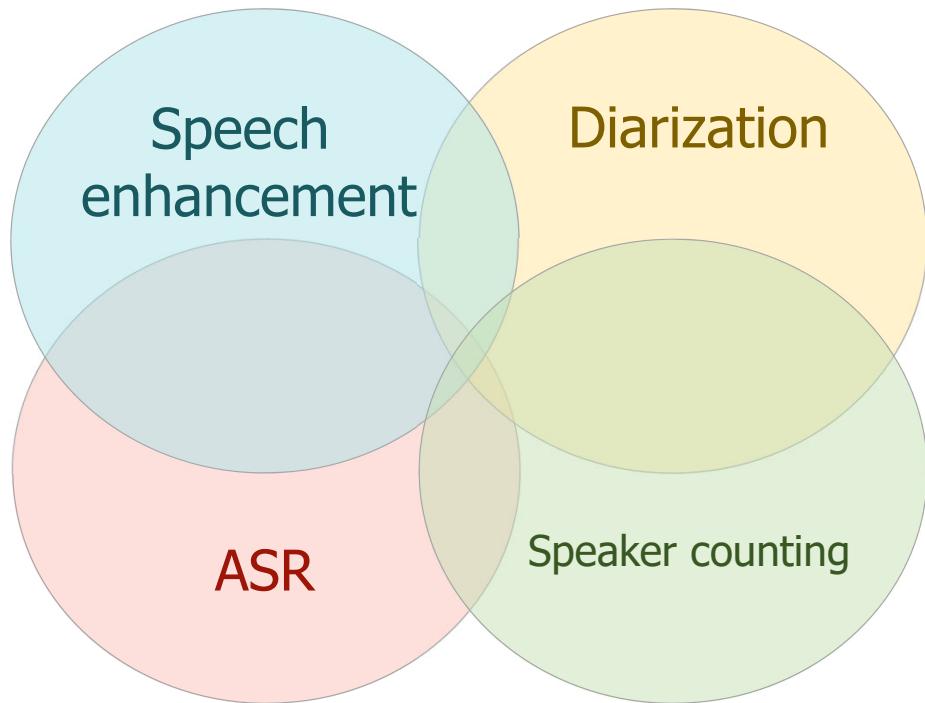
- Acoustic point of view
  - Far-field effect (noise, reverberation)  Speech enhancement
  - Highly interactive, and thus overlapped speech Speaker counting
  - Varied recording environments (e.g., varied number of speakers)
  - Unsegmented (no explicit utterance timing information) Diarization
- Lexical point of view
  - Spontaneous speech (e.g., variable articulation, inconsistent speaking rate, pause fillers, false-starts and self-edits) ASR

# Introduction of sub-tasks

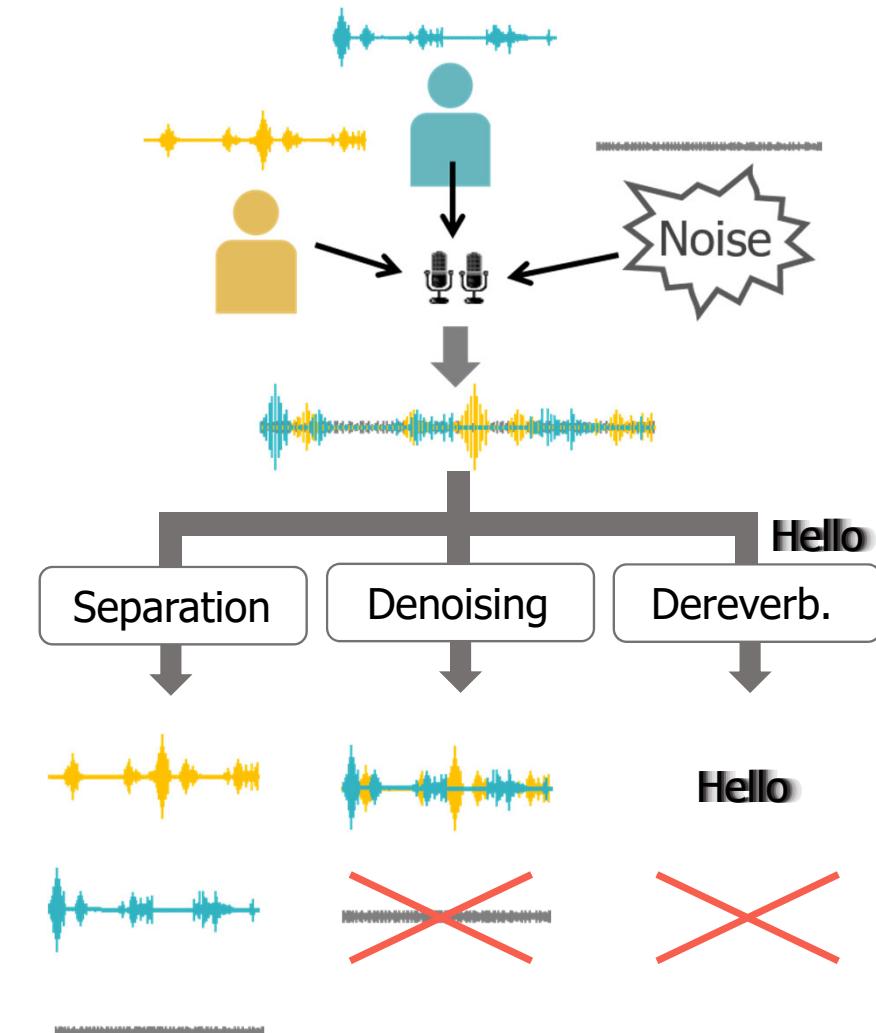
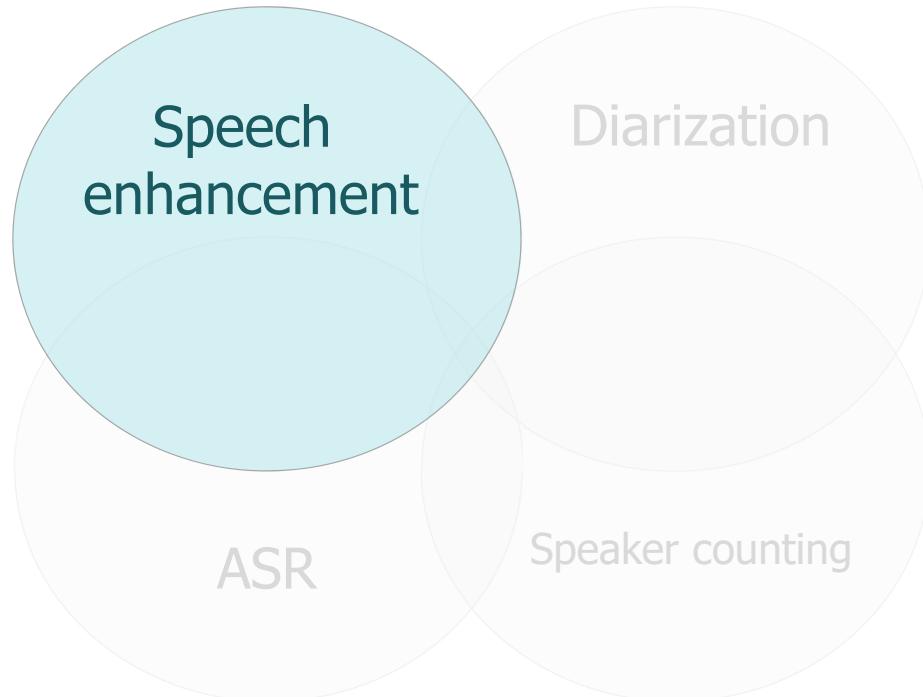
- Distant conversational ASR and analysis can comprises **4 sub-tasks**, speech enhancement, ASR, diarization, and speaker counting.
- **All these sub-tasks must be accomplished** to achieve the goal.



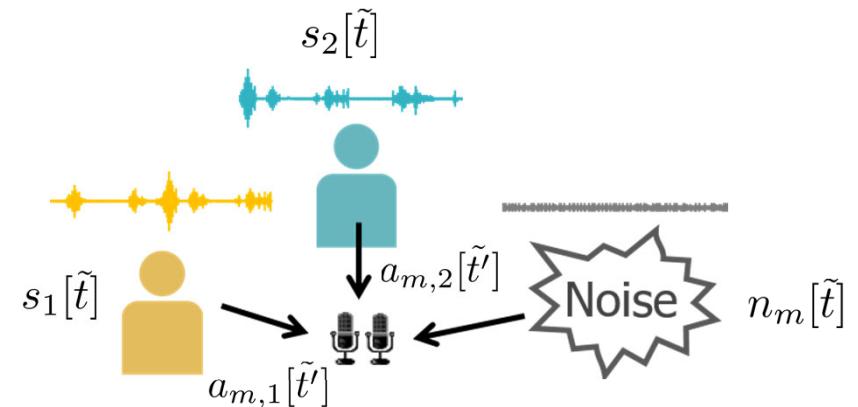
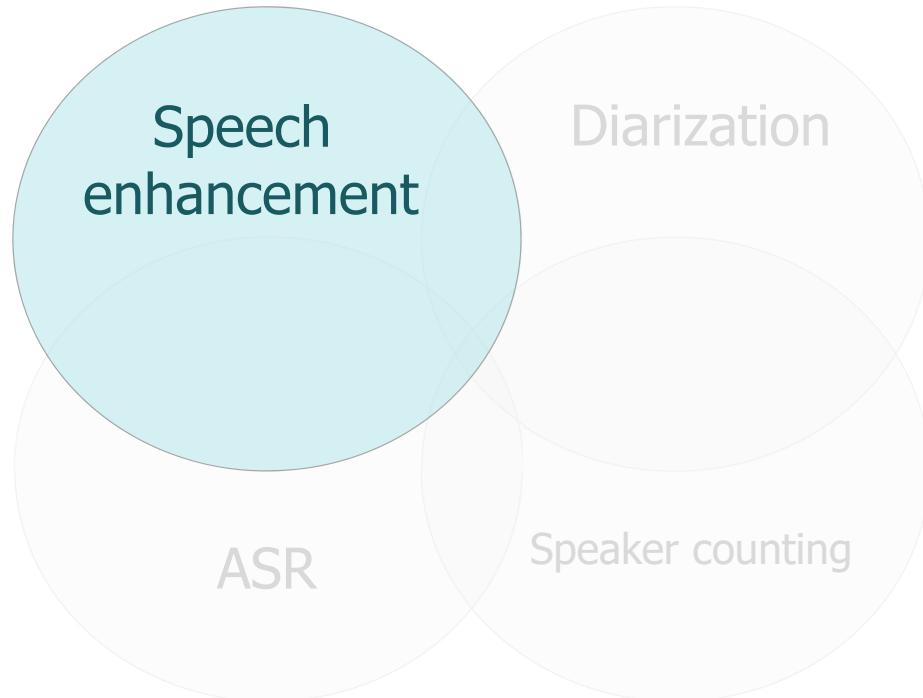
# Introduction of sub-tasks



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# Introduction of sub-tasks



$$\begin{aligned}y_m[\tilde{t}] &= \sum_{k=1}^K \sum_{\tilde{t}'} a_{m,k}[\tilde{t}'] s_k[\tilde{t} - \tilde{t}'] + n_m[\tilde{t}], \\&= \sum_{k=1}^K x_{m,k}[\tilde{t}] + n_m[\tilde{t}],\end{aligned}$$

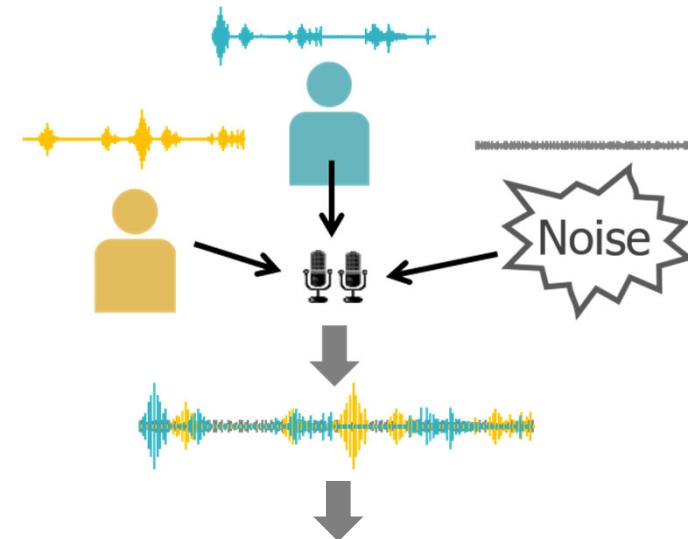
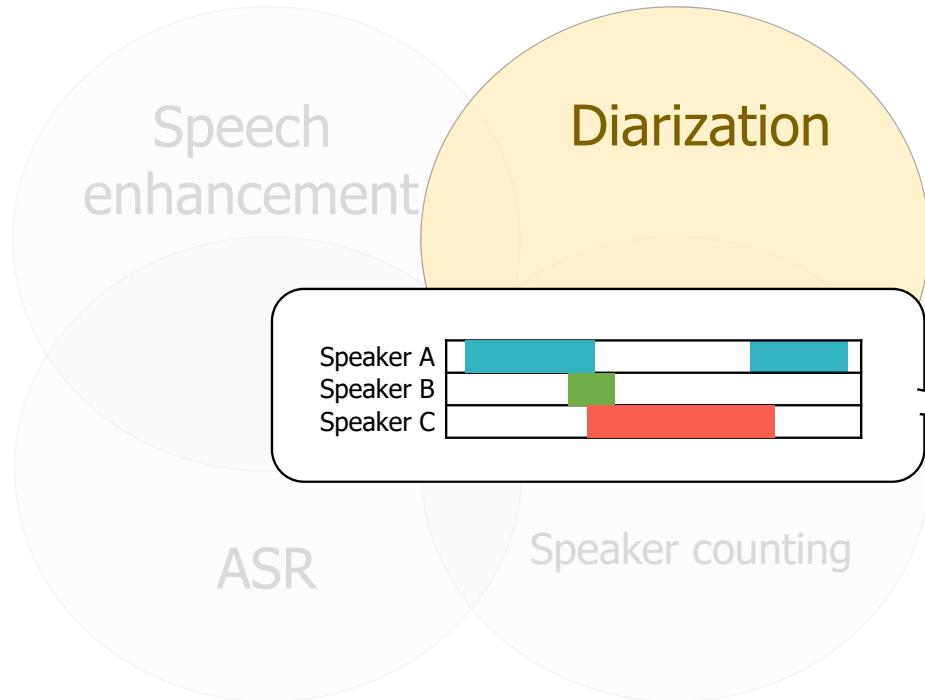
$m$  : Microphone index  
 $k$  : Speaker index

**Speech separation:** Extracting  $s_k[\tilde{t}]$  or  $x_{m,k}[\tilde{t}]$

**Denoising:** Reducing  $n_m[\tilde{t}]$

**Dereverberation:** Reducing effect of  $a_{m,k}[\tilde{t}']$

# Introduction of sub-tasks



Estimating speech activity label  $d_t \in \{0, 1\}^K$   
for each speaker at each time  $t$

(RTTM file)

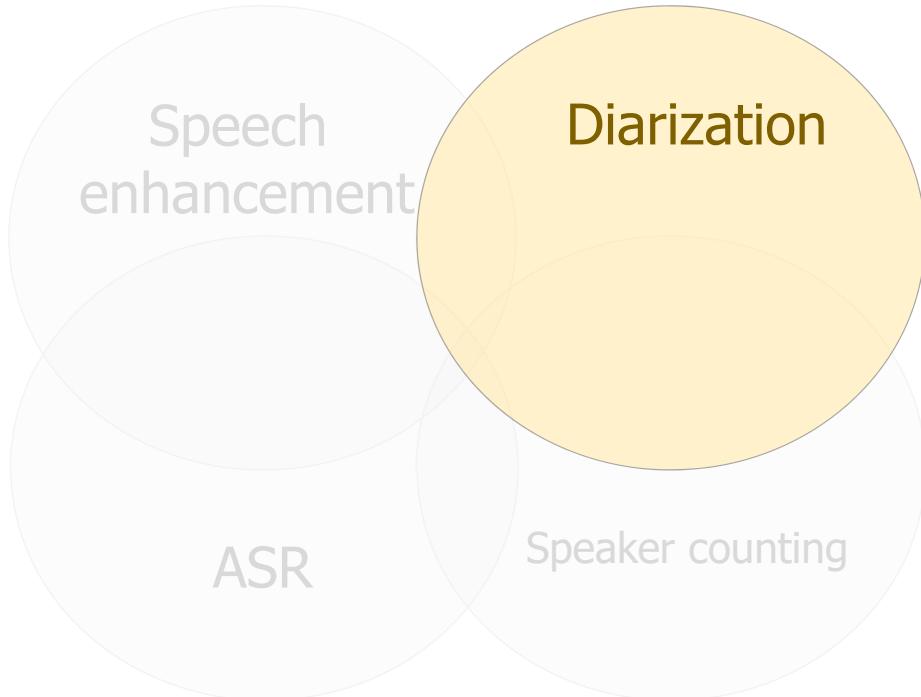
Spk ID	Start	Duration
A	0.5s	3.0s
B	2.5s	10.0s
C	14.5s	5.0s
A	20.0s	3.0s
:		

*Equivalently,*

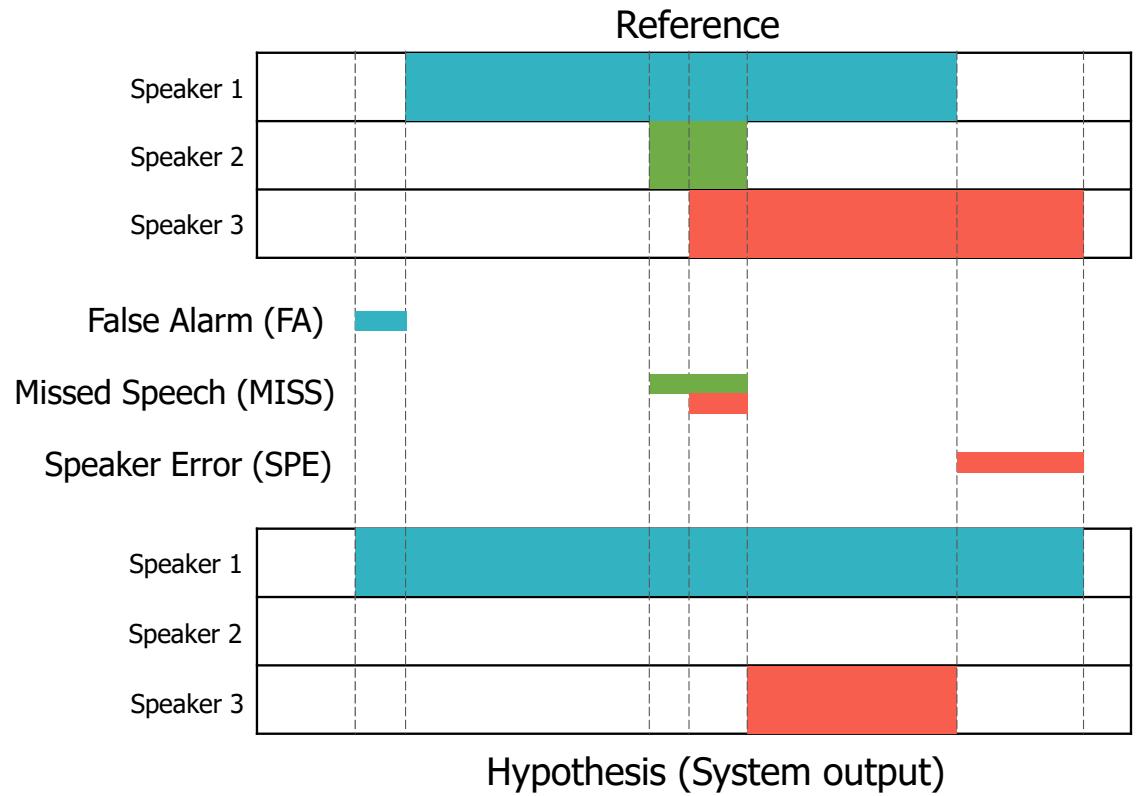
Estimating speech segments  
with speaker IDs, start timings,  
and their duration.

RTTM: Rich Transcription Time Marked

# Introduction of sub-tasks

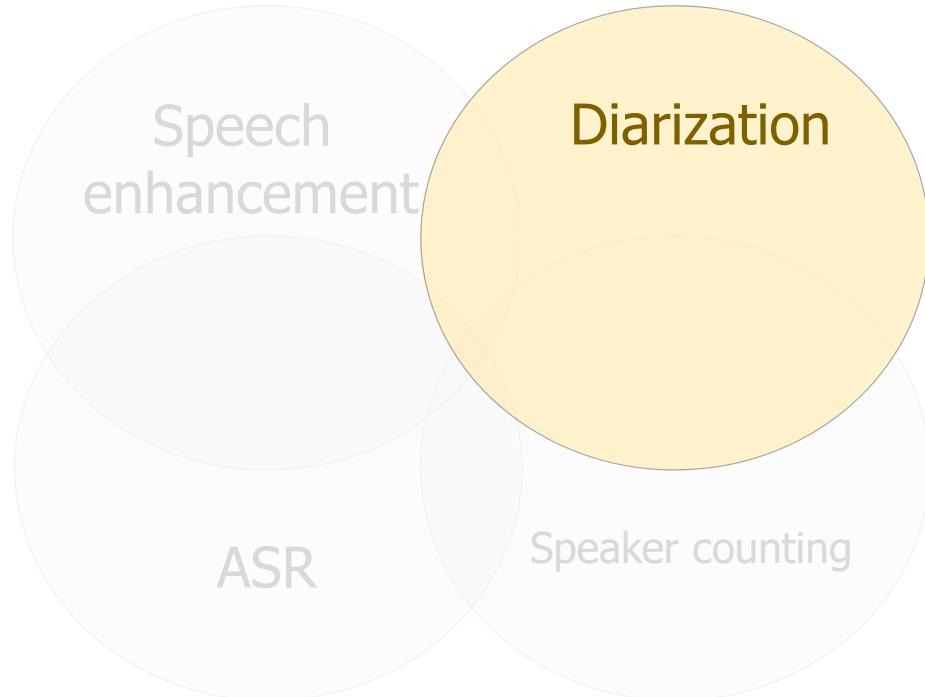


Evaluation metric: Diarization error rate (DER)

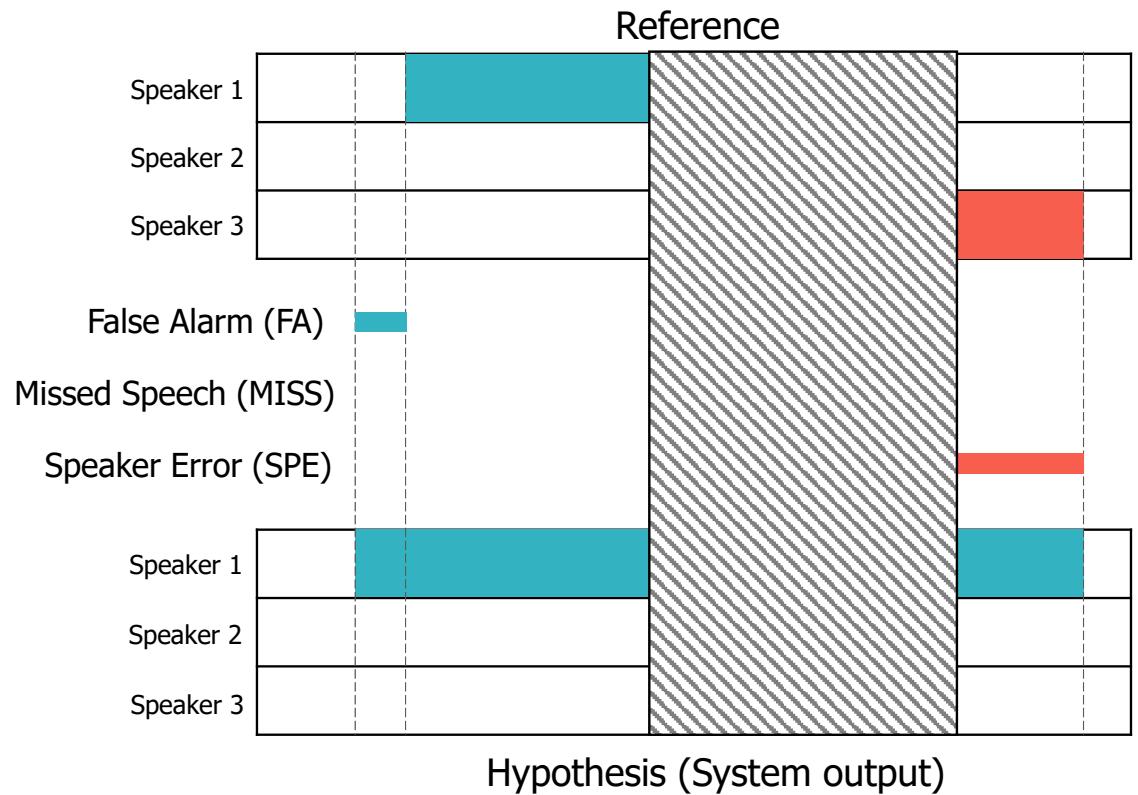


$$\text{DER} = \frac{\text{( \% of time ) FA + MISS + SPE}}{\text{Total Speech Duration}}$$

# Introduction of sub-tasks

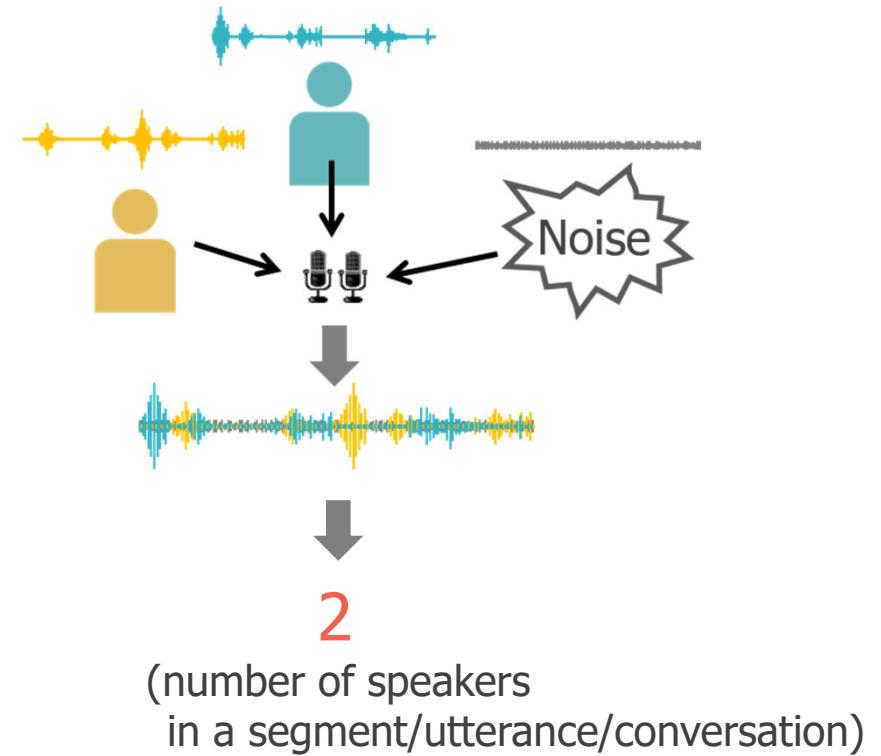
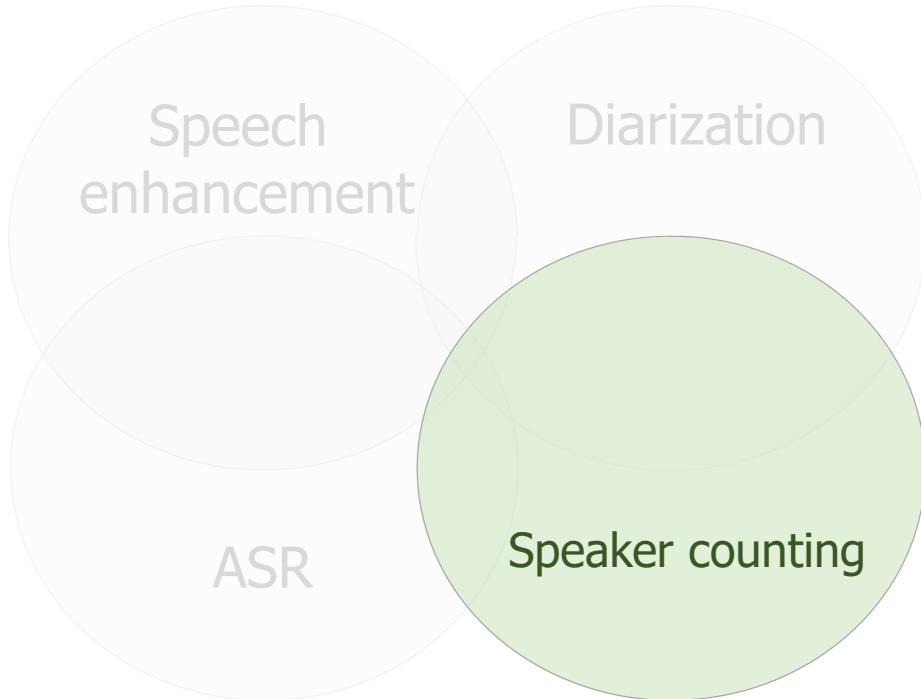


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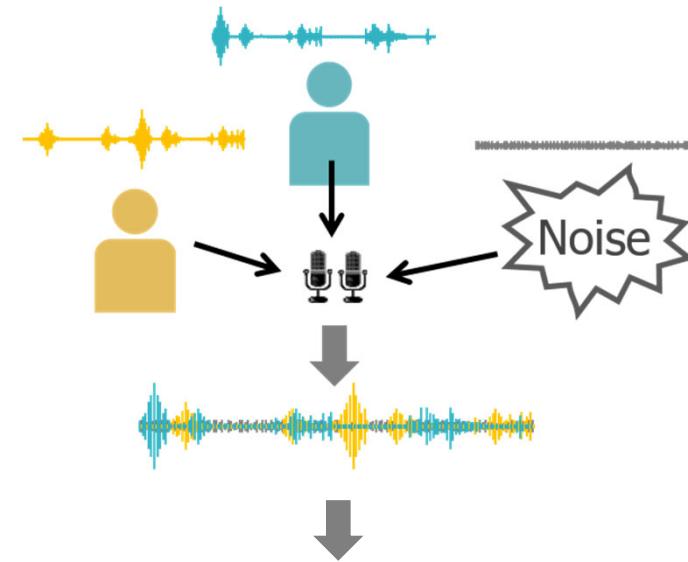
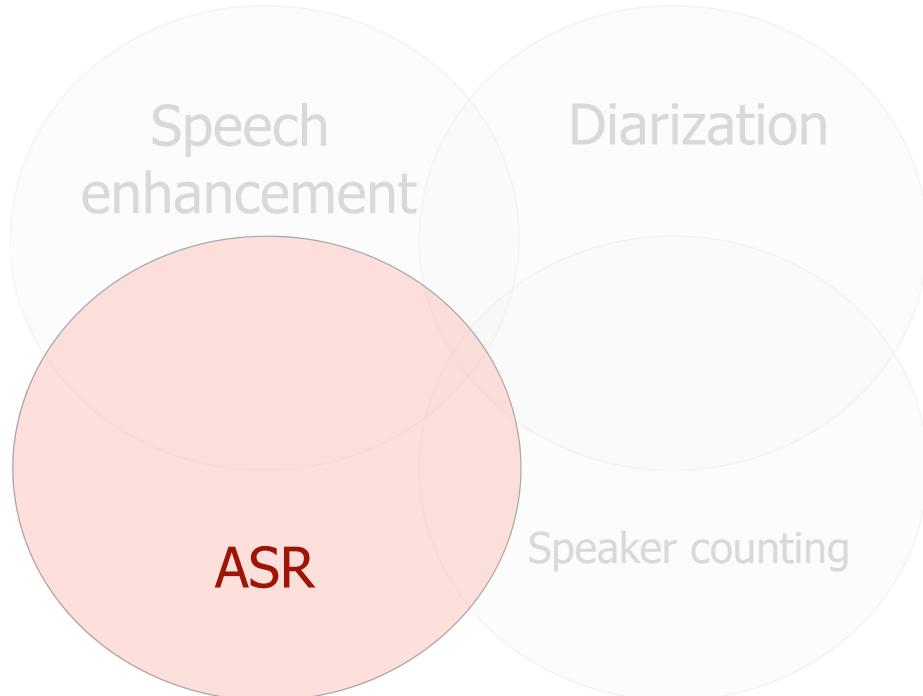


$$\text{DER} = \frac{\text{( \% of time ) } \text{ FA } + \text{ MISS } + \text{ SPE }}{\text{ Total Speech Duration }}$$

# Introduction of sub-tasks



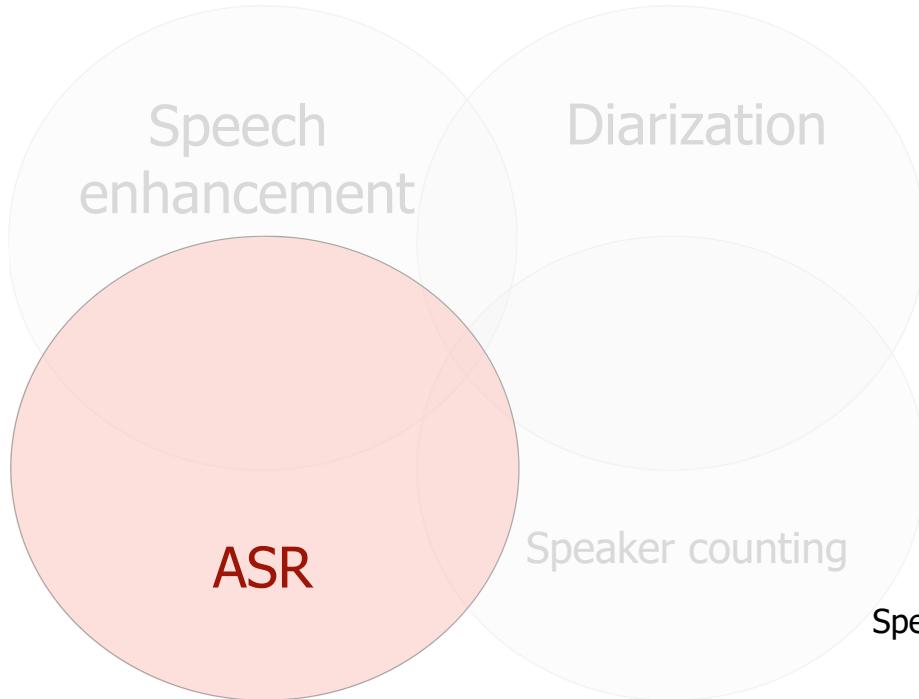
# Introduction of sub-tasks



Finding the most probable word sequence,  
typically, with the following Bayes decision theory:

$$\hat{\mathbf{W}} = \operatorname{argmax}_{\mathbf{W}} p(\underbrace{\mathbf{W} | \mathbf{O}}_{\text{Word sequence}}, \underbrace{\mathbf{O}}_{\text{Observed feature sequence}}) = \operatorname{argmax}_{\mathbf{W}} p(\mathbf{O} | \mathbf{W}) p(\mathbf{W}).$$

# Introduction of sub-tasks



Evaluation metric: Word Error Rate (WER)

$$WER = \frac{\text{Deletion (D)} + \text{Insertion (I)} + \text{Substitution (S)}}{\text{Total}}$$

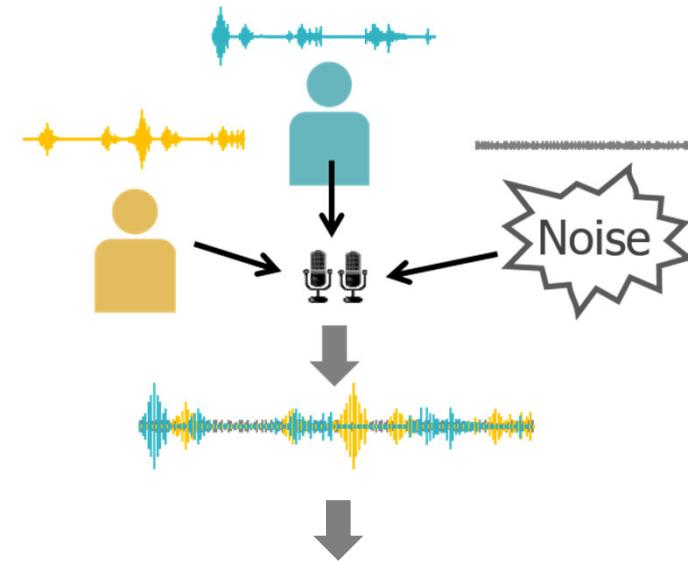
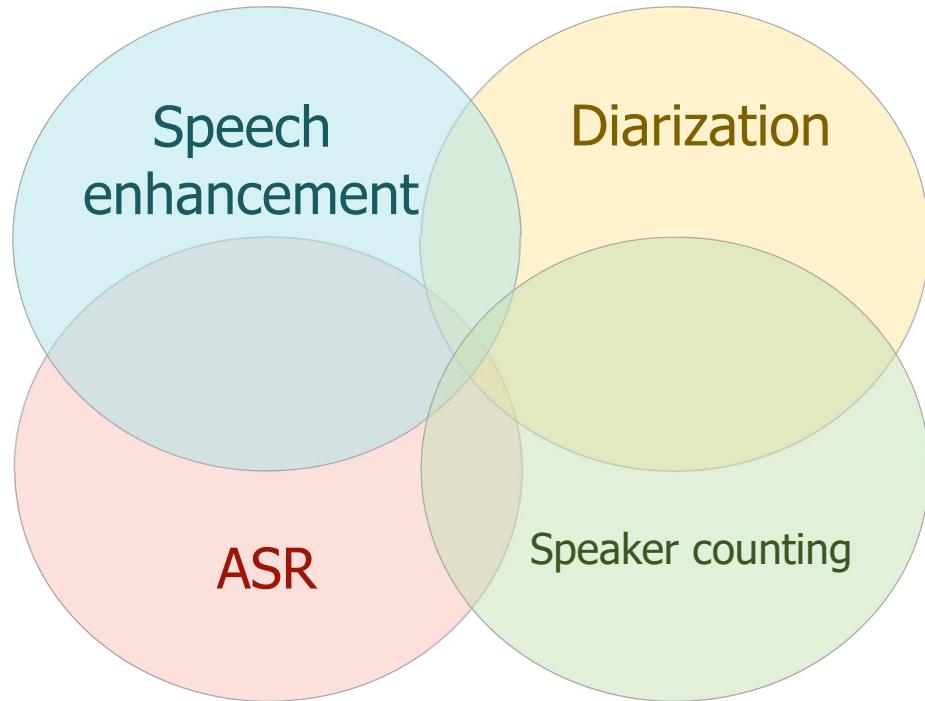
Reference:	A	B	C	D	F	G	H	I	J	K	L	M	N
Hypothesis:	A		C	D	E	F	G	H	I	J	K	M	M
Error type:				<i>D</i>		<i>I</i>							<i>S</i>

Evaluation metric: Speaker attributed Word Error Rate (SWER)

$$SWER = \frac{\text{Deletion (D)} + \text{Insertion (I)} + \text{Substitution (S)}}{\text{Total}}$$

Speaker 1	Reference:	A	B	C	D	E	F						
	Hypothesis:	A		C	D	E	F	G	H	I			
	Error type:				<i>D</i>			<i>I</i>	<i>I</i>	<i>I</i>			
Speaker 2	Reference:							G	H	I	J	K	L
	Hypothesis:										J	K	L
	Error type:							<i>D</i>	<i>D</i>	<i>D</i>			<i>I</i>

# Introduction of sub-tasks

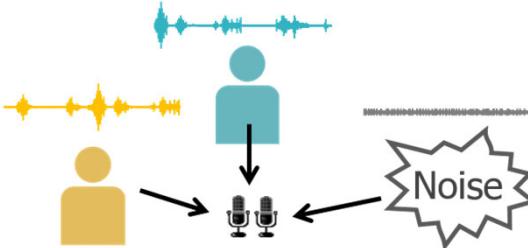


Speaker 1 (0~1s) : Hi, there.  
Speaker 2 (1~3s) : Hi. How are you.  
⋮      ⋮

All these sub-tasks must be accomplished to achieve the distant conversational speech recognition and analysis.

# Why is distant conversational ASR and analysis difficult?

## 1. Difficulties originated from the scenario

- Acoustic point of view
    - Far-field effect (noise, reverberation)
    - Highly interactive, and thus overlapped speech
    - Varied recording environments (e.g., varied number of speakers)
    - Unsegmented (no explicit utterance timing information)
  - Lexical point of view
    - Spontaneous speech (e.g., variable articulation, inconsistent speaking rate, pause fillers, false-starts and self-edits)
- 
- The diagram illustrates a distant conversational ASR scenario. It shows two stylized human figures: one yellow and one blue, each with a speech bubble above them containing a waveform. Below each figure is a pair of microphones. A yellow waveform is shown between the two microphones. To the right of the blue figure is a starburst shape labeled "Noise". A curved arrow labeled "Speech enhancement" points from the text to the yellow waveform. Another curved arrow labeled "Speaker counting" points from the text to the blue figure. A third curved arrow labeled "Diarization" points from the text to the blue figure. A fourth curved arrow labeled "ASR" points from the text to the yellow waveform.

## 2. Difficulty of system development

- Simply collecting and assembling **many required modules** is already very challenging.
- **Optimization of the whole system** is far more difficult.

**Power of open source**

**End-to-end optimization**

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## Part 4: Summary and discussion

- 4.1. Strength of each approach
- 4.2. Challenges and fundamental difficulties

# Why is achieving this goal important?

- It can **unfold a myriad of new speech applications**, such as
  - Meeting minute and summary generation (Video by Microsoft)
  - Human-robot interaction (Video by Hitachi)
  - Communication agent that can understand, respond to and facilitate conversation
  - Conversation analysis and assessment

⋮

# Why is achieving this goal important?

- Application: Meeting minute and summary generation (Video by Microsoft)

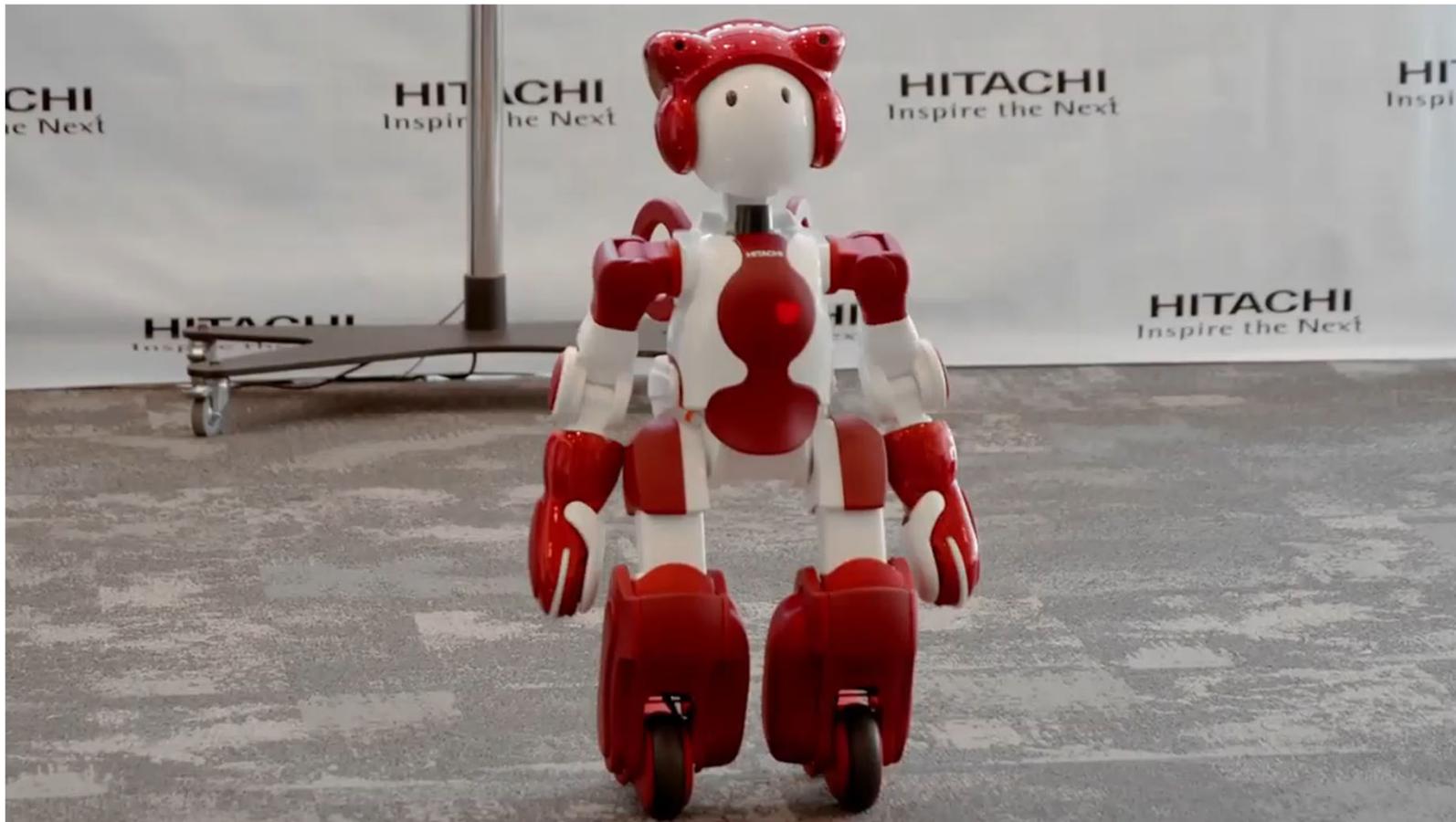


T. Yoshioka et al., Advances in Online Audio-Visual Meeting Transcription, ASRU, 2019

<https://www.youtube.com/watch?v=ddb3ZgAp9TA>

# Why is achieving this goal important?

- Application: Human-robot interaction (Video by Hitachi)



<https://www.youtube.com/watch?v=lPhA31DAO28>

# Why is achieving this goal important?

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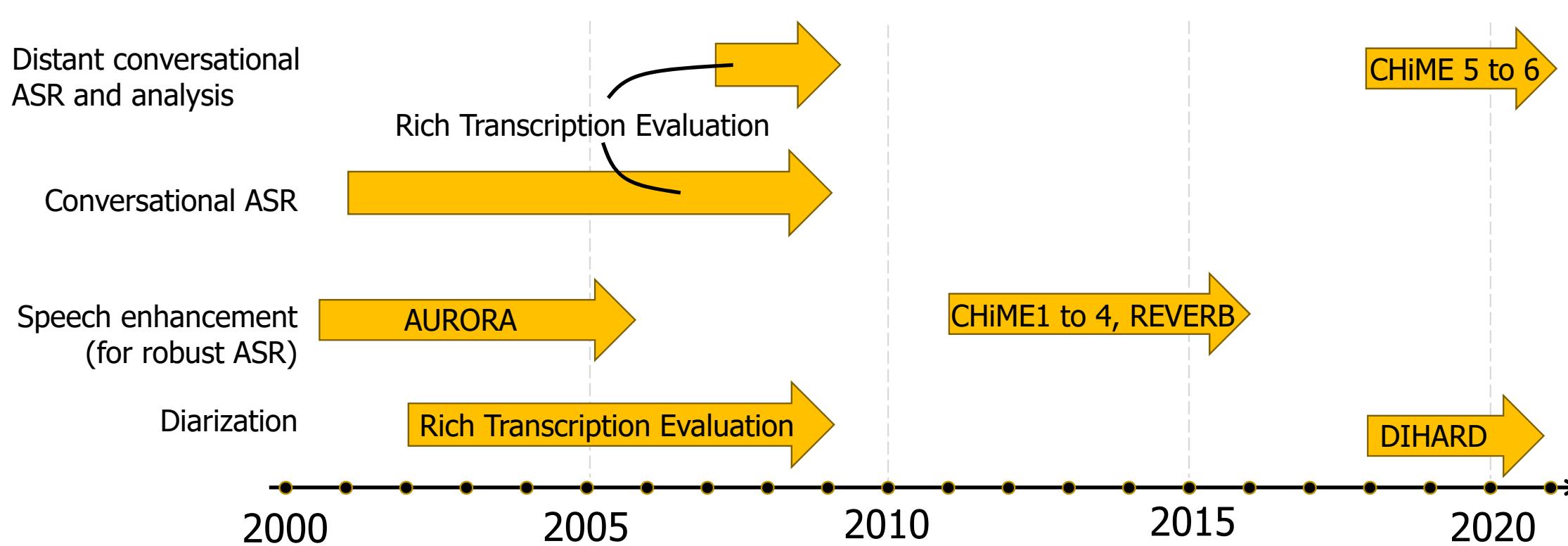
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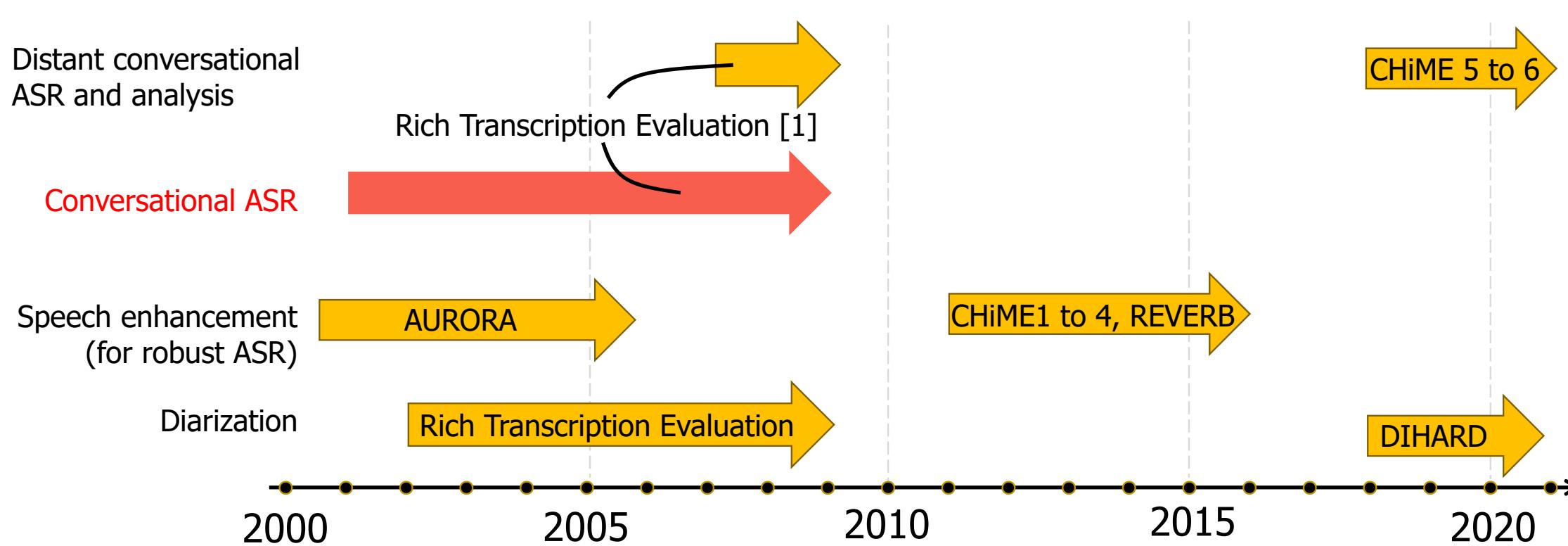
# History of distant conversational ASR and analysis research

## Tasks



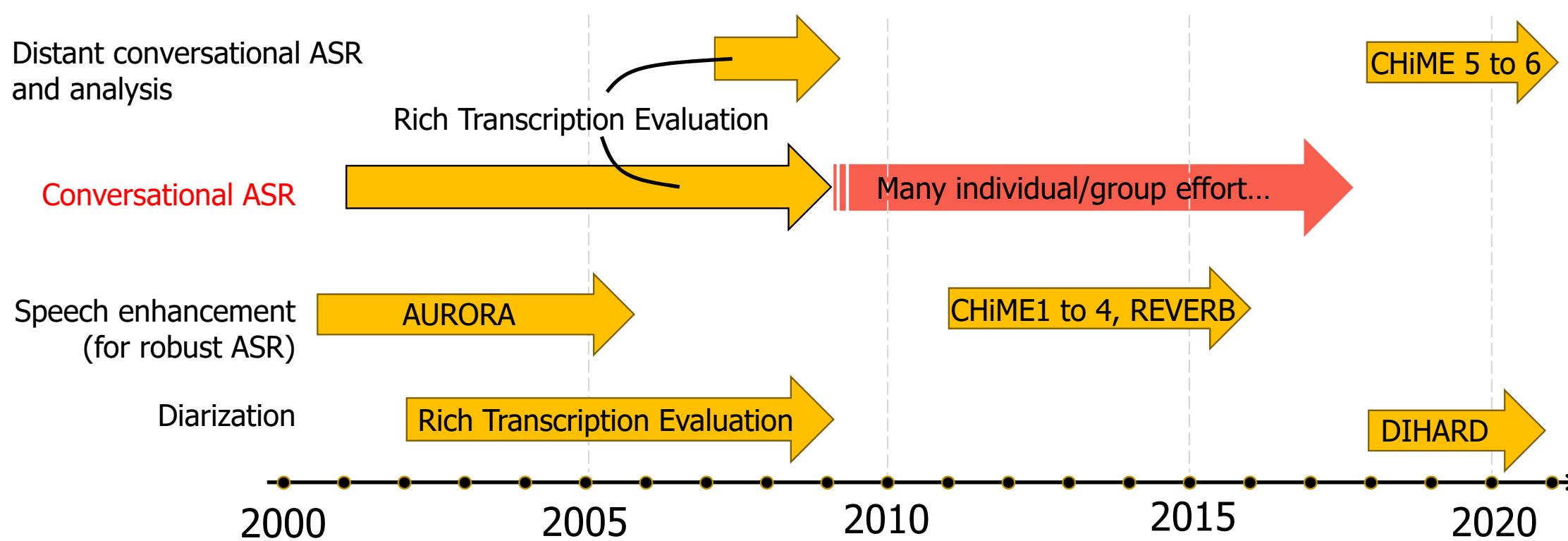
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## Tasks



- [1] J. G. Fiscus et al., The Rich Transcription 2007 Meeting Recognition, Multimodal Technologies for Perception of Humans, 2007  
[2] A Janin et al., The ICSI meeting corpus, ICASSP, 2003  
[3] J. Carletta et al., The AMI meeting corpus., MLMI, 2005

# History of distant conversational ASR and analysis research

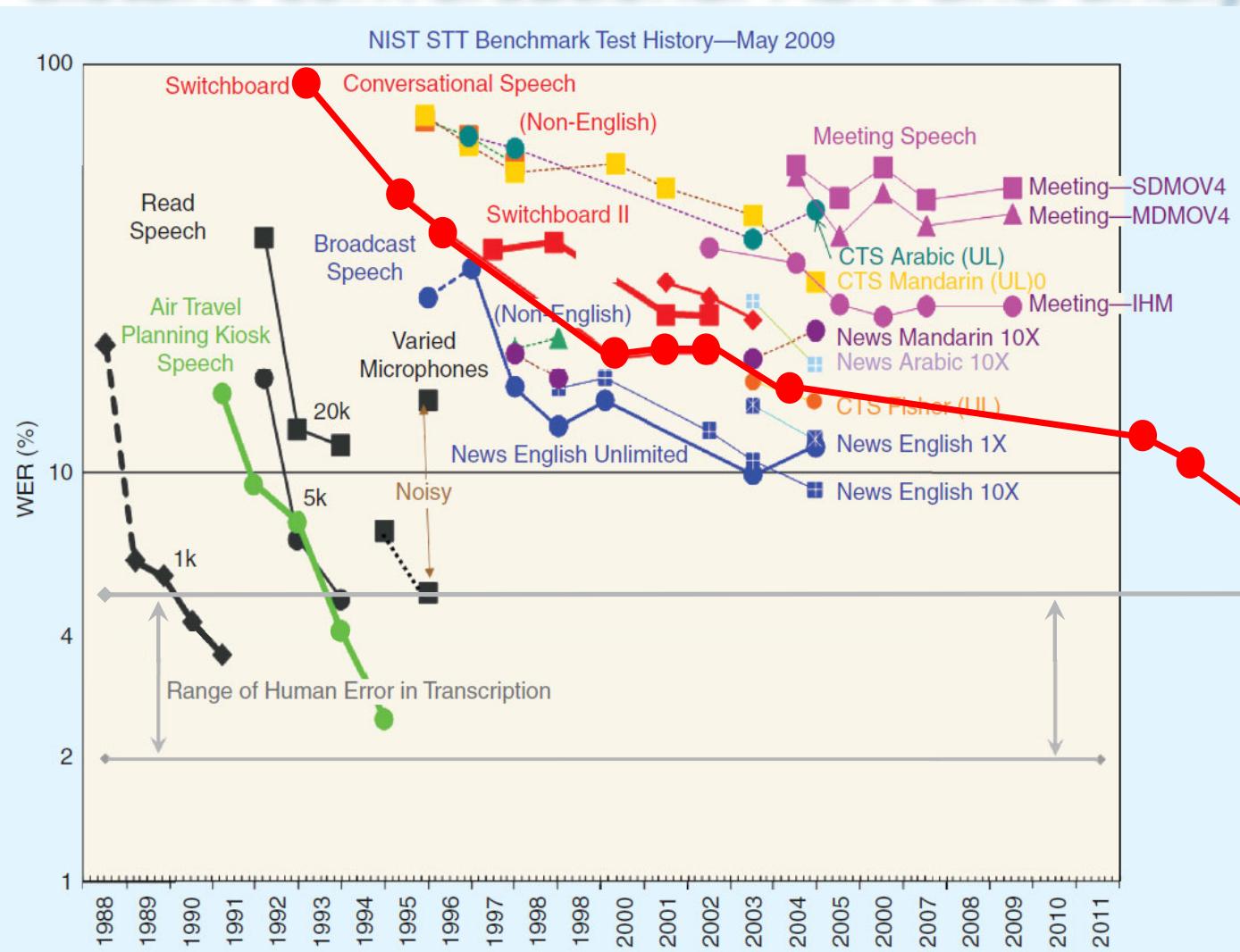


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A. Stolcke et al., Comparing human and machine errors in conversational speech transcription, Interspeech, 2017  
G. Saon et al., English Conversational Telephone Speech Recognition by Humans and Machines, Interspeech, 2017

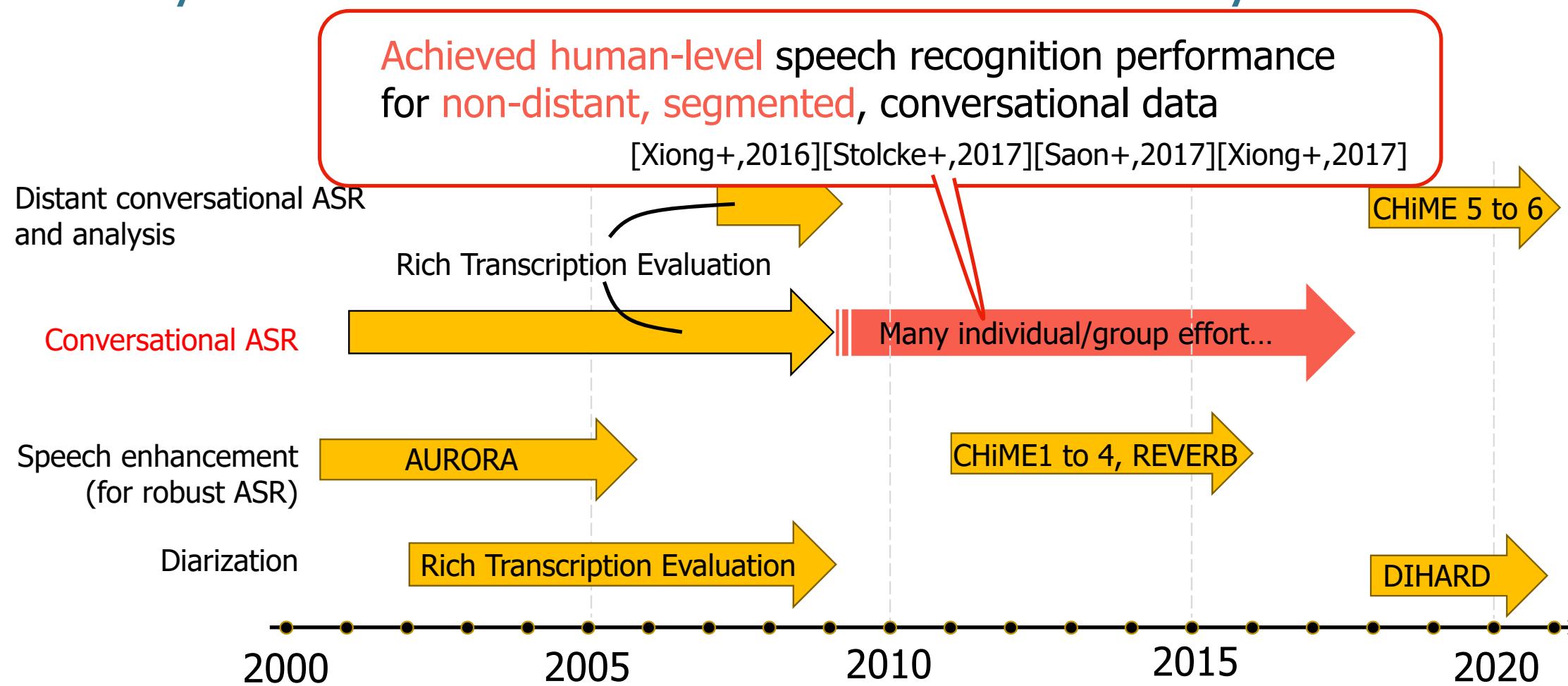
# History of distant conversational ASR and analysis research

(Fig. from [1])



[1] Xiaodong He, Li Deng, Speech Recognition, Machine Translation, and Speech Translation —A Unified Discriminative Learning Paradigm, IEEE SP Magazine, 2011

# History of distant conversational ASR and analysis research

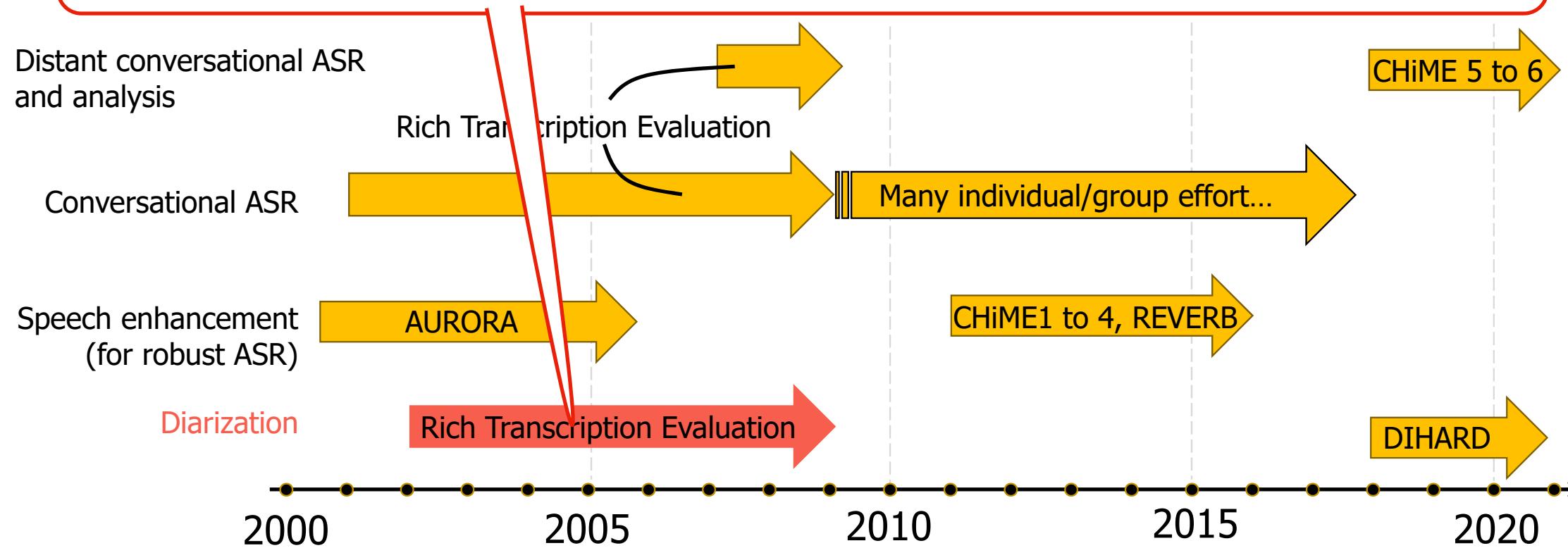


W. Xiong et al., Achieving Human Parity in Conversational Speech Recognition, arXiv, 2016  
W. Xiong et al., The Microsoft 2017 Conversational Speech Recognition System, arXiv, 2017

A. Stolcke et al., Comparing human and machine errors in conversational speech transcription, Interspeech, 2017  
G. Saon et al., English Conversational Telephone Speech Recognition by Humans and Machines, Interspeech, 2017

# History of distant conversational ASR and analysis research

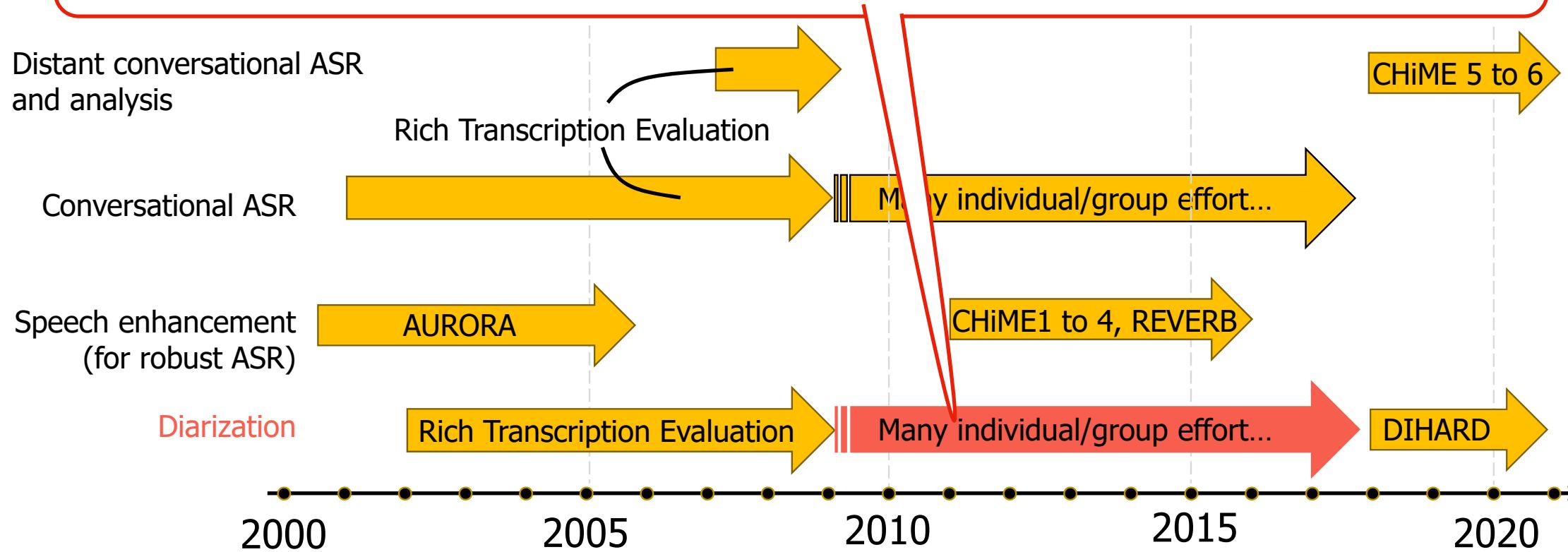
Developed basis of the current-state-of-the-art diarization system,  
i.e., speaker embedding + clustering [Anguera+, 2012]



X. Anguera et al., Speaker Diarization: A Review of Recent Research, IEEE TASLP, 2012,

# History of distant conversational ASR and analysis research

Solid improvement on embeddings, and clustering [Anguera+, 2012]

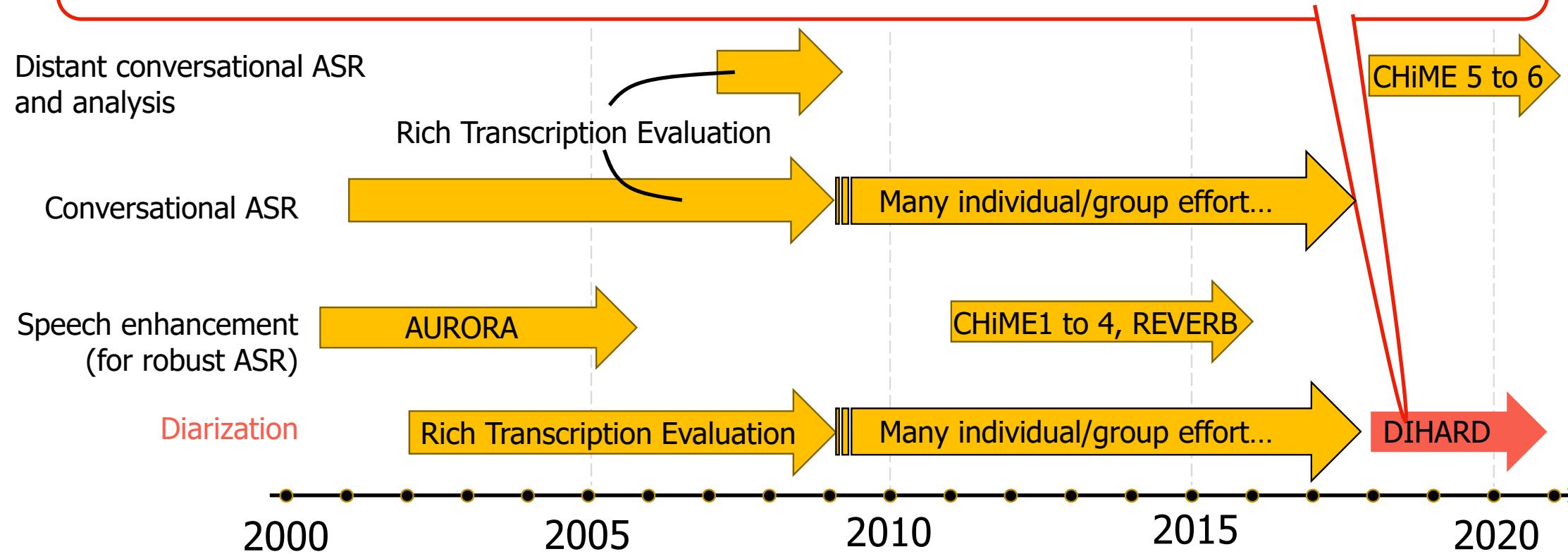


X. Anguera et al., Speaker Diarization: A Review of Recent Research, IEEE TASLP, 2012,

# History of distant conversational ASR and analysis research

- Speaker embedding clustering
- End-to-end diarization

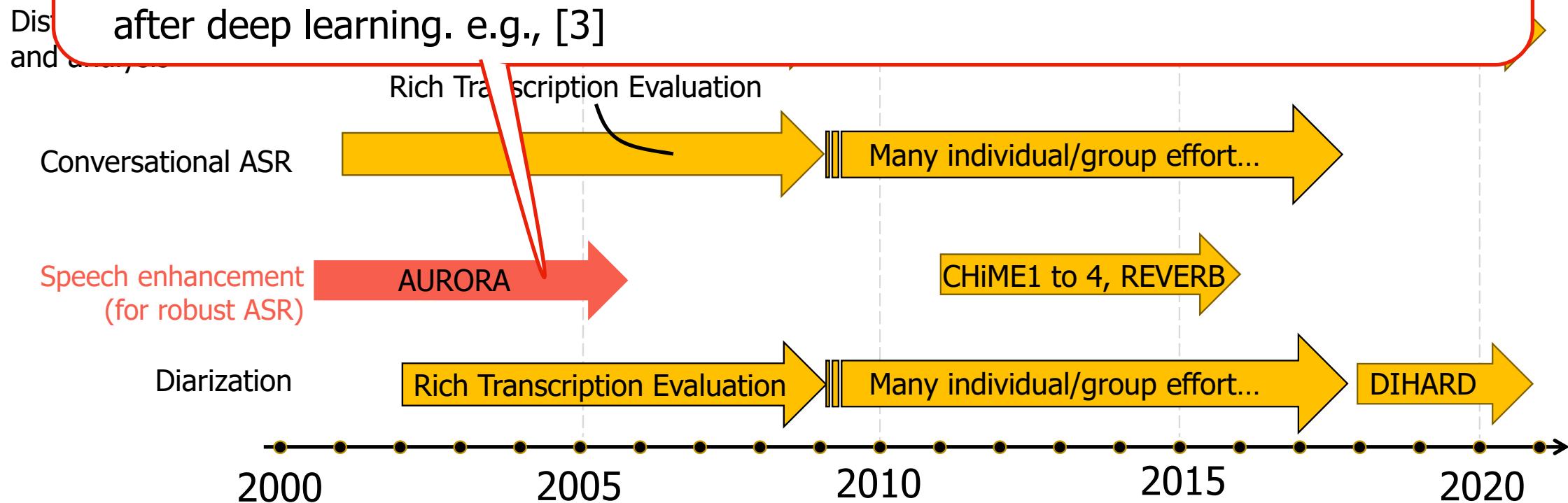
(Details will be covered in Part 3.1!)



# History of distant conversational ASR and analysis research

(Single-channel read-speech corpora for noise-robust ASR)

- Developed robust features, and feature enhancement algorithms [1][2]
- Most 1ch speech enhancement technologies became obsolete after deep learning. e.g., [3]



[1] T. Virtanen et al., Techniques for Noise Robustness in Automatic Speech Recognition, Wiley, 2012

[2] J. Li et al., An Overview of Noise-Robust Automatic Speech Recognition, IEEE TASLP, 2014

[3] M. Seltzer et al., An investigation of deep neural networks for noise robust speech recognition, ICASSP, 2013

# History of distant conversational ASR and analysis research

(Multichannel read-speech corpora for noise/reverb-robust ASR)

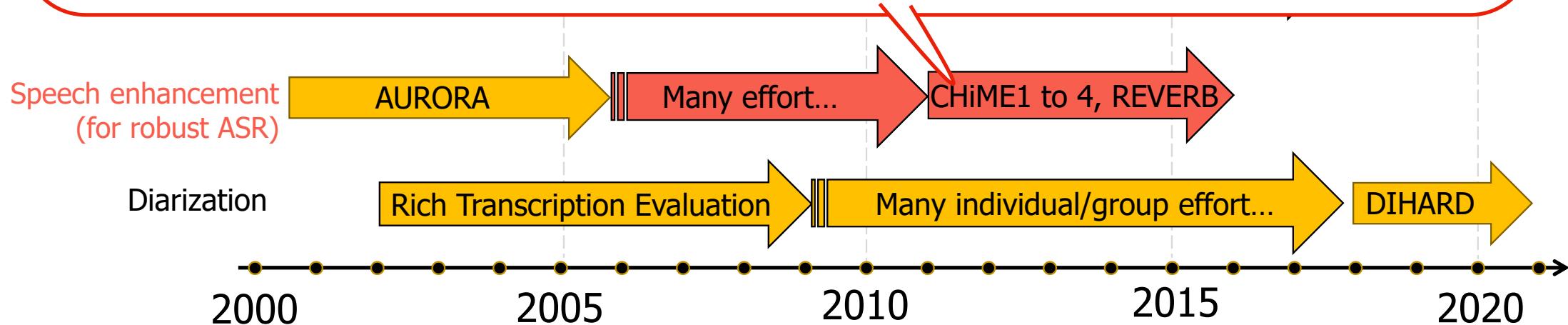
- Multichannel enhancement worked nicely with deep-learning ASR

[Yoshioka+, 2015][Heymann+, 2016][Delcroix+, 2015]

Dis... and - Achieved **super-human recognition performance** for **noisy** data [Amodei+, 2015].

- Techniques developed/used in these challenges are still employed  
as a part of the state-of-the-art systems in today's challenge systems.

Ch... (will be introduced in Part 2!)



T. Yoshioka et al., The NTT CHiME-3 system: Advances in speech enhancement and recognition for mobile multi-microphone devices, ASRU, 2015

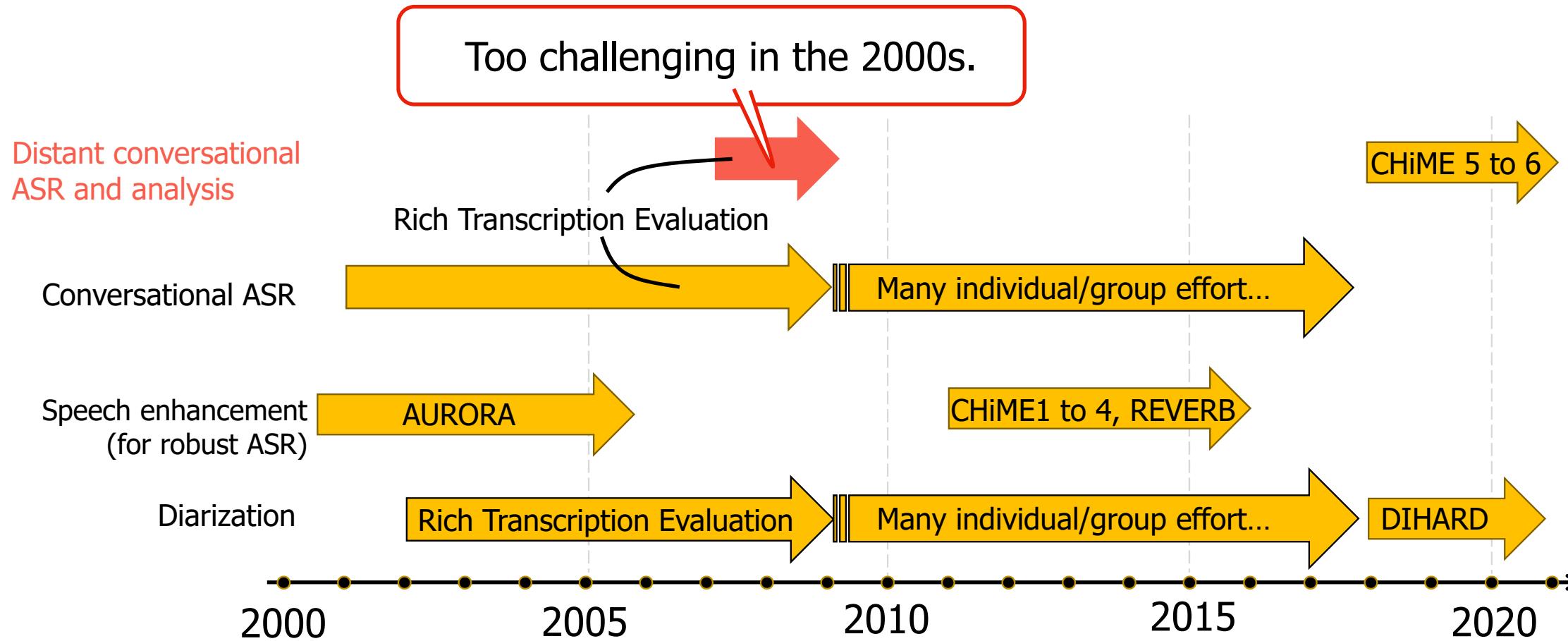
J. Heymann et al., Neural network based spectral mask estimation for acoustic beamforming, ICASSP, 2016

M. Delcroix et al., Strategies for distant speech recognition reverberant environments, EURASIP Journal on Advances in SP, 2015

D. Amodei, Deep Speech 2 : End-to-End Speech Recognition in English and Mandarin, arXiv, 2015

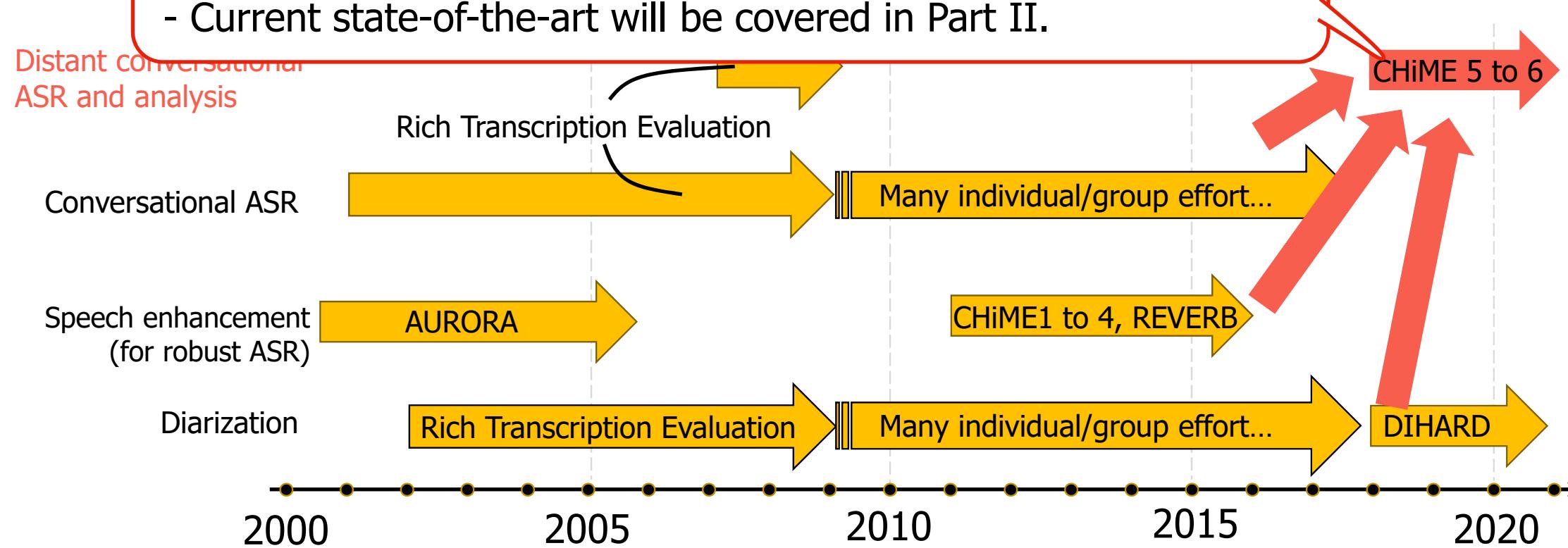
K. Kumatani et al. Microphone Array Processing for Distant Speech Recognition: From Close-Talking Microphones to Far-Field Sensors, IEEE Signal Processing Magazine, 2012

# History of distant conversational ASR and analysis research



# History of distant conversational ASR and analysis research

- Now that we have human-level/super-human modules, we should be ready to tackle distant conversational ASR again!
- Current state-of-the-art will be covered in Part II.



# Agenda

## Part 1: Introduction

- 1.1. What is distant conversational ASR and analysis?
- 1.2. Why is it difficult?
- 1.3. Why is it important?
- 1.4. Its research history
- 1.5. Typical systems for distant conversational ASR and analysis**

## Part 2: Current state-of-the-art systems

- 2.1. Descriptions of the techniques
- 2.2. Reproducible baselines

## Part 3: A new research trend: Jointly optimal systems

- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$      ( $x$ : other functionalities)
- 3.3. ASR +  $x$

## Part 4: Summary and discussion

- 4.1. Strength of each approach
- 4.2. Challenges and fundamental difficulties

# Typical systems for distant conversational ASR and analysis

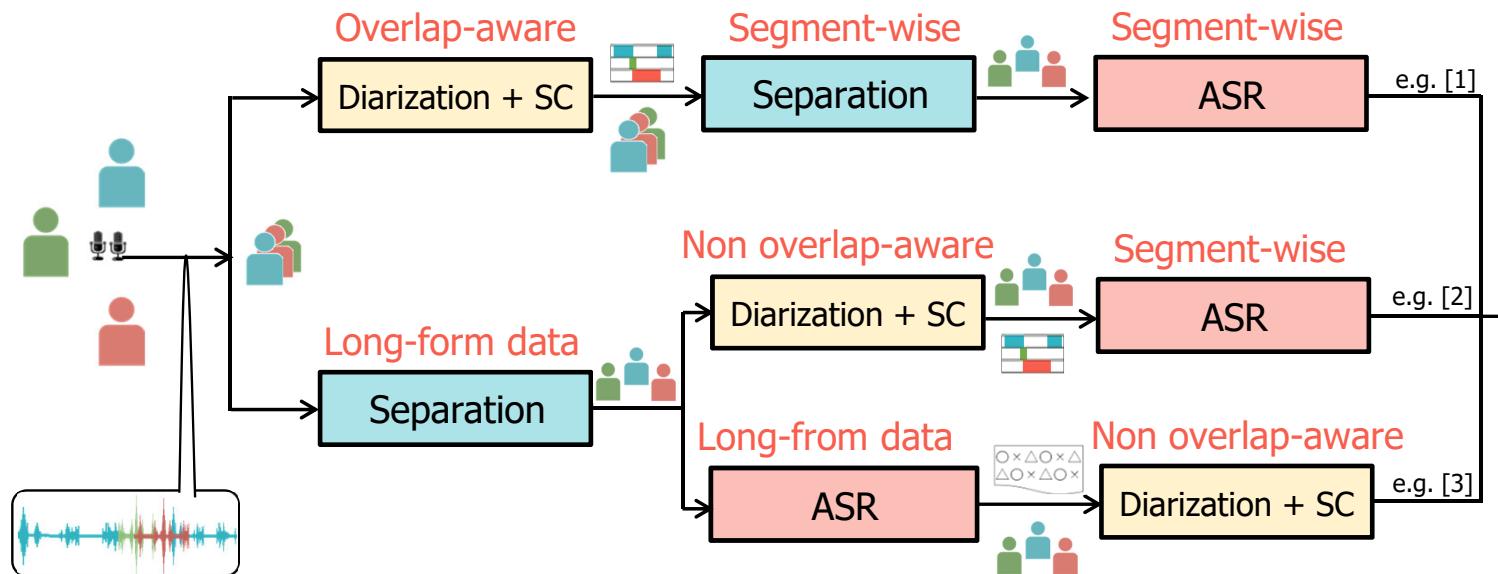
- Several different styles, having different characteristics

Non-overlapped speech signal: 

Diarization results: 

Overlapped speech signal: 

Transcription: 



Speaker-attributed transcription

Spk ID	Start	Duration	Transcription
A	0.5s	1.5s	How are you
B	1.0s	2.0s	Hi I'm fine
			:

[1] I. Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

[2] T. Hori *et al.*, "Low-Latency Real-Time Meeting Recognition and Understanding Using Distant Microphones and Omni-Directional Camera," in IEEE TASLP, 2012

[3] T. Yoshioka et al., "Advances in Online Audio-Visual Meeting Transcription", ASRU, 2019

# Typical systems for distant conversational ASR and analysis

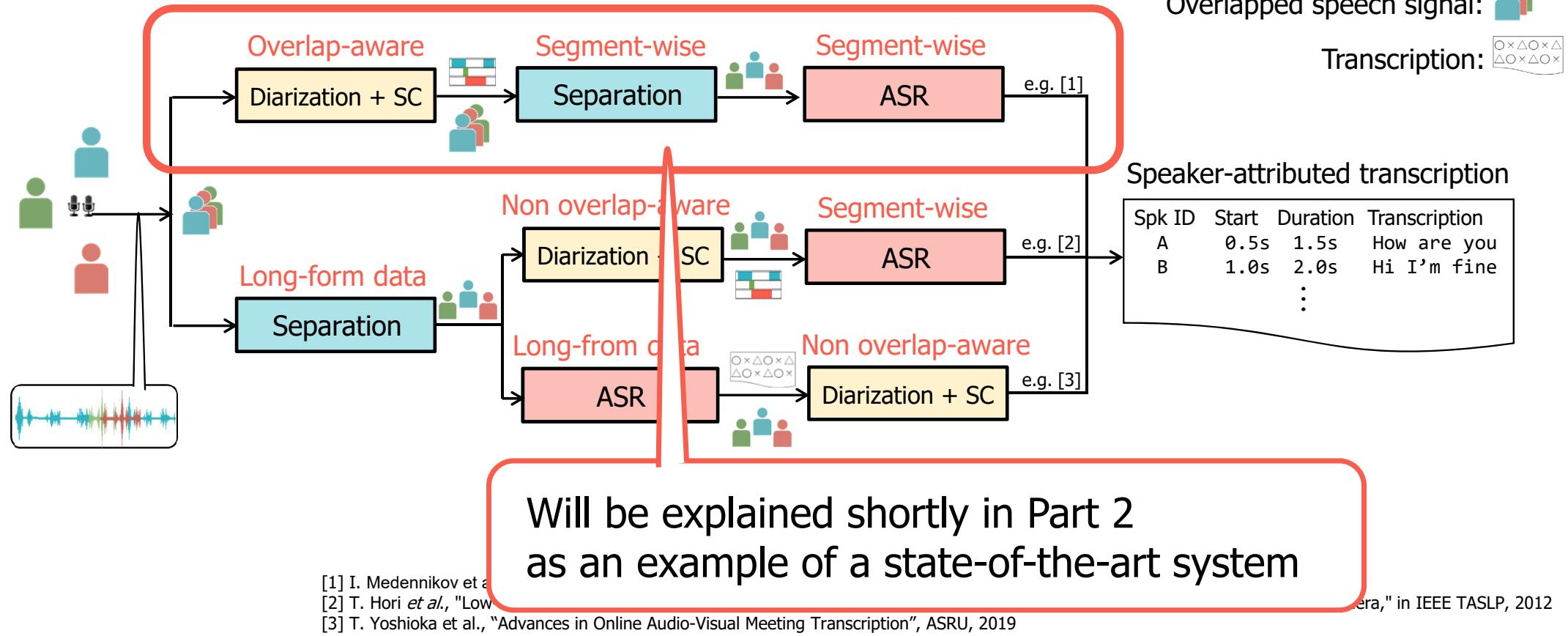
- Several different styles, having different characteristics

Non-overlapped speech signal: 

Diarization results: 

Overlapped speech signal: 

Transcription: 



# Things we cover😊 and don't cover😢 in this tutorial

- 😊 State-of-the-art multichannel enhancement.
  - 😊 Neural speech enhancement (mainly separation) and its combination with other modules such as ASR.
  - 😊 State-of-the-art diarization systems
  - 😊 Combination with other modules
  - 😊 Multimodal approaches, e.g., audio-visual approaches [Afouras+, 2018]
  - 😊 Combination with other modules
  - 😢 Systems that perform only speaker counting, e.g. [Stöter+, 2019]
- 
- A Venn diagram illustrating the overlap of four speech processing modules. There are four overlapping circles: 'Speech enhancement' (light blue), 'Diarization' (yellow), 'Speaker counting' (green), and 'ASR' (pink). The regions of overlap between any two or more circles are shaded in a darker shade of the respective color, representing the combination of those modules. The regions where only one module is present are left white.

- 😊 State-of-the-art system
- 😊 End-to-end ASR combined with other modules
- 😢 Language model-based approaches, i.e., language model for conversation [Mikolov+, 2012].

T. Mikolov et al., Context dependent recurrent neural network language model, SLT, 2012  
F. Stöter et al., CountNet: Estimating the Number of Concurrent Speakers Using Supervised Learning Speaker Count Estimation, TASLP, 2019  
T. Afouras et al., Deep audio-visual speech recognition, IEEE Trans. PAMI, 2018

## 2. Current state-of-the-art systems

## 2. Current state-of-the-art systems

- 2.1. Descriptions of the techniques
- 2.2. Reproducible baselines

# Distant conversational speech recognition and analysis

- We have to solve all of **enhancement**, **diarization**, and **ASR**  
(We skip the discussion about speaker counting in this part)
- We mainly explain state-of-the-art techniques used in the **CHiME-6** systems because of most recent challenge activities
- Note that there are a lot of excellent benchmark activities including Rich transcriptions, AMI, LibriCSS, etc.
  - Many of the techniques introduced here were developed in these previous activities
  - Their effectiveness are verified in the CHiME-6 scenario

# CHiME-6 Challenge



## Dinner party scenario

- Recorded in people's actual homes
- Parties of 4 - typically, two hosts and two guests
- Collection of 20 parties each lasting 2 to 3 hours

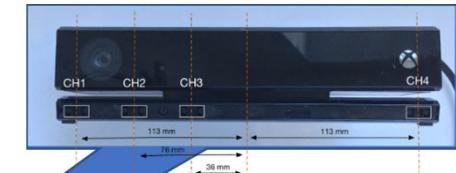


# CHiME-6 Challenge



## Dinner party scenario

- Recorded in people's actual homes
- Parties of 4 - typically, two hosts and two guests
- Collection of 20 parties each lasting 2 to 3 hours



- Six separate Microsoft Kinect devices
- Two Kinects per living area (kitchen, dining, sitting)
- Channel: 6 x 4 audio

# CHiME-6 Challenge



## Dinner party scenario

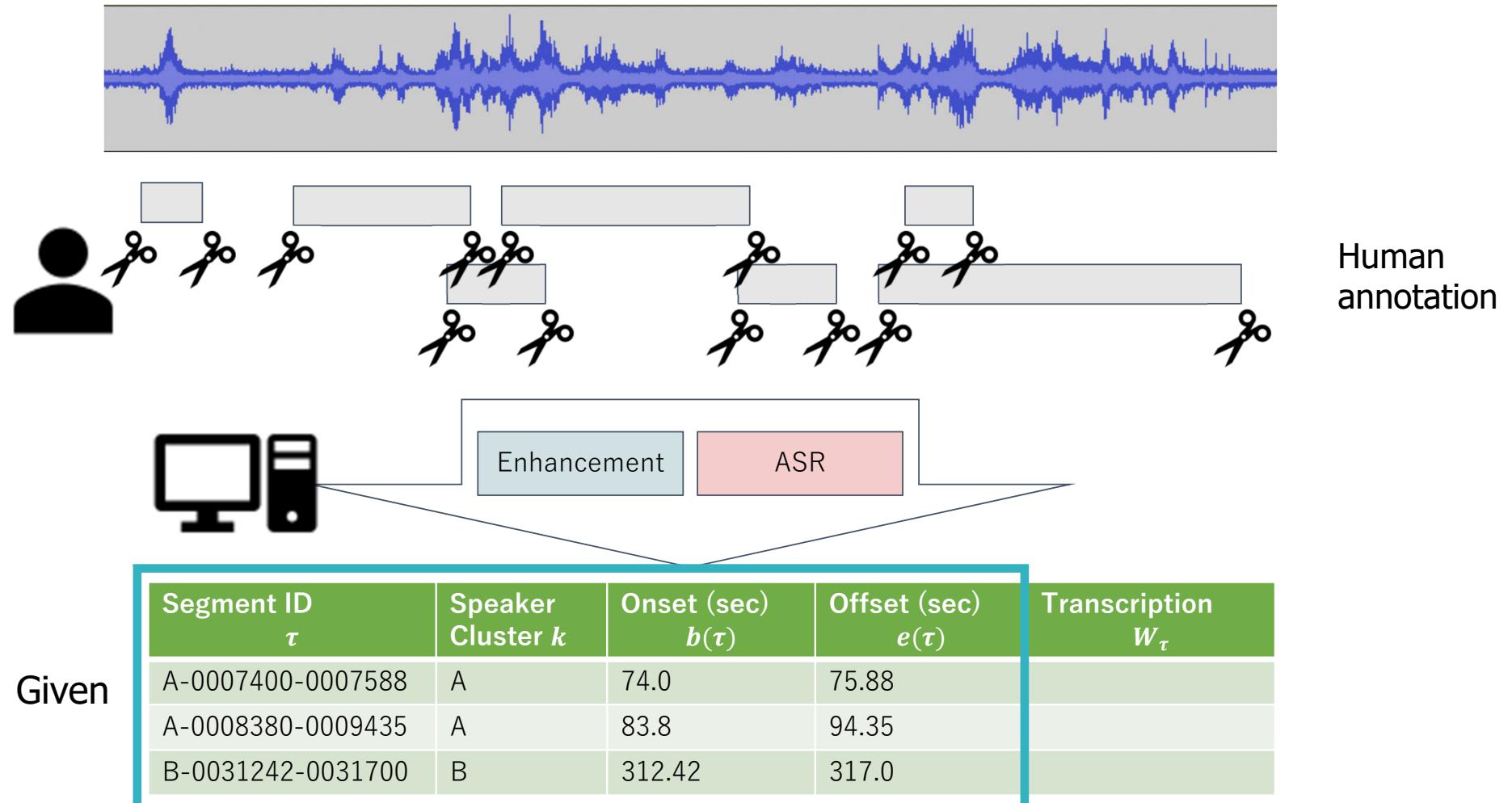
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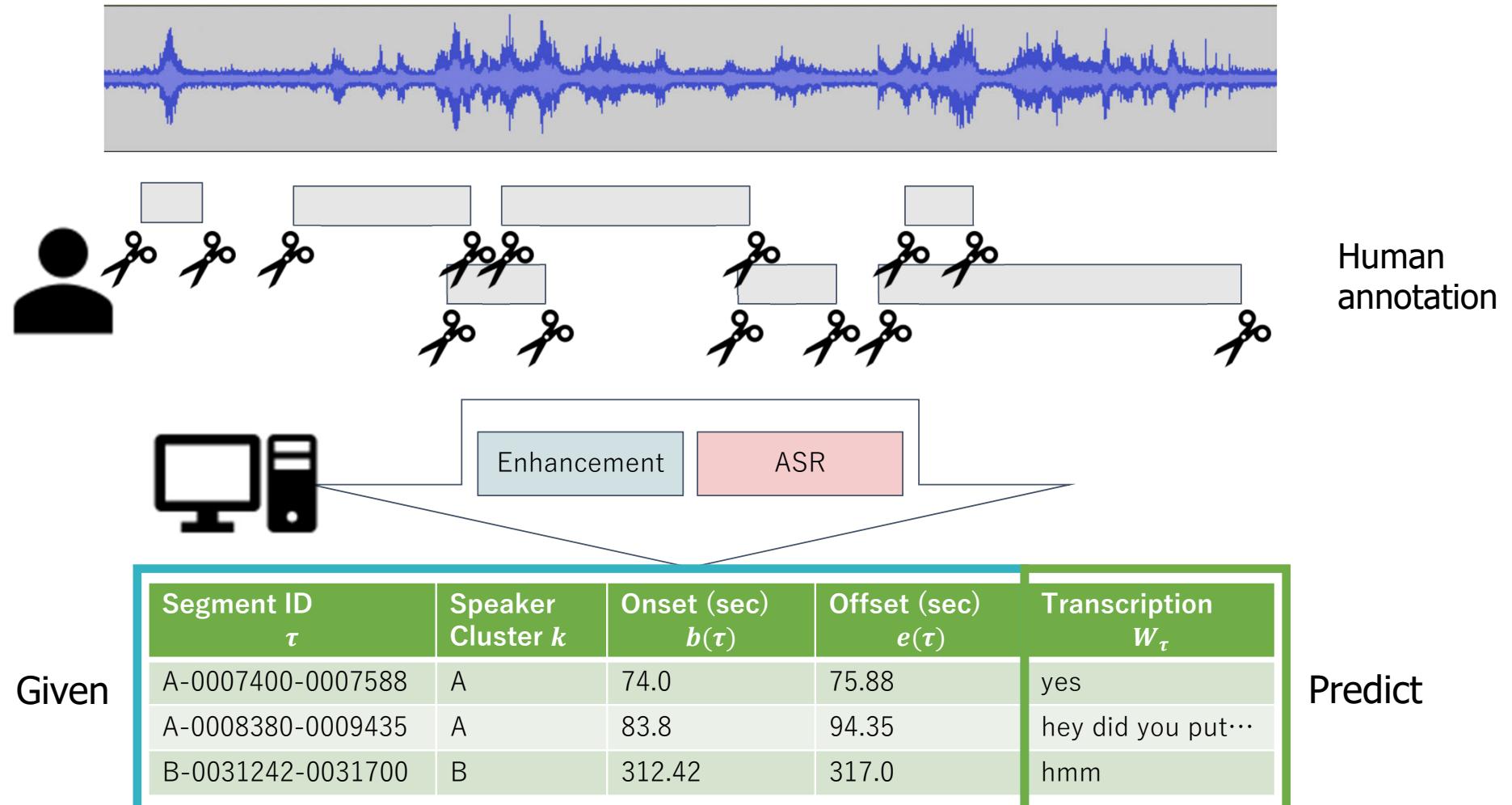
- Binaural in-ear microphones (worn mic)
- Relatively clean conditions



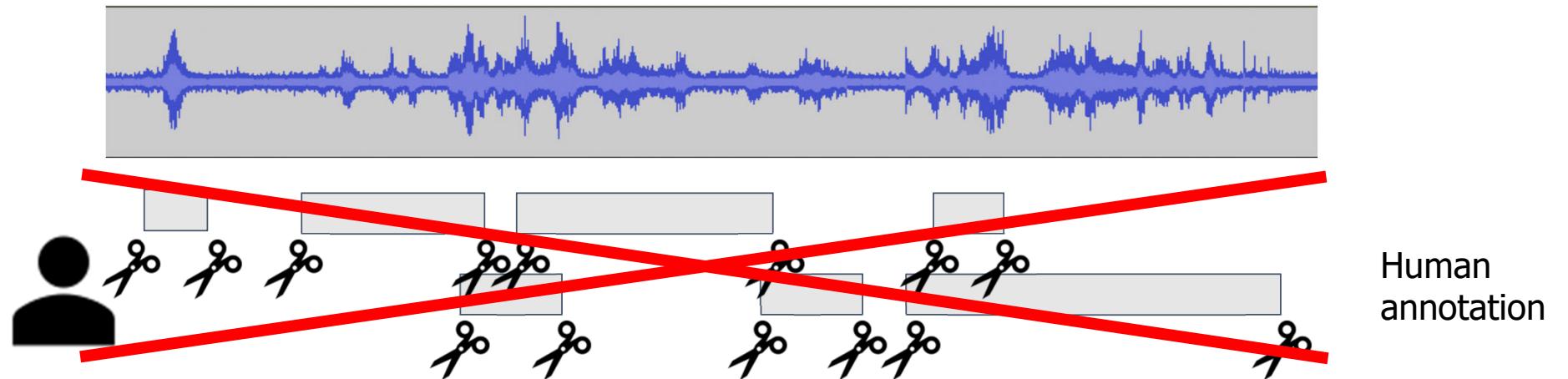
# Classical ASR tasks (including previous CHiME challenges)



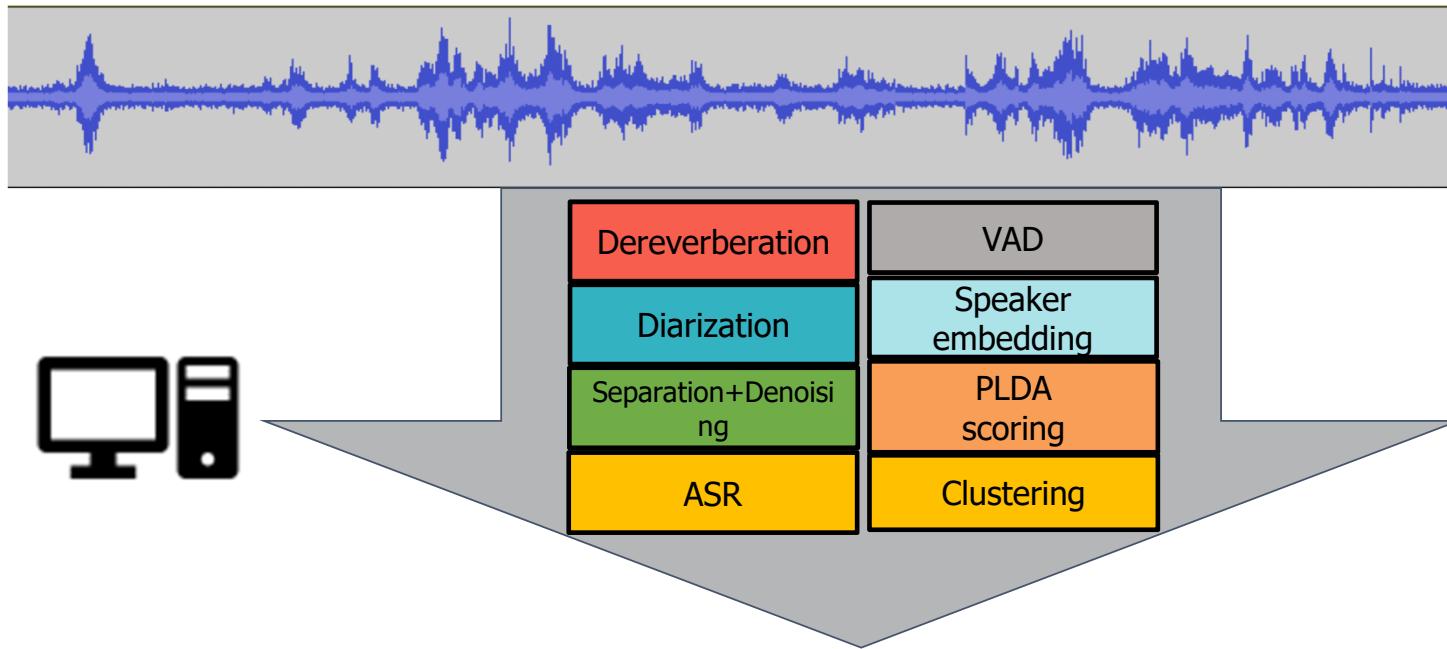
# Classical ASR tasks (including previous CHiME challenges)



# Distant conversational speech recognition and analysis



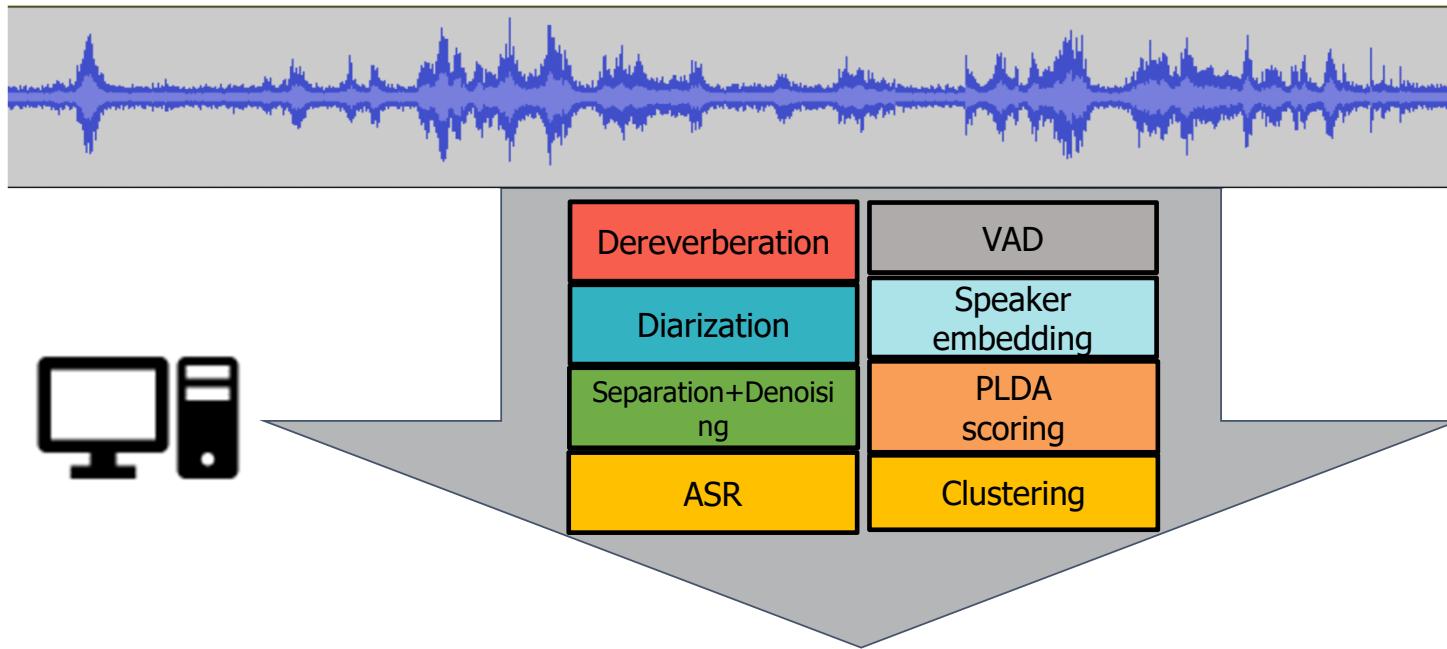
# Distant conversational speech recognition and analysis → CHiME-6 track 2



Without any given information  
(except for #speakers in CHiME-6)

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$

# Distant conversational speech recognition and analysis → CHiME-6 track 2



Without any given information  
(except for #speakers in CHiME-6)

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
A-0007400-0007588	A	74.0	75.88	yes
A-0008380-0009435	A	83.8	94.35	hey did you put...
B-0031242-0031700	B	312.42	317.0	hmm

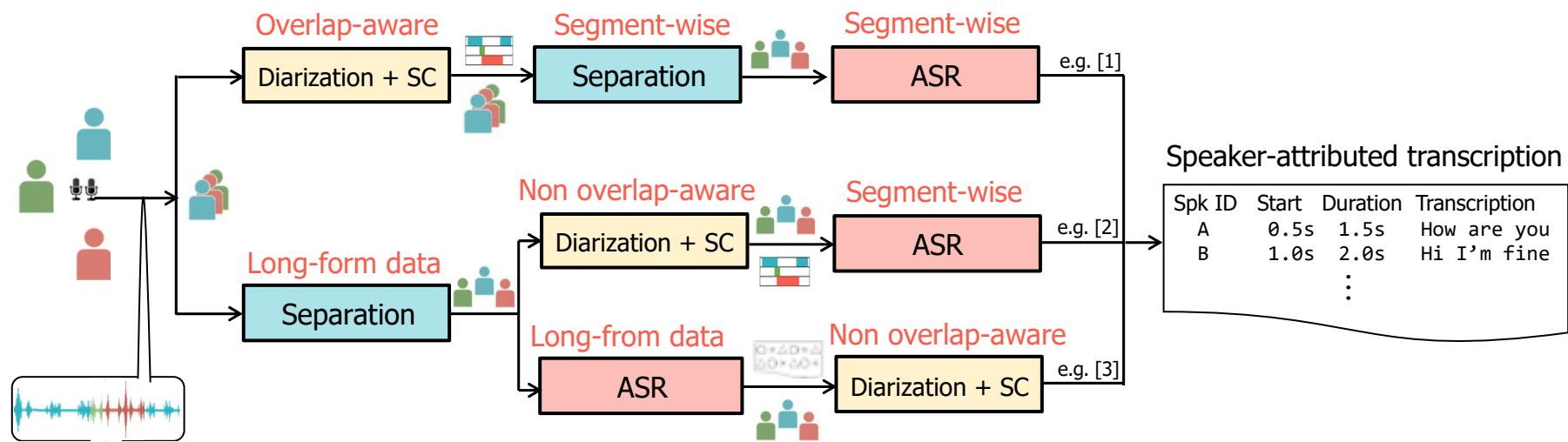
Predict everything

## 2. Current state-of-the-art systems

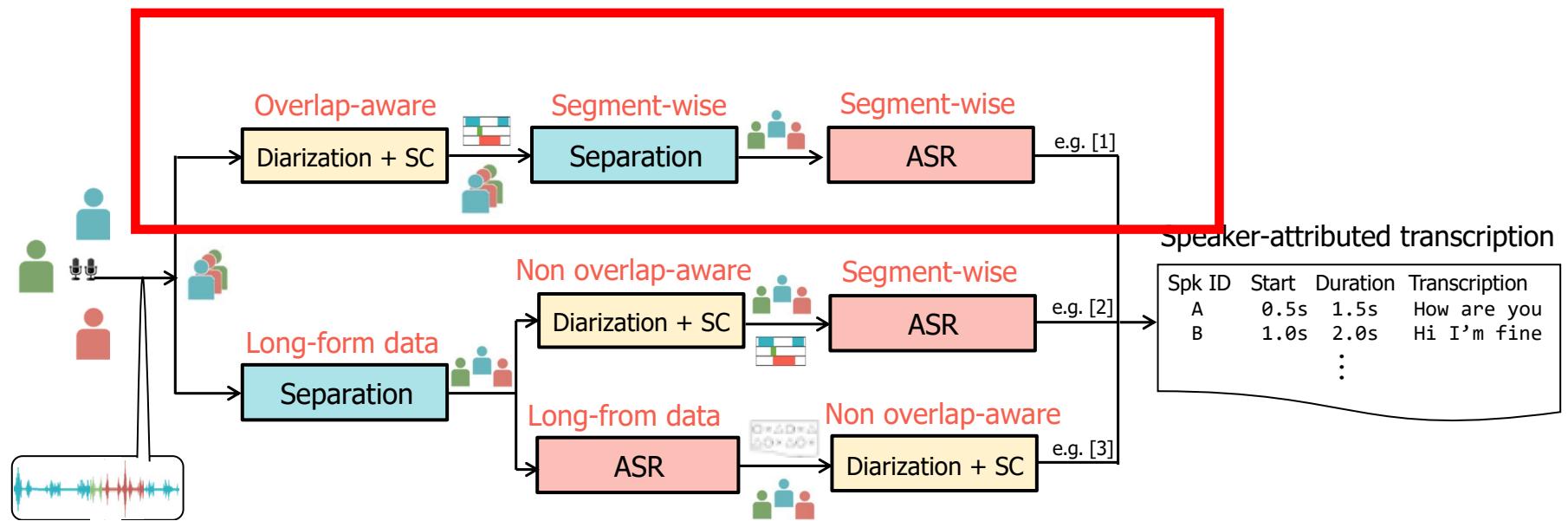
### 2.1. Descriptions of the techniques

### 2.2. Reproducible baselines

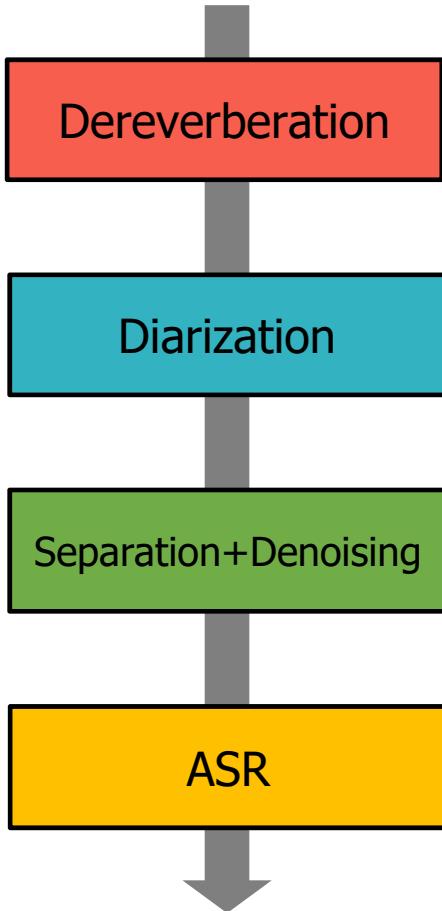
# CHiME-6 pipeline



# CHiME-6 pipeline



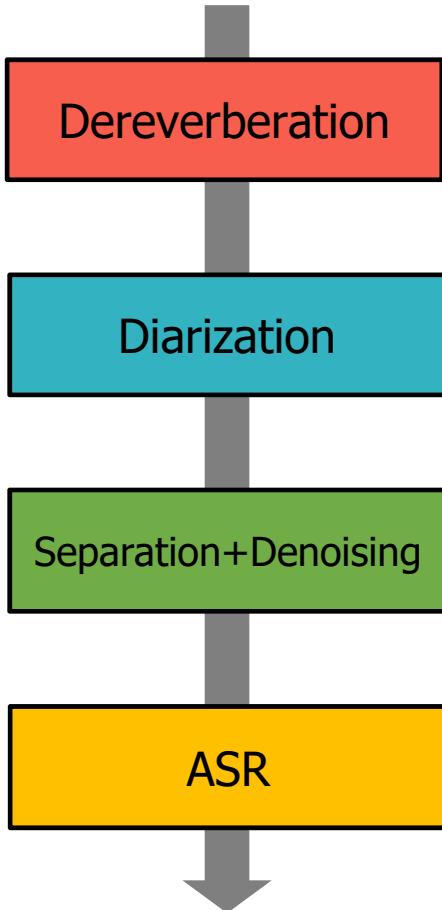
# CHiME-6 pipeline



The process gradually fills out the missing RTTM information

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$

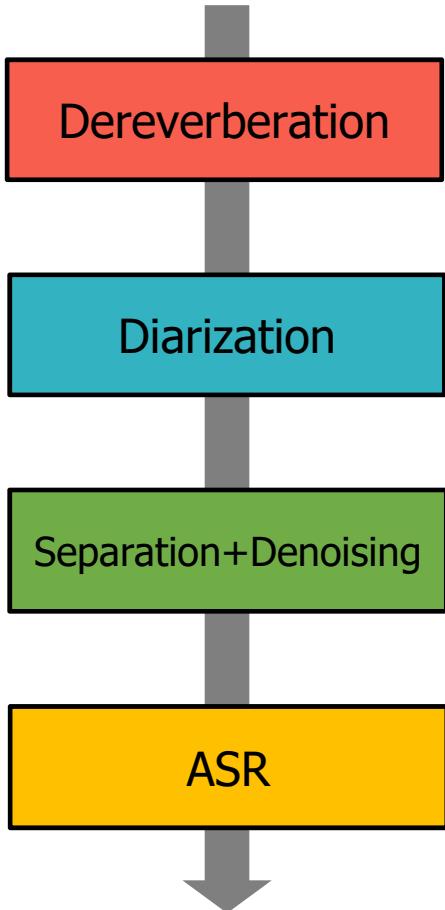
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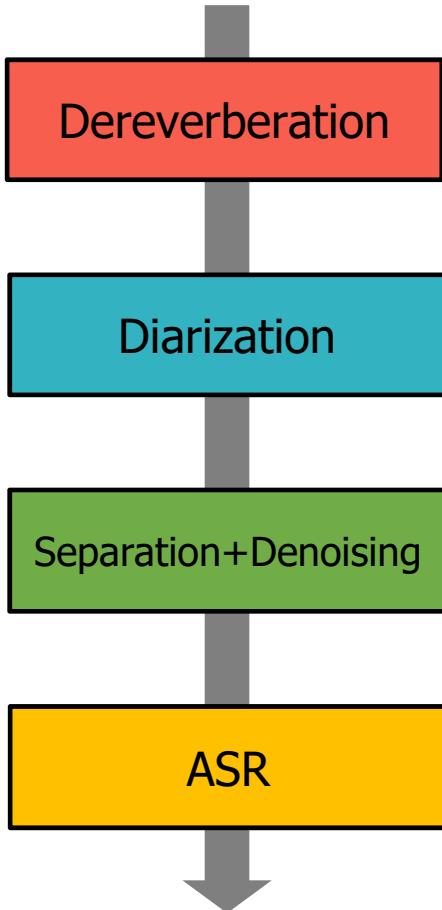
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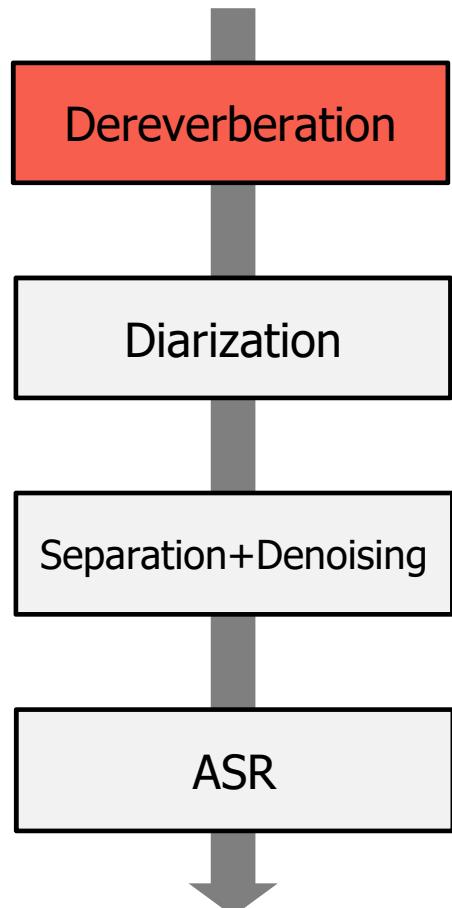
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# Speech enhancement



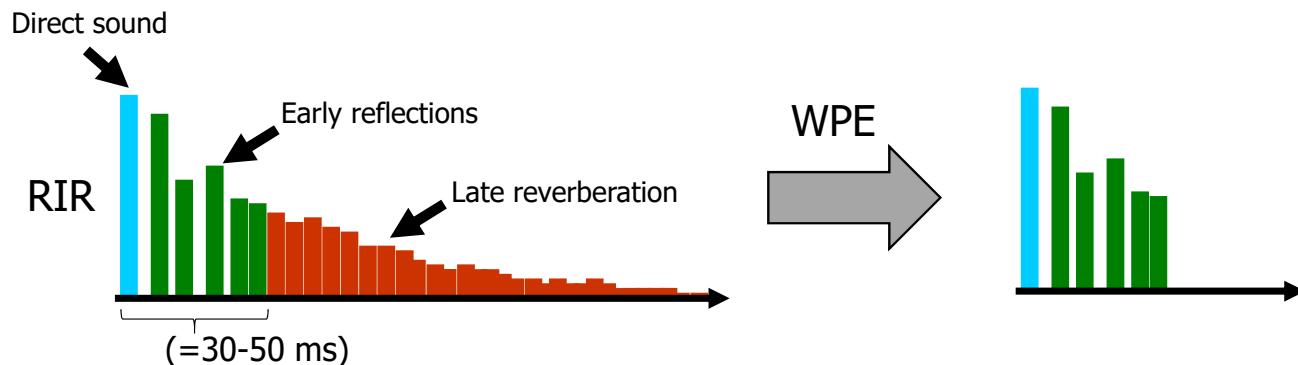
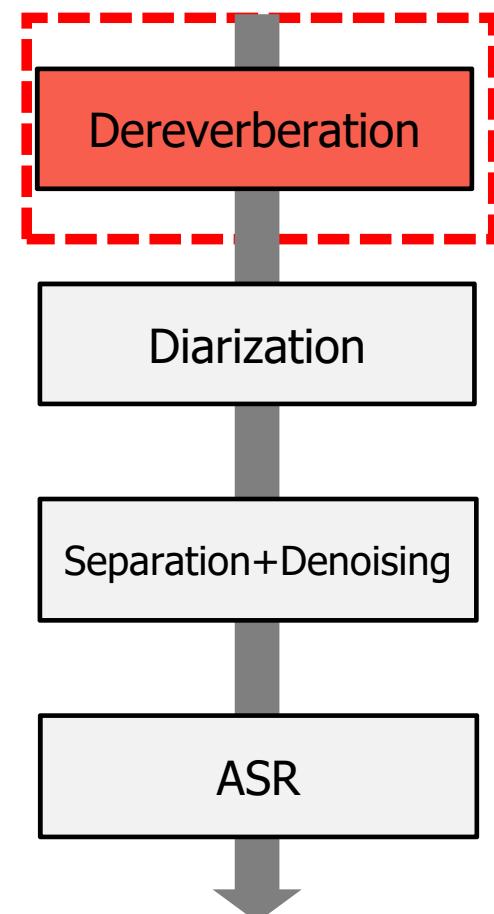
Haeb-Umbach and Nakatani, "Microphone array signal processing and deep learning for speech enhancement – strong together," Interspeech-Tutorial, 2019

# Speech enhancement (Dereverberation)

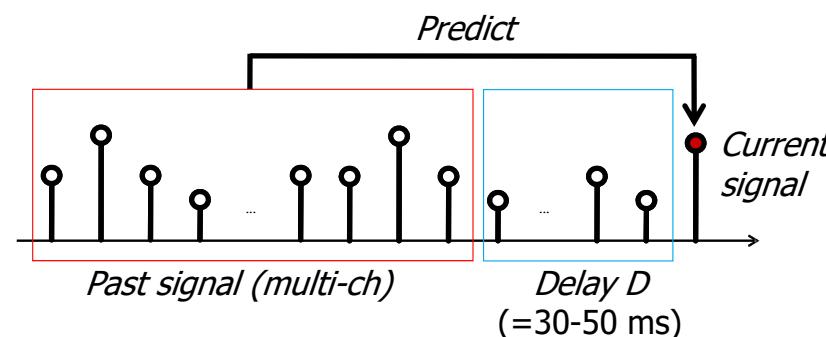
[Nakatani+, 2010][Yoshioka, 2012]

## Weighted Prediction Error (WPE)-based dereverberation

-Effect: Reduce late reverberation harmful for ASR, separation, diarization



-How to estimate and reduce late reverberation

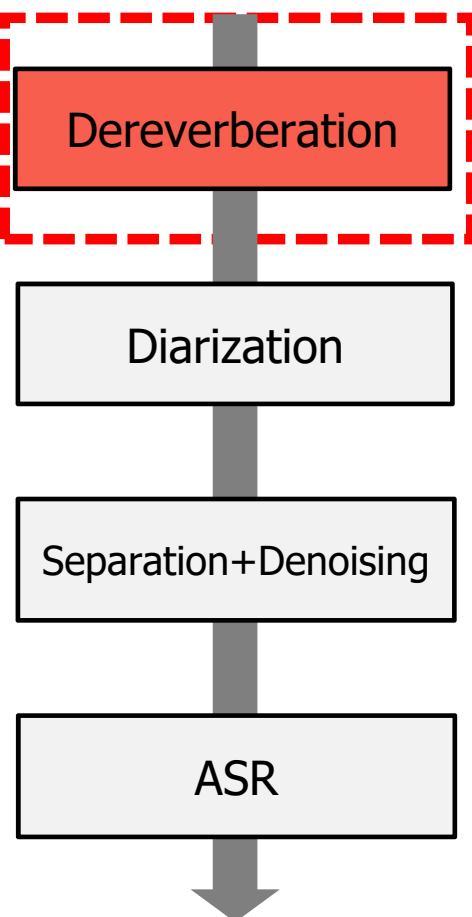


- Predictable component → late reverberation
- Unpredictable component → Direct sound

T. Nakatani et al., Speech Dereverberation Based on Variance-Normalized Delayed Linear Prediction, TASLP, 2010,  
T. Yoshioka et al., Generalization of Multi-Channel Linear Prediction Methods for Blind MIMO Impulse Response Shortening, TASLP, 2012

# Speech enhancement (Dereverberation) [Nakatani+, 2010]

## Weighted Prediction Error (WPE)-based dereverberation



- How do we estimate WPE filter  $\hat{\mathbf{W}}$  from  $M$ -channel observed signal  $\mathbf{y}_{t,f} = \{y_{m,t,f}\}_{m=1}^M \in \mathbb{C}^M$ ?  
→ Minimize the following prediction error with delay  $D$ , i.e., dereverberated signal  $\hat{\mathbf{s}}_{t,f}$  is represented as

$$\hat{\mathbf{s}}_{t,f} = \mathbf{y}_{t,f} - \sum_{\tau=D}^L \hat{\mathbf{W}}_{\tau,f}^H \mathbf{y}_{t-\tau,f}$$

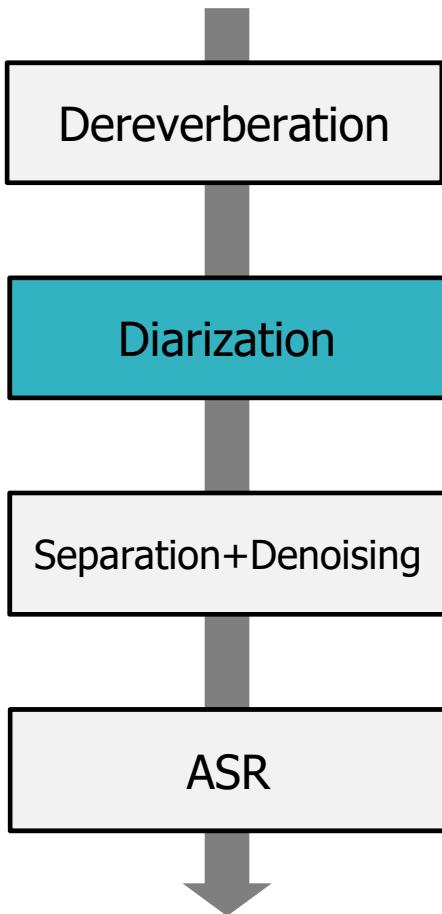
such that  $\hat{\mathbf{s}}_{t,f}$  follows time-varying Gaussian distribution

$$p(\mathbf{s}_{t,f}; \theta) = \mathcal{N}_c(\mathbf{s}_{t,f}; \mathbf{0}, \sigma_{t,f}^2 \mathbf{I})$$

$\sigma_{t,f}^2$ : power spectrum of ideal dereverberated speech

T. Nakatani et al., Speech Dereverberation Based on Variance-Normalized Delayed Linear Prediction, TASLP, 2010,

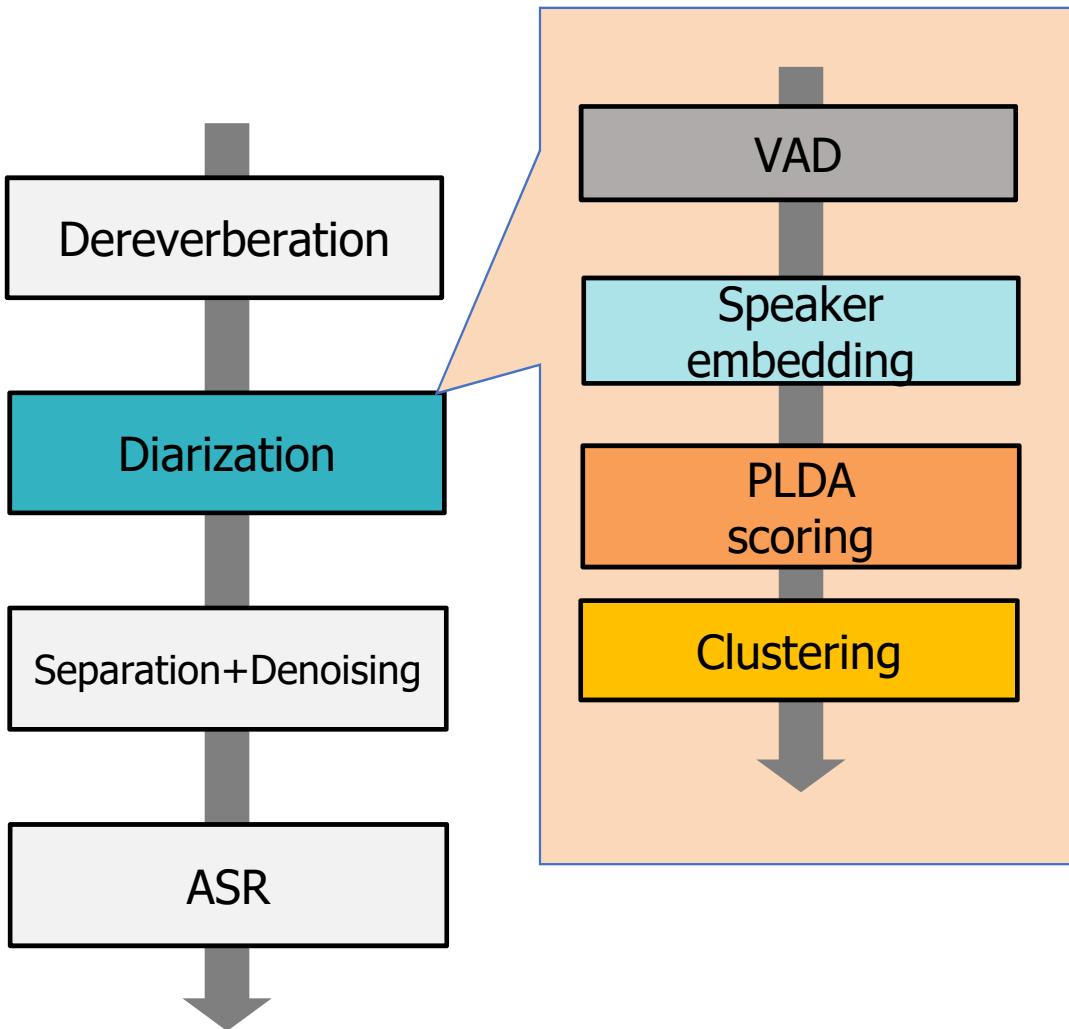
# CHiME-6 pipeline



We will introduce two diarization systems

- 1. Clustering based diarization**
  - Standard system in the speaker diarization studies
- 2. Target-speaker voice activity detection based diarization**
  - Refinement of clustering based diarization
  - Based on CHiME-challenge winning systems
  - Significantly reduce the diarization and ASR errors in the CHiME system

# CHiME-6 pipeline



We will introduce two diarization systems

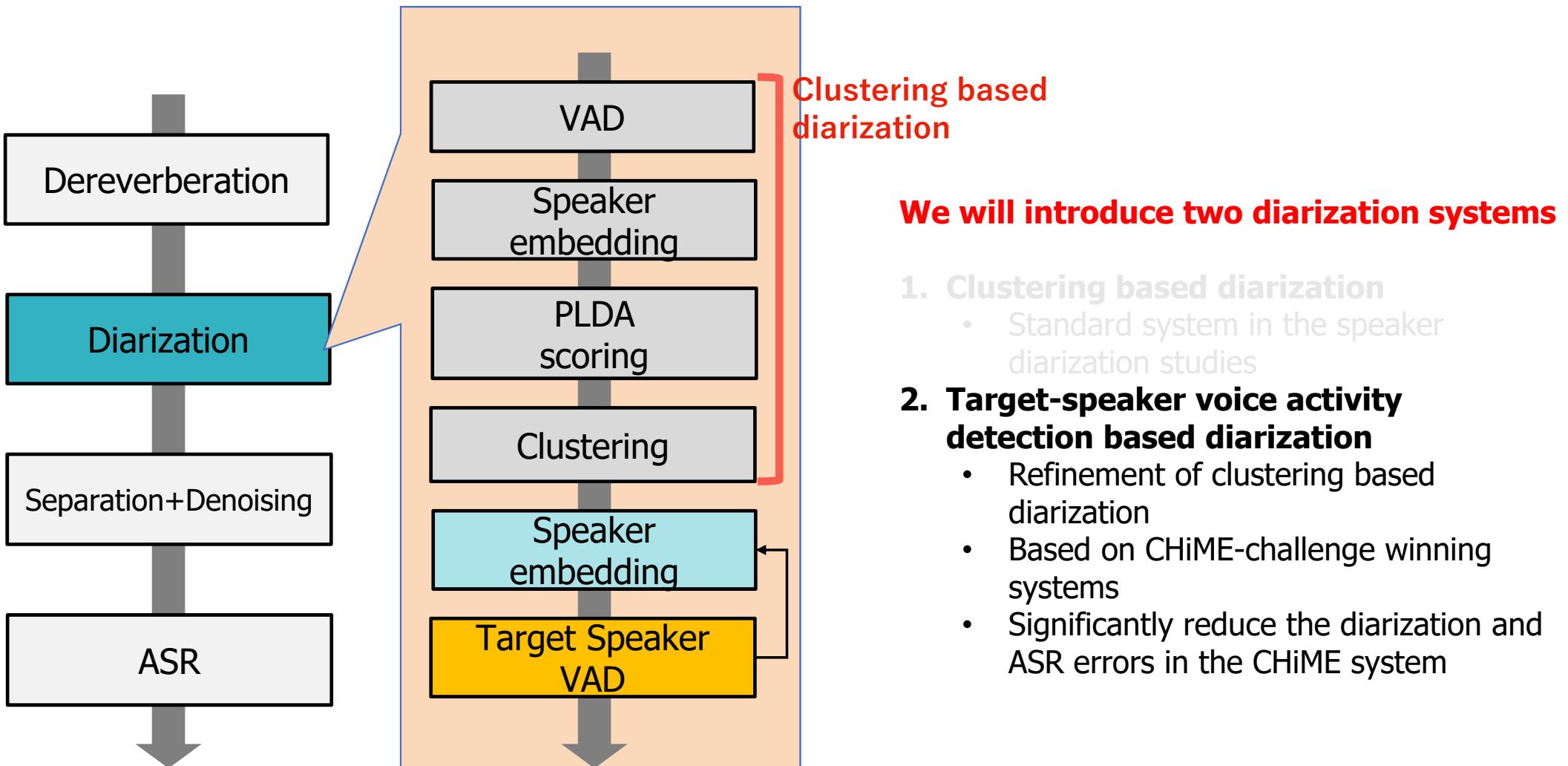
## 1. Clustering based diarization

- Standard system in the speaker diarization studies

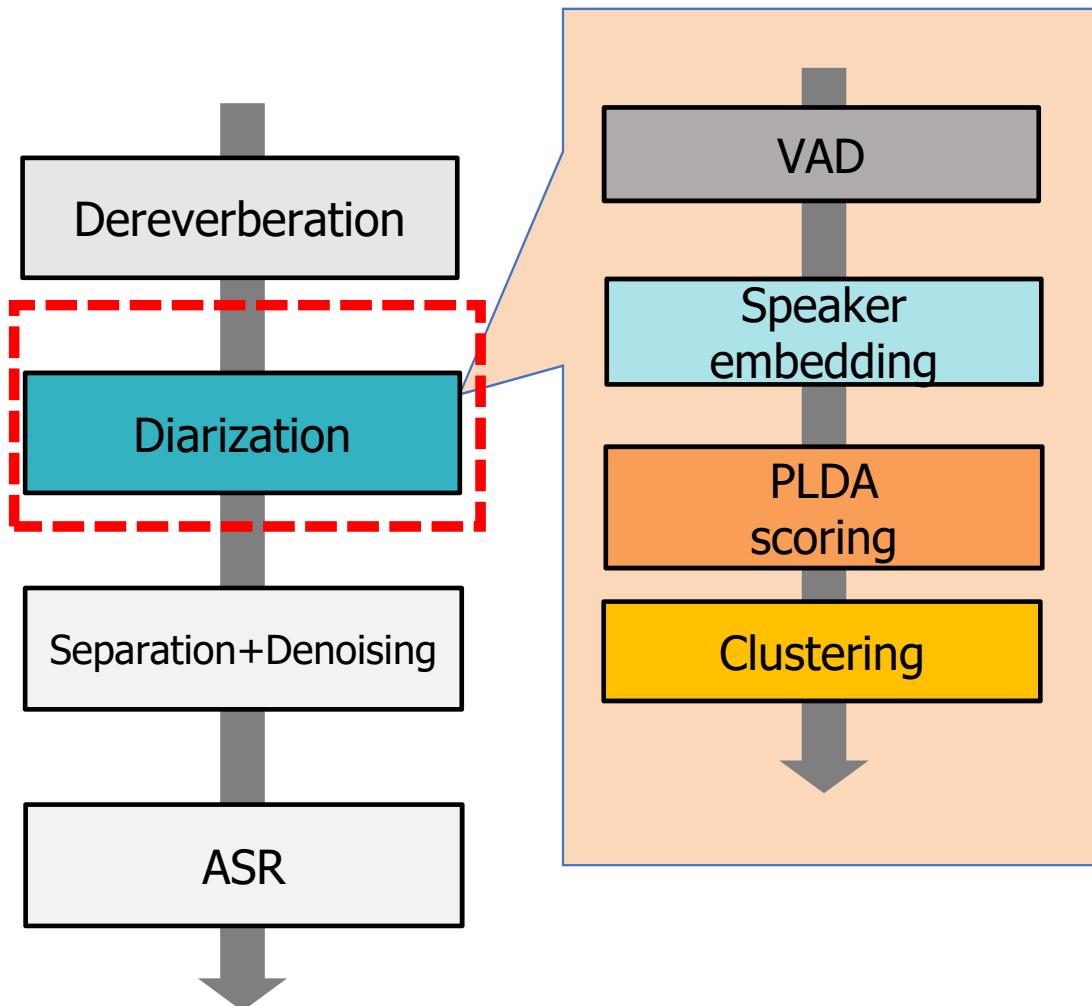
## 2. Target-speaker voice activity detection based diarization

- Refinement of clustering based diarization
- Based on CHiME-challenge winning systems
- Significantly reduce the diarization and ASR errors in the CHiME system

# CHiME-6 pipeline



# Diarization



**Input:** Dereverberated speech

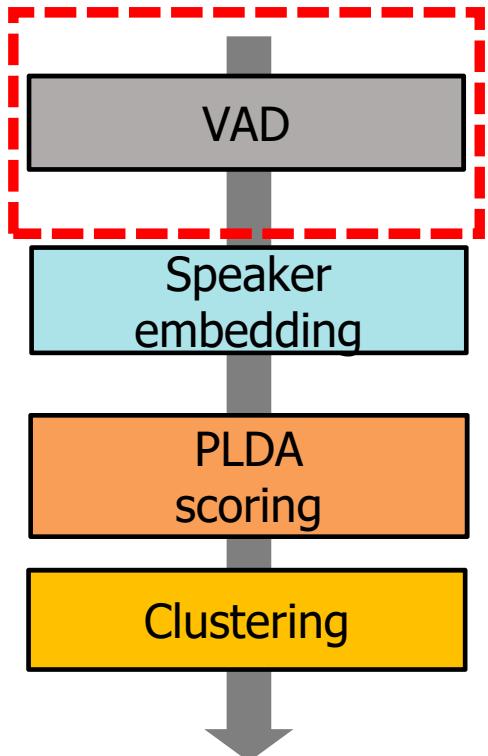
**Output:** Diarization output (e.g., RTTM)

Clustering based diarization is composed of

- **Voice Activity Detection (VAD)**, also called Speaker Activity Detection (SAD)
  - Generates speech segments
- **Speaker Embedding**
  - Compute a vector for each segment
- **Scoring** based on Probabilistic linear discriminant analysis (**PLDA**)
- **Clustering**

We will briefly discuss each module

# VAD module in Diarization

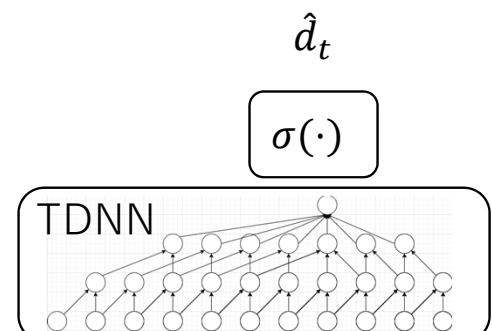


**Input:** Dereverberated speech

**Output:** Speech segment

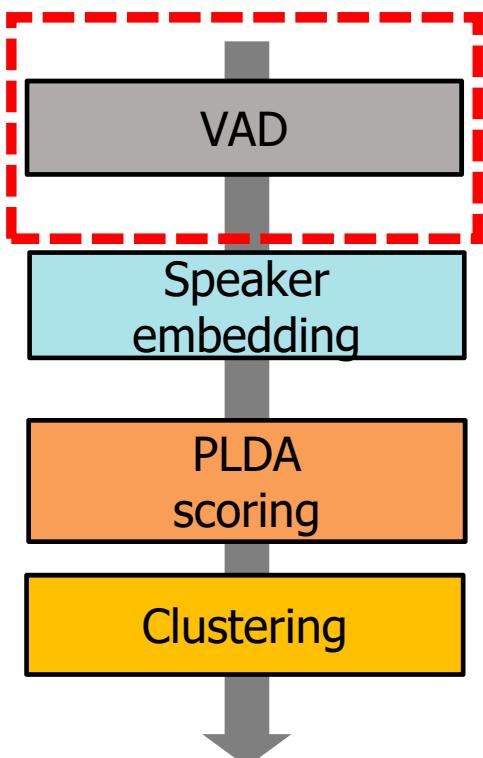


- Predict speech activity  $\hat{d}_t$  based on LSTM, SyncNet or Time-delay neural network (TDNN)  
$$\hat{d}_t = \sigma(\text{TDNN}(\mathbf{O})) \in (0, 1)$$
  - TDNN( $\cdot$ ): Time-delay neural network
  - $\mathbf{O}$ : Enhanced speech feature
  - $\sigma(\cdot)$ : Sigmoid function
- We use a threshold to obtain  $d_t \in \{0, 1\}$

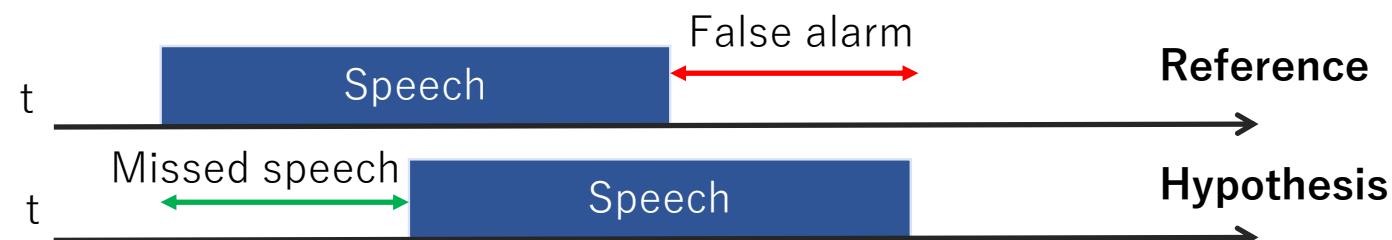


Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
0007400-0007588		74.0	75.88	
0008380-0009435		83.8	94.35	

# VAD module in Diarization



- VAD's performance metric



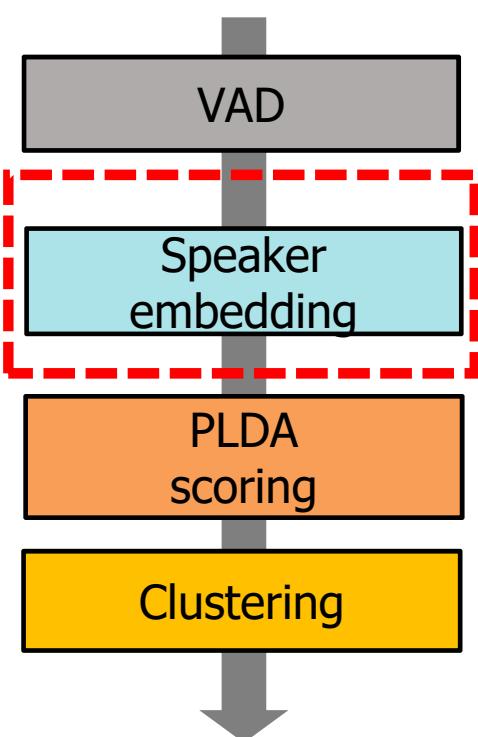
- VAD performance

Dev.			Eval.		
Missed speech	False alarm	Total error	Missed speech	False alarm	Total error
1.0%	1.0%	2.0%	2.0%	3.2%	5.2%

from [https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b\\_track2](https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b_track2)

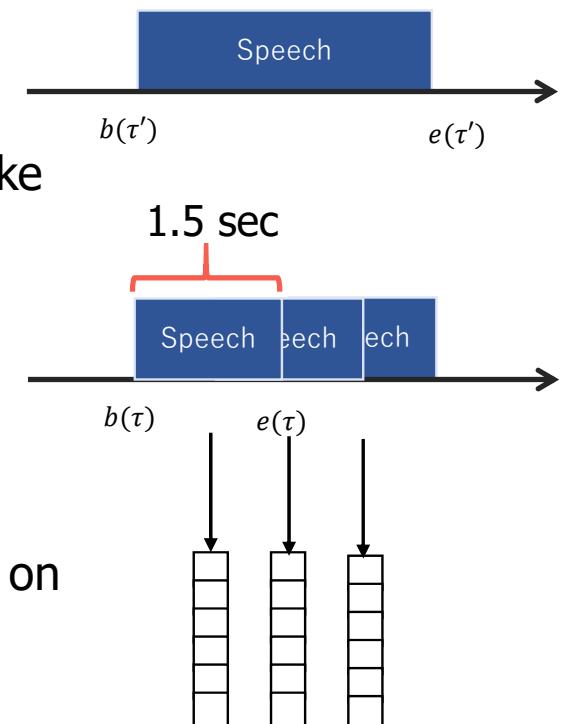
- VAD is fairly working well

# Embedding and clustering in Diarization

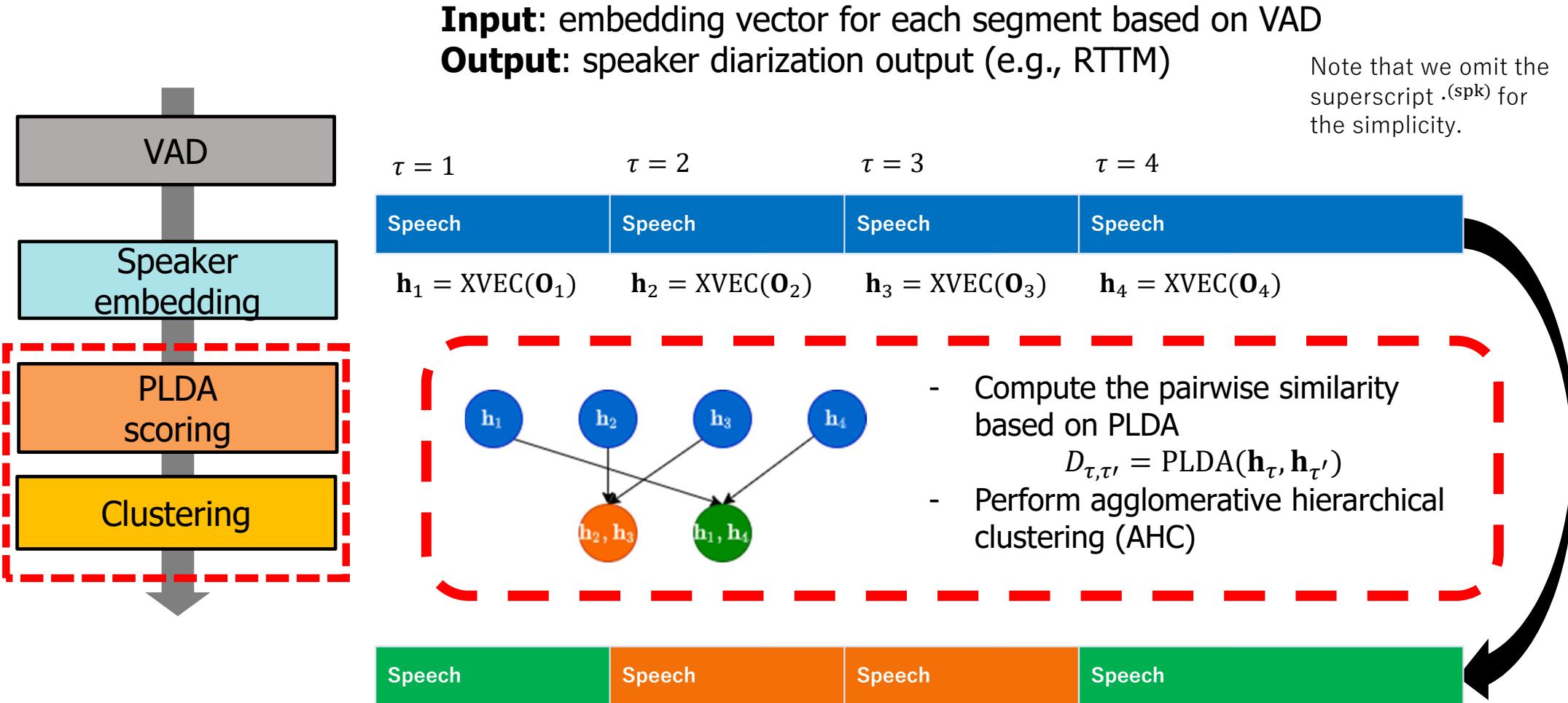


**Input:** dereverberated speech and VAD segment  
**Output:** embedding vector for each segment

- Original VAD:  $\mathbf{O}_{\tau'} = (\mathbf{o}_{t=b(\tau')}, \dots, \mathbf{o}_{t=e(\tau')})$ 
  - $\tau'$ -th speech segment from VAD
  - $b(\tau')$ : onset time,  $e(\tau')$ : offset time
- Further segment it with every **1.5sec** and make it the same length  $e(\tau) - b(\tau) = 1.5$  sec
  - 0.25 sec shift
- x-vector  $\mathbf{h}_\tau^{(spk)}$ : Neural speaker embedding XVEC( $\cdot$ ) for segment  $\tau$   
$$\mathbf{h}_\tau^{(spk)} = \text{XVEC}(\mathbf{O}_\tau) = \text{StatPool}(\text{TDNN}(\mathbf{O}_\tau))$$
  - StatPool( $\cdot$ ): Statistical pooling layer based on mean and variance operations
  - TDNN( $\cdot$ ): Time delay neural network
  - ResNet is also used

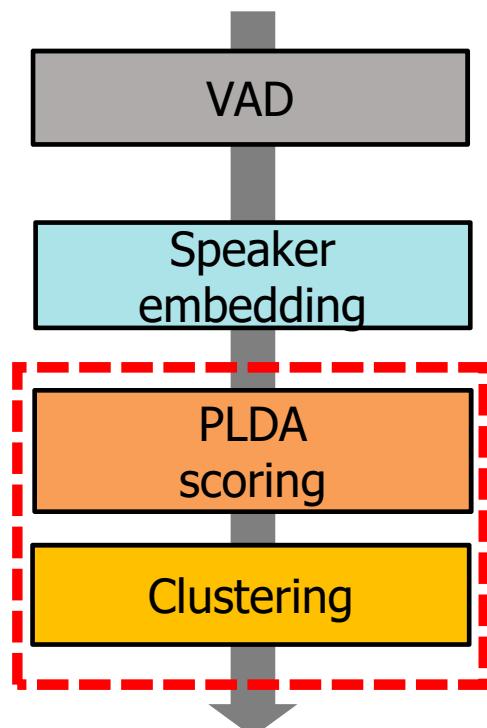


# Embedding and clustering in Diarization



# Embedding and clustering in Diarization

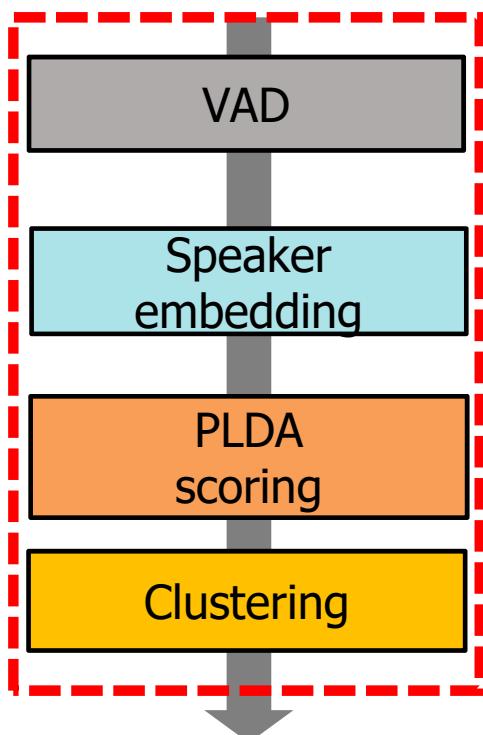
**Input:** embedding vector for each segment based on VAD  
**Output:** speaker diarization output (e.g., RTTM)



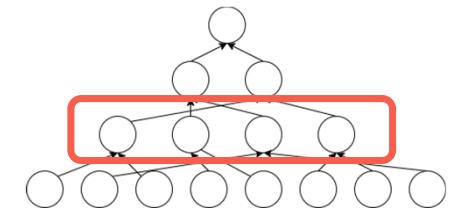
Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
0007400-0007588		74.0	75.88	
0008380-0009435		83.8	94.35	
0031242-0031700		312.4	317.0	

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
A-0007400-0007588	A	74.0	75.88	
A-0008380-0009435	A	83.8	94.35	
B-0031242-0031700	B	312.4	317.0	

# Embedding and clustering in Diarization



- x-vector neural speaker embedding model is trained with VoxCeleb
- PLDA model is trained with CHiME-6
- The number of speakers is **given** (=4)



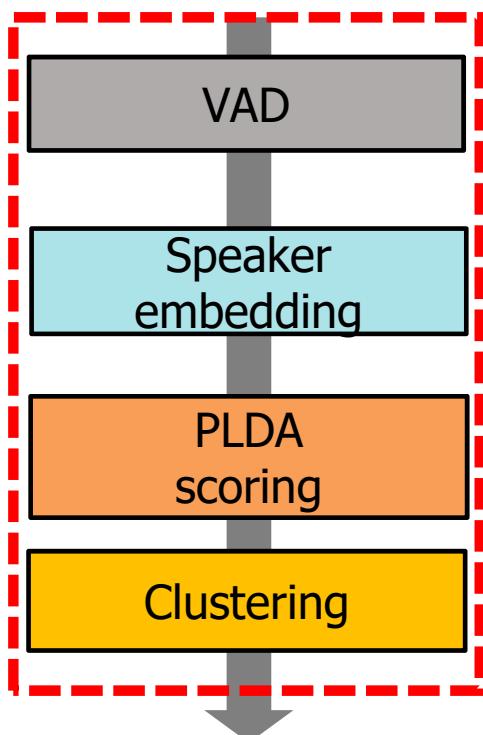
Stop the clustering  
when #nodes become 4

Dev. DER	Eval. DER
63.4%	68.2%

from [https://chimechallenge.github.io/chime6/track2\\_instructions.html](https://chimechallenge.github.io/chime6/track2_instructions.html)

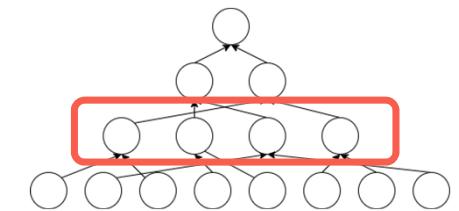
- Diarization performance was poor in the CHiME scenario due to the large speaker overlap
  - However, this shows very strong performance in the other scenarios including DIHARD (See the discussion in Section 3.1)

# Embedding and clustering in Diarization



- x-vector neural speaker embedding model is trained with VoxCeleb
- PLDA model is trained with CHiME-6
- The number of speakers is **given** (=4)

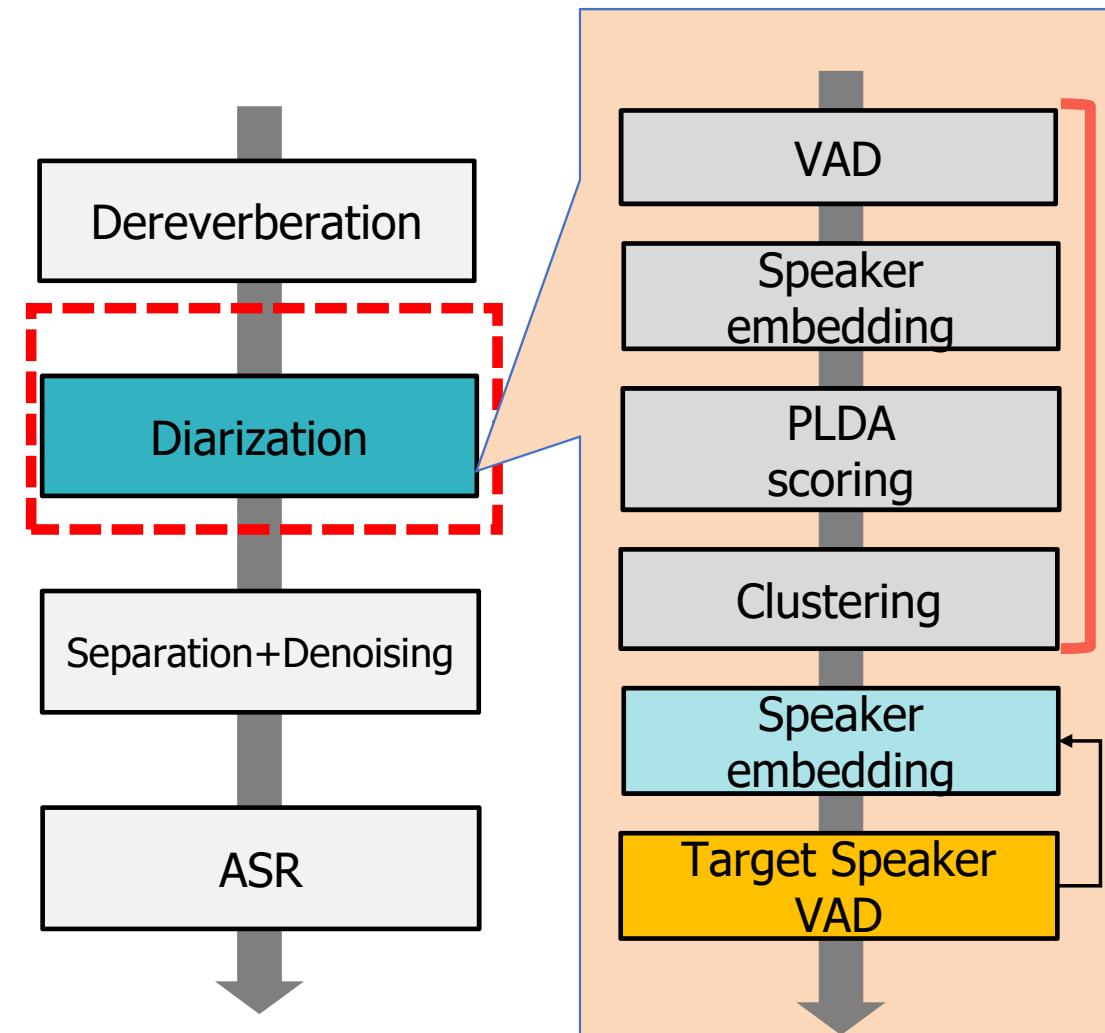
Dev.	Eval.
Diarization error rate (DER)	Diarization error rate (DER)
59.0%	64.7%



Stop the clustering  
when #nodes become 4

- Diarization performance was poor in the CHiME scenario due to the large speaker overlap
- However, this shows very strong performance in the other scenarios including DIHARD

# Diarization based on target speaker VAD

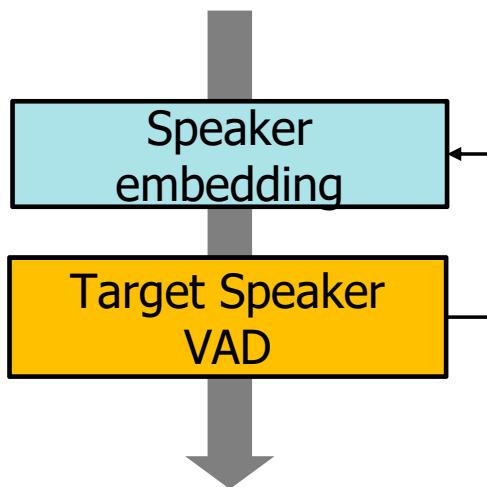


## Clustering based diarization

- Initialized by clustering based diarization
- Embedding vector extraction for each speaker
- Target-speaker voice activity detection
  - **Multi-label classification** problems for each speaker
  - **Overlap can be handled** by joint label activation
  - Inspired by End-to-End Neural Diarization (c.f. Section 3.1)
  - Performed **iteratively**

Medennikov et al., “The STC System for the CHiME-6 Challenge,” CHiME 2020

# Diarization based on target speaker VAD

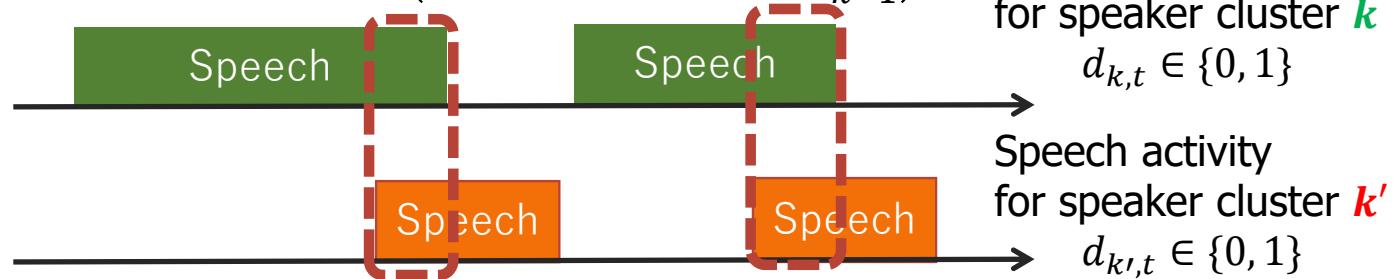


Target-speaker voice activity detection

- **Multi-label classification** problems for each speaker, i.e.,

$$p\left(\{d_{k,t}\}_{k=1}^K \mid \mathbf{O}, \{\mathbf{h}_k^{(\text{spk})}\}_{k=1}^K\right)$$

Speech activity  
for speaker cluster  $\mathbf{k}$   
 $d_{k,t} \in \{0, 1\}$



- **Naturally handles overlap**, e.g.,  $d_{k,t} = d_{k',t} = 1$  at frame  $t$
- Note that this is very similar to VAD, i.e.,  $p(d_t | \mathbf{O})$  except that it is conditioned on the speaker-dependent variable

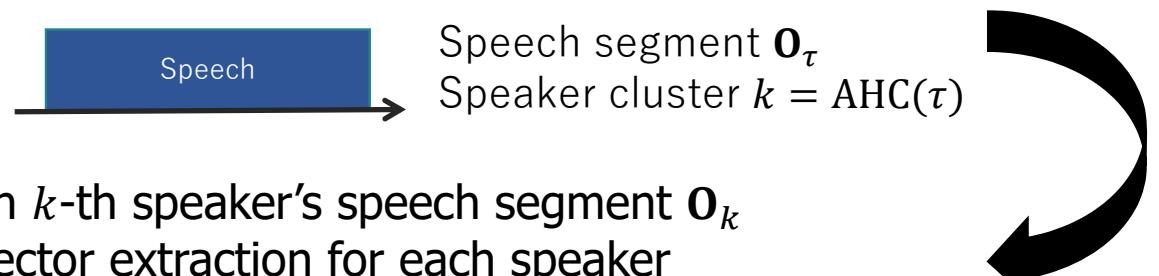
Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

# Diarization based on target speaker VAD

- How to make a speaker-dependent classifier  $p\left(\{d_{k,t}\}_{k=1}^K \mid \mathbf{0}, \{\mathbf{h}_k^{(spk)}\}_{k=1}^K\right)$ ?

We use speaker embedding vector  $\mathbf{h}_k$

- Based on the information obtained by the preceding (x-vector) diarization system, we can obtain the speaker cluster index  $k = \text{AHC}(\tau)$  for each segment



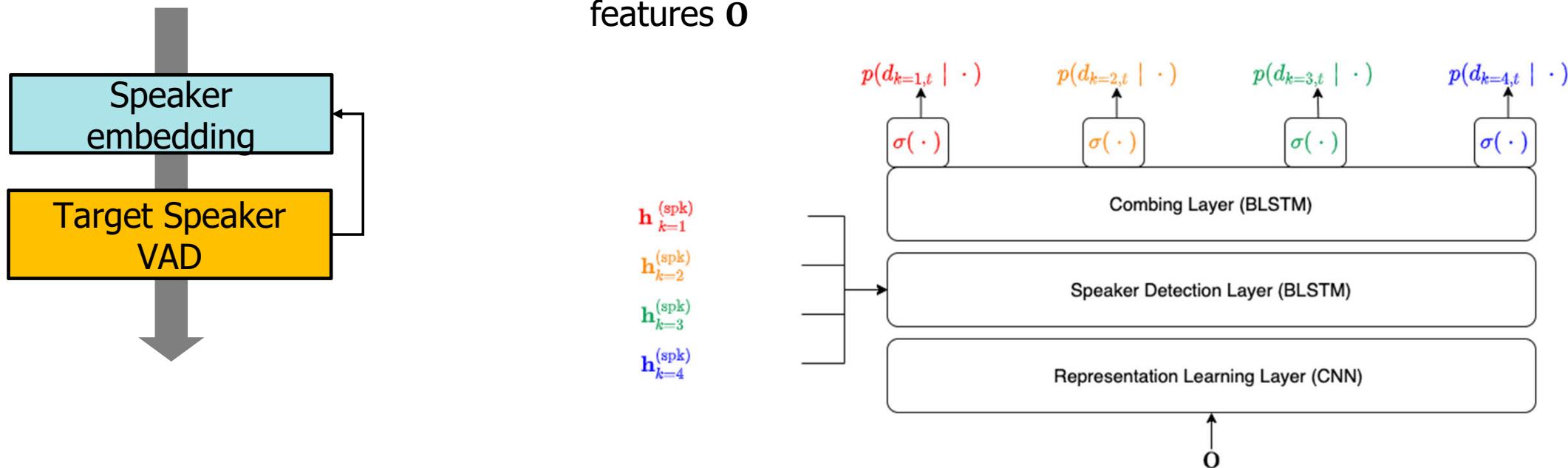
- We can obtain  $k$ -th speaker's speech segment  $\mathbf{0}_k$
- Embedding vector extraction for each speaker
  - Compute an **i-vector** for entire speech  $\mathbf{0}$ , i.e.,

$$\mathbf{h}_k^{(spk)} = \text{IVEC}(\mathbf{0}_k)$$

Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

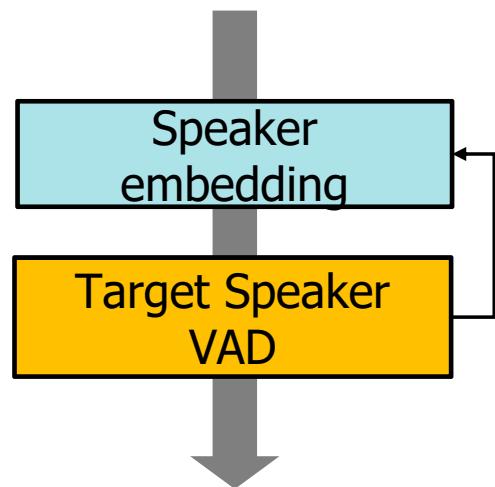
# Diarization based on target speaker VAD

- **Multi-label classification**  $p\left(\{d_{k,t}\}_{k=1}^K \mid \mathbf{0}, \{\mathbf{h}_k^{(\text{spk})}\}_{k=1}^K\right)$  is obtained by feeding all speaker embedding information in addition to MFCC features  $\mathbf{0}$



Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

# Diarization based on target speaker VAD



## Iterative processing

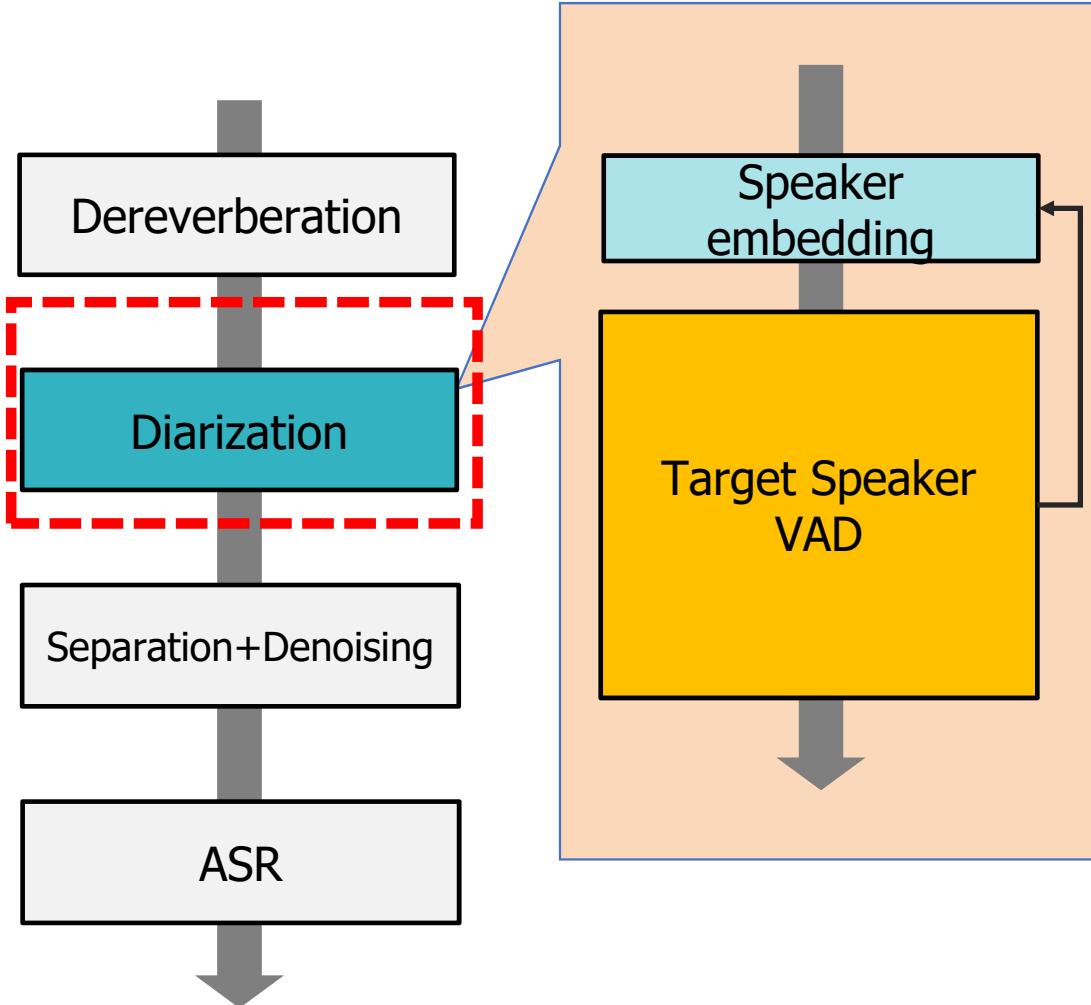
$$\text{TS-VAD} \quad p\left(\left\{d_{k,t}\right\}_{k=1}^K \mid \mathbf{O}, \left\{\mathbf{h}_k^{(\text{spk})}\right\}_{k=1}^K\right)$$

$$\text{Speaker embedding } \mathbf{h}_k^{(\text{spk})} = \text{IVEC}(\mathbf{O}_k)$$

- Note that the initial diarization result is very critical (e.g., x-vector based on ResNet and spectral clustering)

Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

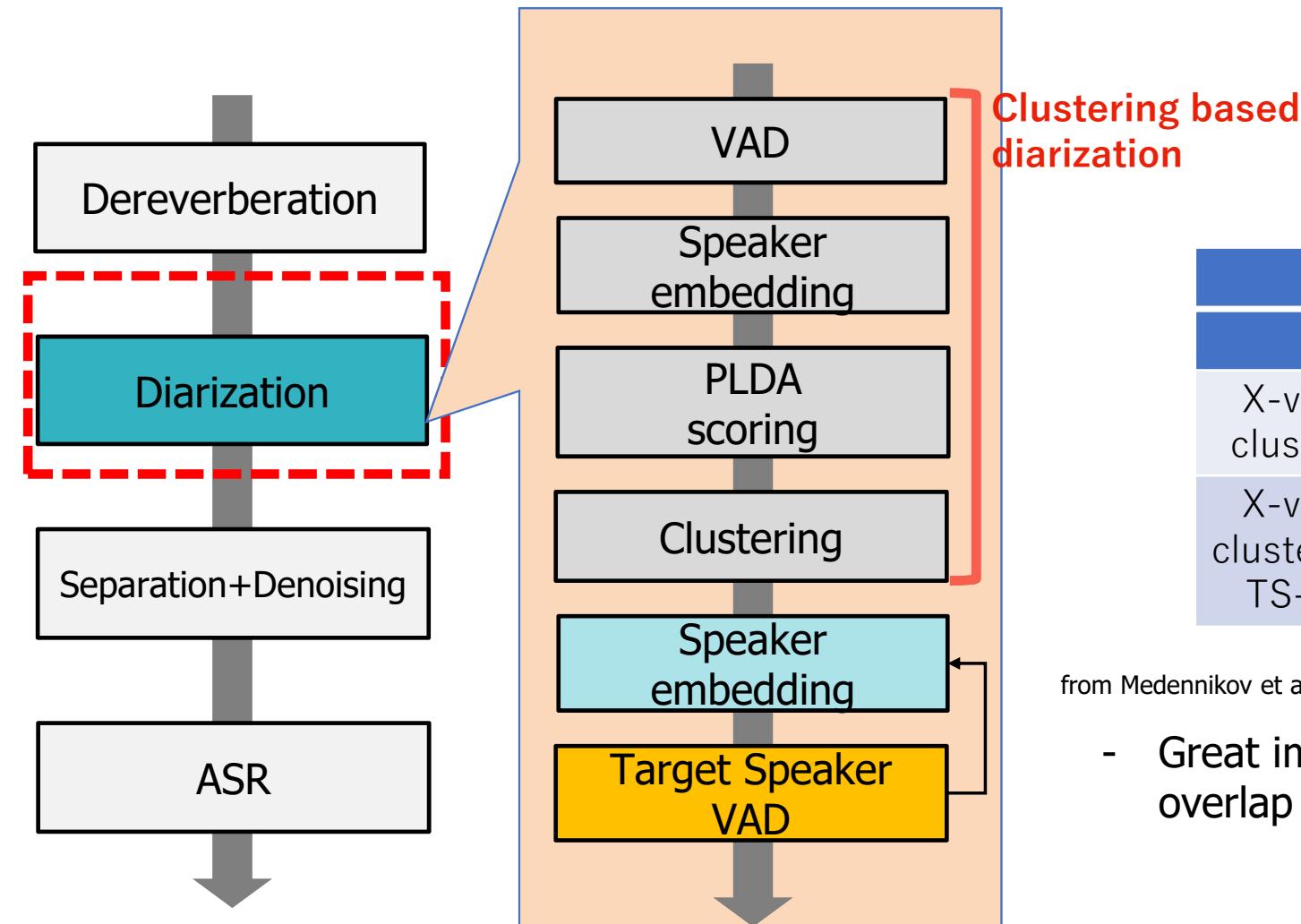
# Diarization based on target speaker VAD



	Dev.	Eval.
	DER	DER
X-vector clustering	59.0%	64.7%
TS-VAD	44.9%	42.0%

- Great improvements based on the explicit overlap handling and iterative procedure

# Diarization based on target speaker VAD



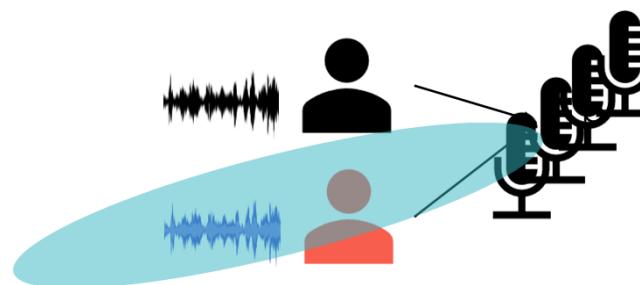
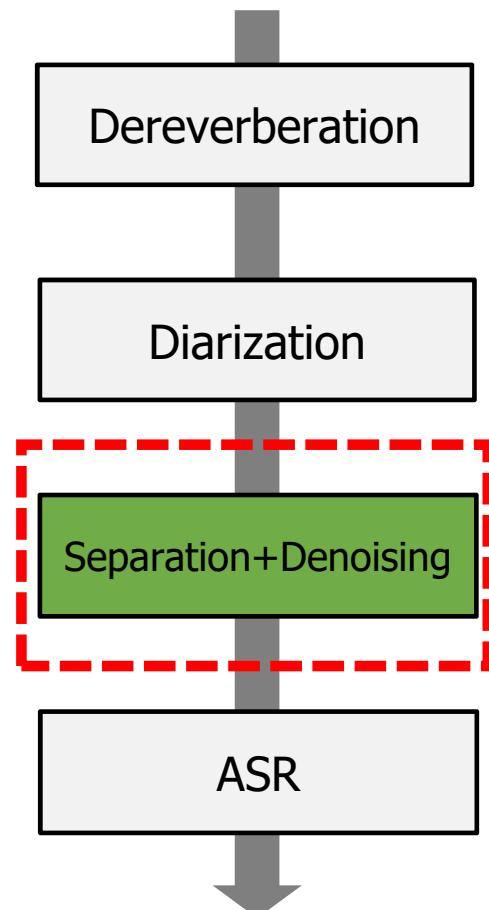
	Dev.	Eval.
	DER	DER
X-vector clustering	63.4%	68.2%
X-vector clustering + TS-VAD	32.8%	36.0%

from Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

- Great improvements based on the explicit overlap handling and iterative procedure

# Speech enhancement (Separation+Denoising)

## - Beamforming

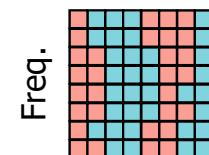
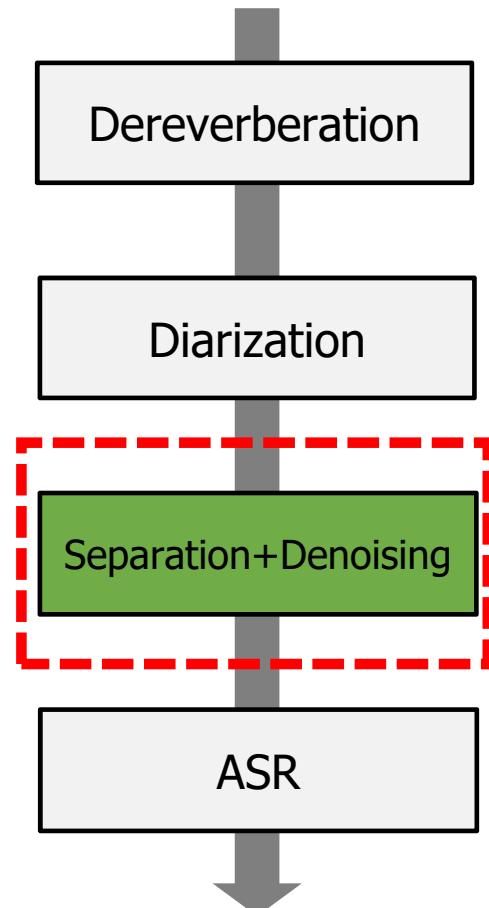


- Only extract a source by making a beam
- Time-frequency (TF) mask is used to estimate the beamforming parameters

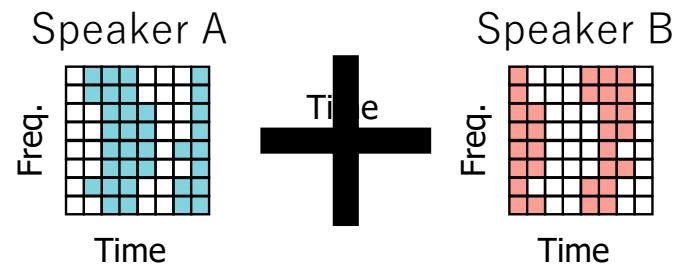
This is called mask-based beamformer (will also be revisited in Section 3.2)

# Speech enhancement (Separation+Denoising)

- TF Mask



(based on the sparsity)



$$p(k|\tilde{\mathbf{y}}_{t,f})$$

Estimate the dominant presence probability  
(mask) for speaker  $k$  given observation  $\mathbf{y}$

# Speech enhancement (Separation+Denoising)

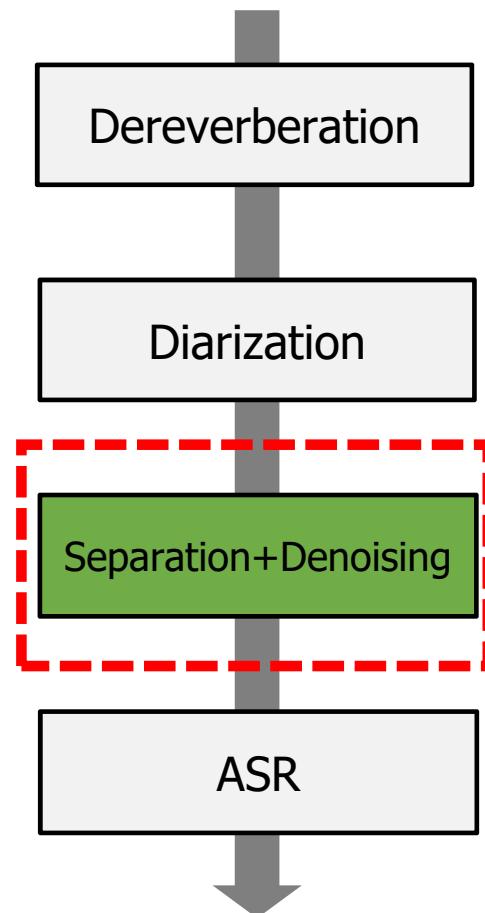
[Ito+, 2016]

[Boedekker+, 2018]

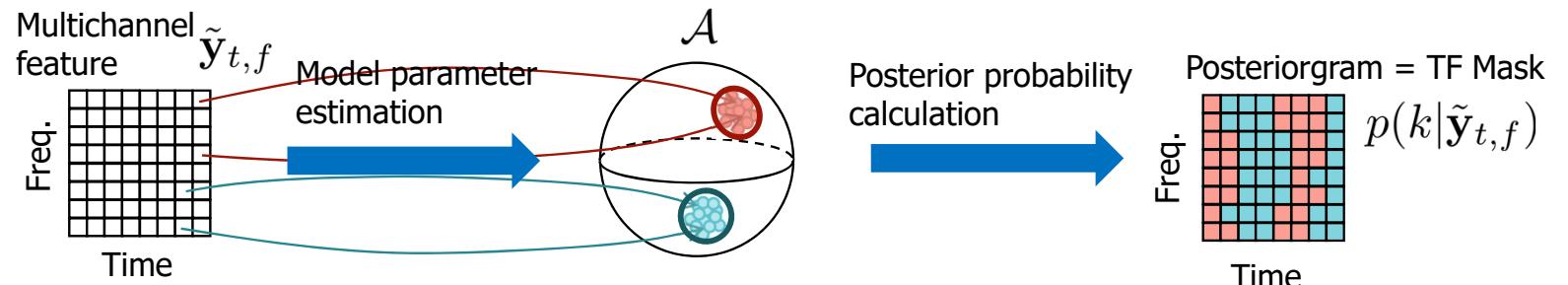
- Mask-based BF = Mask estimation + BF

- TF Mask estimation: complex Angular Central (cAC)-GMM clustering

- BF: Minimum variance distortionless response (MVDR) BF



- cAC-GMM clustering can provide TF-mask  $p(k|\tilde{\mathbf{y}}_{t,f})$



Step1: Parameter estimation of a mixture model  
e.g., complex angular central GMM  $\mathcal{A}$

$$p(\tilde{\mathbf{y}}_{t,f}; \theta_f) = \sum_{k=1}^K \pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})$$

Step2: Calculate a mask for  $k$ -th speaker  
as the posterior probability of  $k$ -th component:

$$p(k|\tilde{\mathbf{y}}_{t,f}) = \frac{\pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})}{\sum_{k=1}^K \pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})},$$

# Speech enhancement (Separation+Denoising)

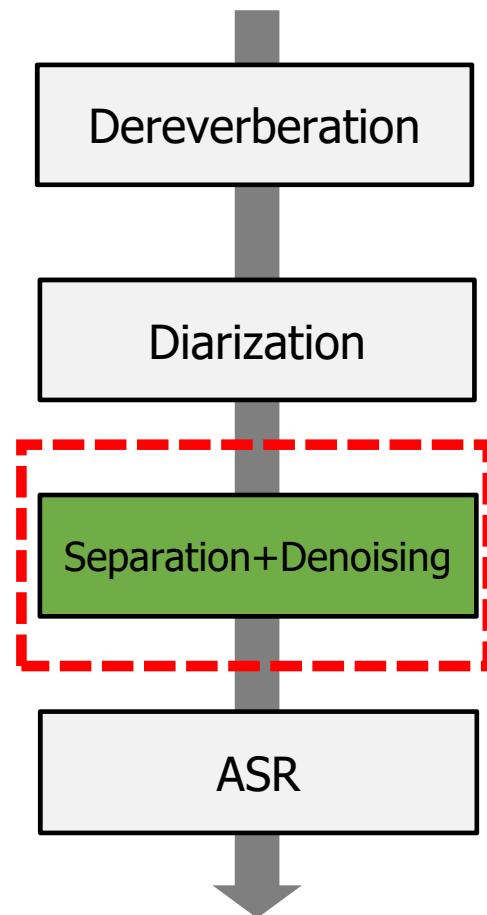
[Ito+, 2016]

[Boedekker+, 2018]

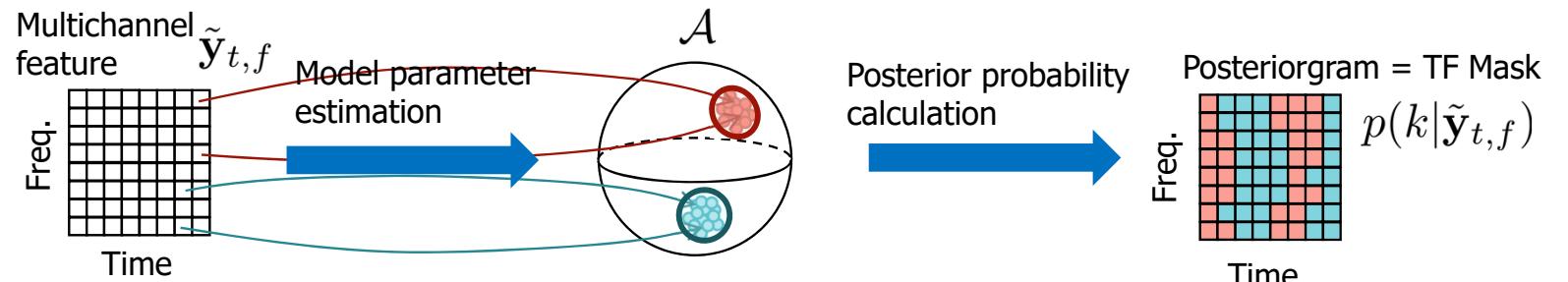
- Mask-based BF = Mask estimation + BF

- TF Mask estimation: complex Angular Central (cAC)-GMM clustering

- BF: MVDR BF



- How can we estimate the mask  $p(k|\tilde{\mathbf{y}}_{t,f})$ ?



Step1: Parameter estimation of a mixture model  
e.g., complex angular central GMM  $\mathcal{A}$

$$p(\tilde{\mathbf{y}}_{t,f}; \theta_f) = \sum_{k=1}^K \pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})$$

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$$p(k|\tilde{\mathbf{y}}_{t,f}) = \frac{\pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})}{\sum_{k=1}^K \pi_{f,k} \mathcal{A}(\tilde{\mathbf{y}}_{t,f}; \mathbf{B}_{f,k})},$$

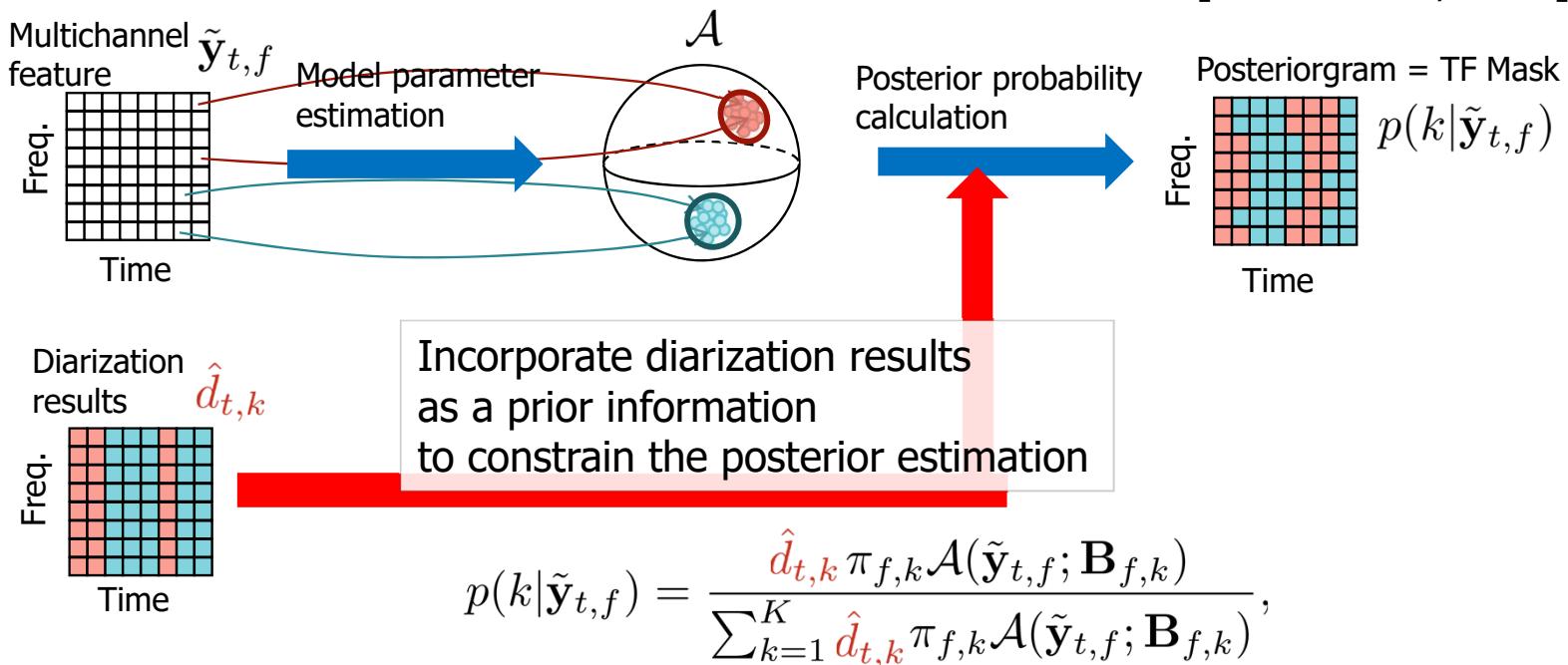
# Speech enhancement (Separation+Denoising)

[Boedekker+, 2018]

- Mask-based BF = Mask estimation + BF
  - TF Mask estimation: complex Angular Central (cAC)-GMM clustering
  - BF: Minimum variance distortionless response (MVDR) BF

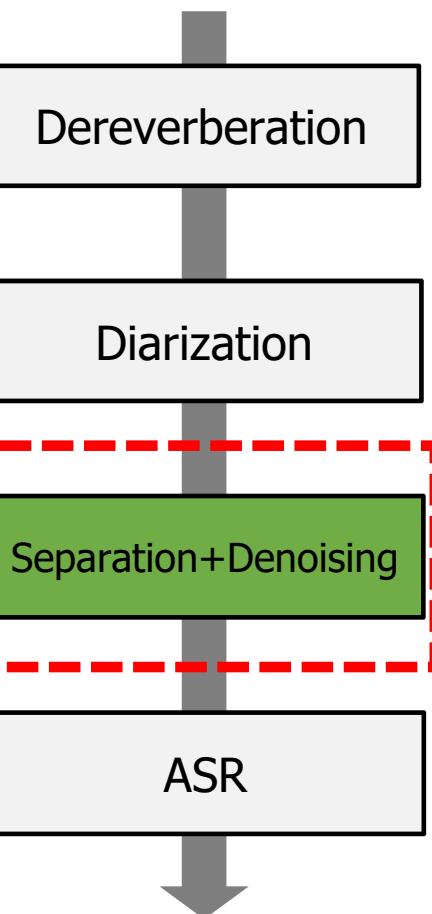
- How to incorporate diarization results into cAC-GMM clustering

[Boedekker+, 2018]



# Speech enhancement (Separation+Denoising)

[Boedekker+, 2018]



- Mask-based BF = Mask estimation + BF
  - TF Mask estimation: complex Angular Central (cAC)-GMM clustering
  - BF: Minimum variance distortionless response (MVDR) BF
- How to calculate a BF filter based on the obtained mask  $\hat{m}_{t,f,k} = p(k|\tilde{\mathbf{y}}_{t,f})$  ?

Step1: Spatial covariance estimation based on the mask

$$\Phi_{f,k} = \frac{1}{\sum_{t=1}^T \hat{m}_{t,f,k}} \sum_{t=1}^T \hat{m}_{t,f,k} \mathbf{y}_{t,f} \mathbf{y}_{t,f}^H$$

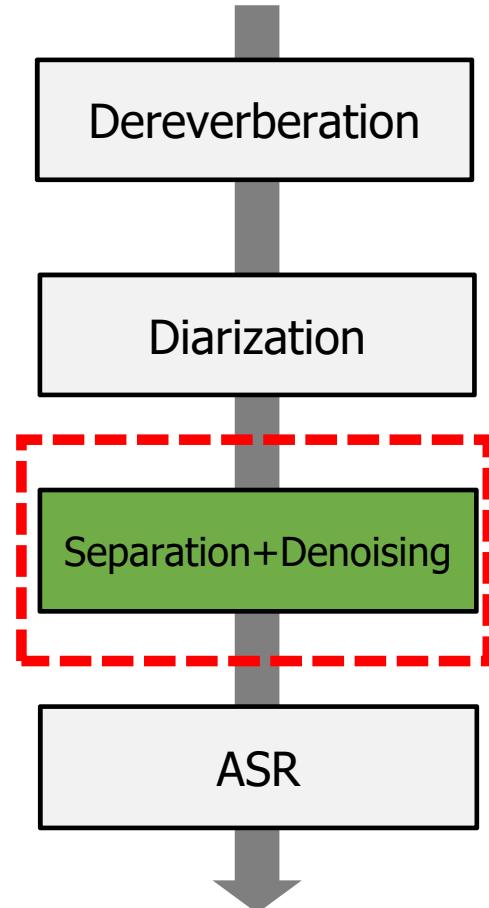
Step2: Filter calculation

$$\mathbf{g}_{f,k} = \frac{(\sum_{k' \neq k} \Phi_{f,k'})^{-1} \Phi_{f,k}}{\text{Tr}((\sum_{k' \neq k} \Phi_{f,k'})^{-1} \Phi_{f,k})} \mathbf{u}$$

Step3: Filtering

$$\hat{x}_{t,f,k} = \mathbf{g}_{f,k}^H \mathbf{y}_{t,f}$$

# Speech enhancement (Separation+Denoising)



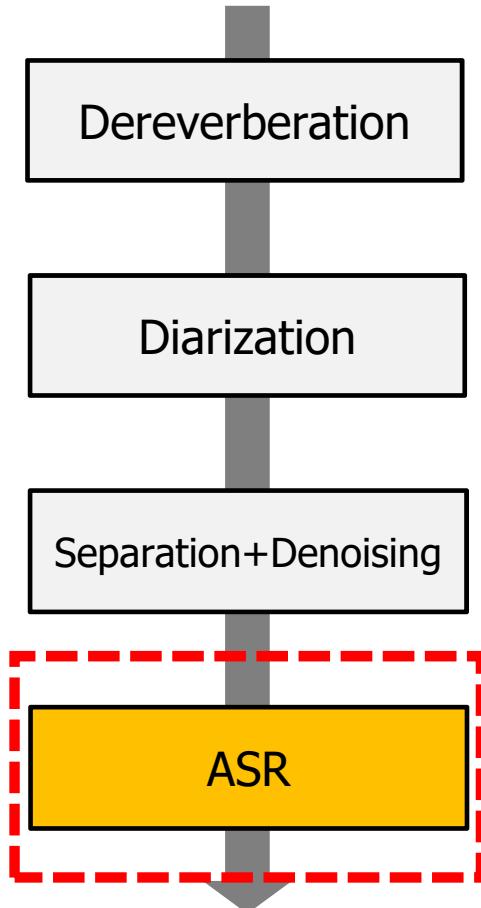
Guided source separation significantly improves the ASR performance

	Dev. SWER	Eval. SWER
w/o GSS	76.7%	72.6%
w/ GSS	<b>70.3%</b>	<b>69.0%</b>

from [https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b\\_track2](https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b_track2)

- This shows the effectiveness of the separation and denoising.

# ASR



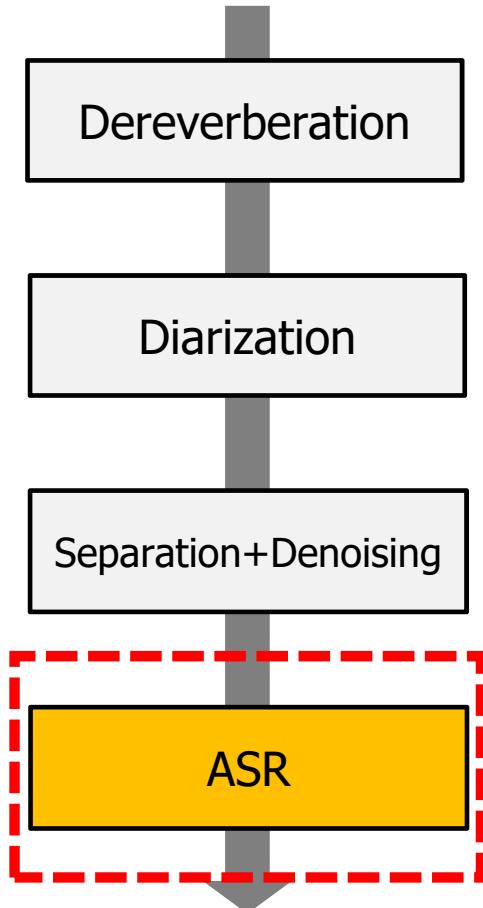
**Input:** Speaker diarization and enhanced speech for each speaker  
**Output:** Transcriptions for each speaker at each segment

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
A-0007400-0007588	A	74.0	75.88	
A-0008380-0009435	A	83.8	94.35	
B-0031242-0031700	B	312.42	317.0	

Given the diarization process, it becomes a standard ASR task  
In this part, we will explain a DNN-HMM hybrid system

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
A-0007400-0007588	A	74.0	75.88	yes
A-0008380-0009435	A	83.8	94.35	hey did you put…
B-0031242-0031700	B	312.42	317.0	hmm

# ASR: Feature extraction



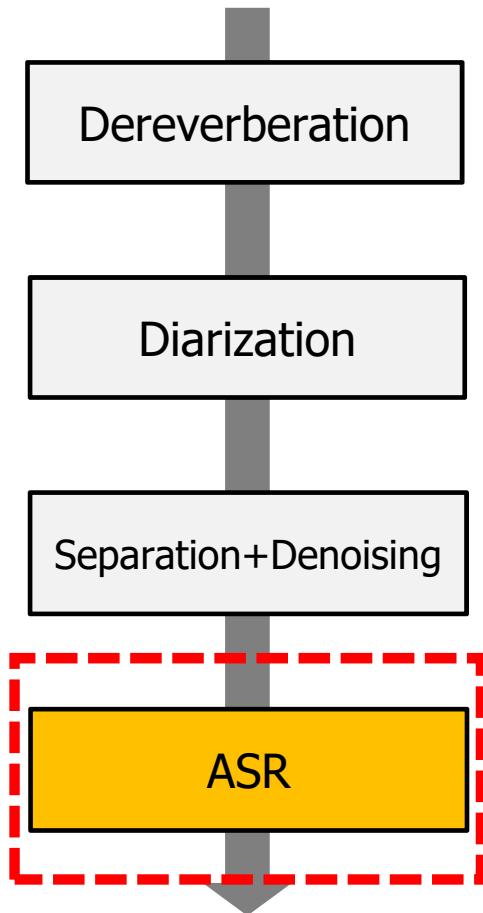
**Feature:** augmented feature based on

- **Log mel filterbank:**  $\mathbf{o}_t \in \mathbf{O}_\tau$  is a  $t$ -th frame Log mel filterbank feature
- **i-vector** for segment  $\mathbf{O}_\tau$ :  $\mathbf{h}_{k,\tau} = \text{IVEC}(\mathbf{O}_\tau)$ 
  - From the speaker diarization information, we can obtain the speaker cluster index  $k = \text{AHC}(\tau)$  for each segment

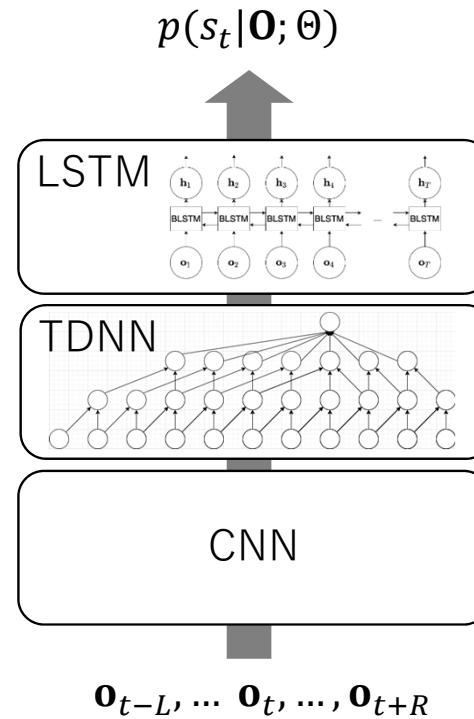


- Concatenate these features:  $\mathbf{o}'_t = \begin{bmatrix} \mathbf{o}_t \\ \mathbf{h}_{k,\tau} \end{bmatrix}$

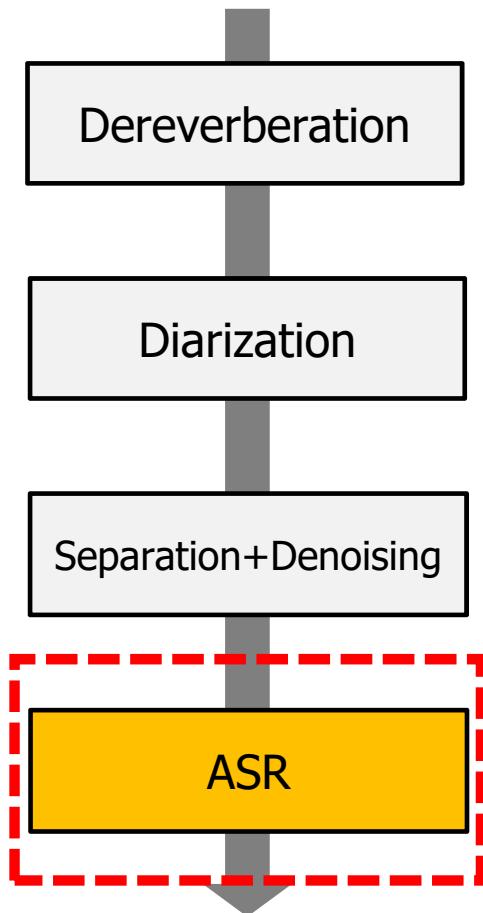
# ASR: Acoustic model



**Acoustic model: CNN + TDNN + LSTM**  
- **Acoustic posterior**  $p(s_t|\mathbf{o}; \Theta)$  for HMM state  $s_t$



# ASR: Acoustic model training



## Acoustic model:

- **Four types of data**

(totally 675 hours)

1. Far-field array microphones  $\mathbf{O}^{\text{kinnect}}$
2. Near-field (worn) microphones  $\mathbf{O}^{\text{worn}}$
3. Augmented data with noises and simulated RIRs  
 $\mathbf{X}^{\text{nr}} = \text{RIR}(\mathbf{X}^{\text{worn}}) + \text{Noise}$   
RIR( $\cdot$ ) is sampled from simulated RIRs  
Noise is sampled from CHiME-6 noise data
4. Augmented data with enhanced signals obtained with GSS  $\mathbf{O}^{\text{gss}}$



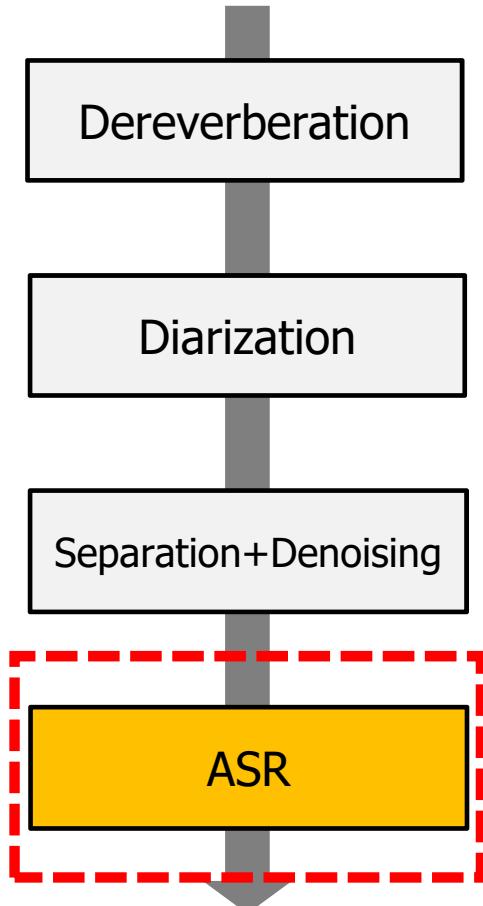
- **Lattice-free MMI-based loss**

$$\theta^{\text{LF-MMI}} = \underset{\theta}{\operatorname{argmin}} \mathcal{L}(\mathbf{O}^{\text{kinnect}}, \mathbf{O}^{\text{worn}}, \mathbf{O}^{\text{nr}}, \mathbf{O}^{\text{gss}}; \theta)$$

Zorila et al., "Toshiba's Speech Recognition System for the CHiME 2020 Challenge," CHiME 2020

Povey et al., "Purely sequence-trained neural networks for ASR based on lattice-free MMI," Interspeech 2016

# ASR: Inference



**Language model:** Lattice rescoring based on LSTMs

$$\text{Score}(\mathbf{O}, \mathbf{W}) = \log p(\mathbf{O}|\mathbf{W}; \Theta^{\text{LF-MM}}) + \lambda^{\text{ngram}} \log p^{\text{ngram}}(\mathbf{W}) + \lambda^{\text{lstm}} \log p^{\text{lstm}}(\mathbf{W})$$

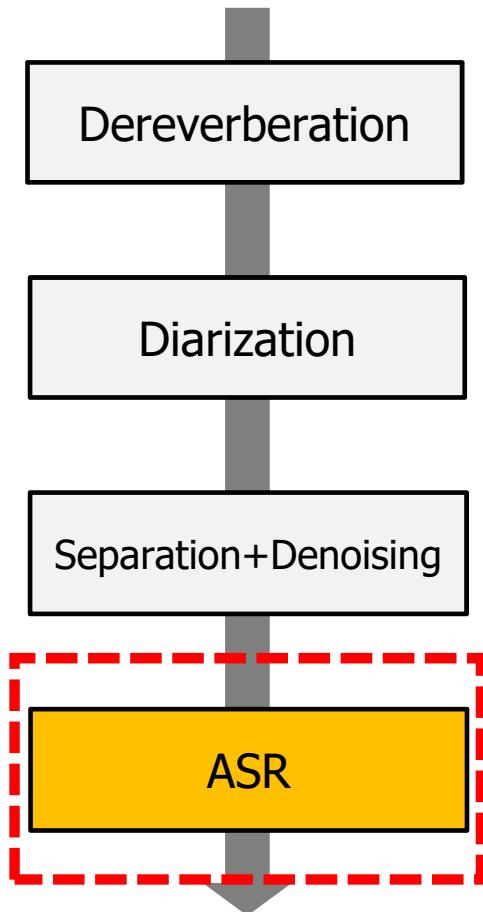
A new score is computed based on the following three scores:

- $p(\mathbf{O}|\mathbf{W}; \Theta^{\text{LF-MM}})$ : Acoustic score (pseudo likelihood)
- $p^{\text{ngram}}(\mathbf{W})$ : N-gram LM score with scaling factor  $\lambda^{\text{ngram}}$
- $p^{\text{lstm}}(\mathbf{W})$ : LSTM LM score with scaling factor  $\lambda^{\text{lstm}}$

**Inference:** Two stage decoding

- Refine i-vector  $\mathbf{h}_{k,\tau}$  based on the first pass decoding result  
(The onset time  $b(\tau)$  and offset time  $e(\tau)$  will be updated based on the decoding result)

# ASR: Inference



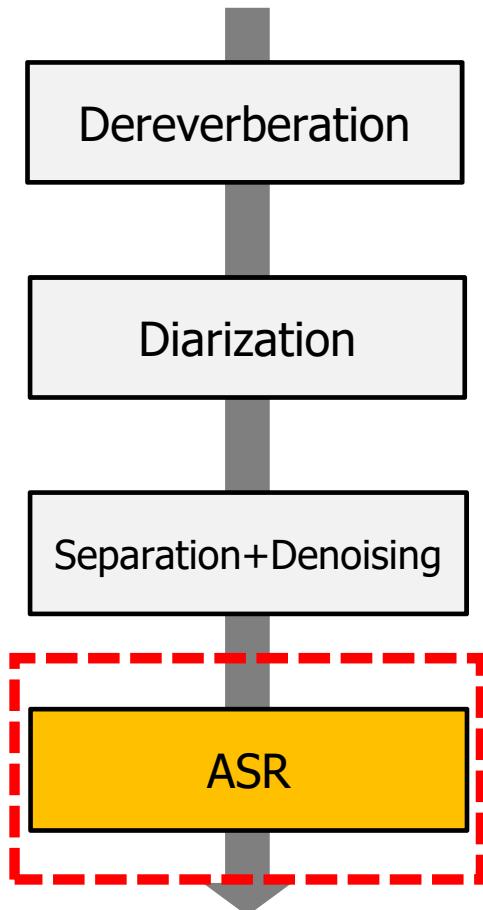
**Language model:** Lattice rescoring based on LSTMs

$$\text{Score}(\mathbf{O}, \mathbf{W}) = \log p(\mathbf{O}|\mathbf{W}; \Theta^{\text{LF-MM}}) + \lambda^{\text{ngram}} \log p^{\text{ngram}}(\mathbf{W}) + \lambda^{\text{lstm}} \log p^{\text{lstm}}(\mathbf{W})$$

A new score is computed based on the following three scores:

- $p(\mathbf{O}|\mathbf{W}; \Theta^{\text{LF-MM}})$ : Acoustic score (pseudo likelihood)
- $p^{\text{ngram}}(\mathbf{W})$ : N-gram LM score with scaling factor  $\lambda^{\text{ngram}}$
- $p^{\text{lstm}}(\mathbf{W})$ : LSTM LM score with scaling factor  $\lambda^{\text{lstm}}$

# ASR results



	Diarization	Dev. SWER	Eval. SWER
Original baseline	Oracle	51.8%	51.3%
System [1]	Oracle	30.8%	30.5%
	Diarization	Dev. WER	Eval. WER
Original baseline	X-vector	84.3%	77.9%
System [2]	X-vector + TS-VAD	39.6%	42.7%

from [1] Du et al, "The USTC-NELSLIP Systems for CHiME-6 Challenge," CHiME 2020  
and [2] Medennikov et al., "The STC System for the CHiME-6 Challenge," CHiME 2020

- Significantly improves the baseline by combining the all above techniques
- TS-VAD further improves the speaker attributed WER (SWER)
- Note that these CHiME challenge systems are based on iterative processing, intensive data augmentation, and system combinations, and quite complicated

## 2. Current state-of-the-art systems

2.1. Descriptions of the techniques

2.2. Reproducible baselines

# Reproducible baselines

- **Reproducible (all open source) baseline**
  - Includes all modules, e.g., speech activity detection (SAD), speaker embedding, and speaker diarization in addition to ASR
  - **All-in-one recipe** including training and inference
- But maintain the **simplicity**
  - Excludes iterative processing, intensive data augmentation, and system combinations
  - Unfortunately, this will lead degradations from the state-of-the-art systems in the previous section (42.7% → 60.3%)
- This is one of the first baselines that integrate all multi speaker speech processing in this real scenario ☺

# CHiME-6 command line demo

[https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b\\_track2/run\\_demo.sh](https://github.com/desh2608/kaldi/blob/demo/egs/chime6/s5b_track2/run_demo.sh)

Special thanks to Desh Raj Ashish Arora, Aswin Subramanian, and Sanjeev Khudanpur

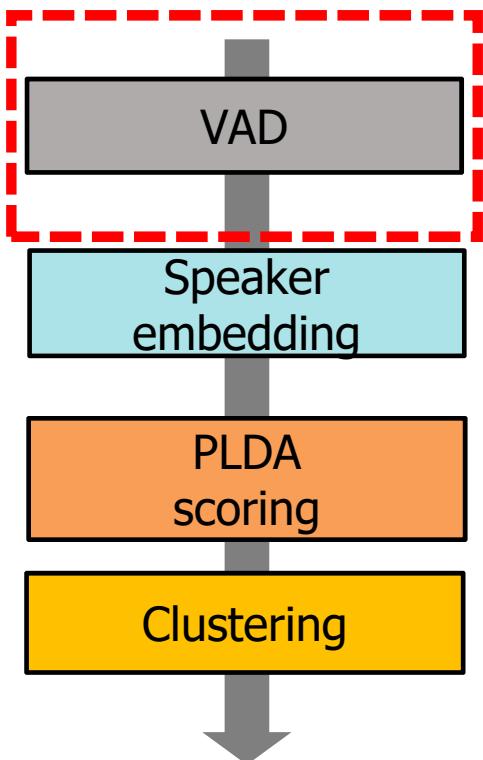
We have three systems in the Kaldi repository

- s5\_track2              Original Kaldi CHiME-6 baseline
- s5b\_track2              **Latest Kaldi system (x-vector clustering) ← mainly introduce this**
- s5c\_track2              **Latest Kaldi system (TS-VAD by STC)** thanks to help from the STC-innovation researchers

```
cd <your_kaldi>/kaldi/egs/chime6/s5b_track2  
./run_demo.sh --stage-name prep
```

Preparation of pre-trained model download and installation of required tools

# VAD



```
./run_demo.sh --stage-name sad
```

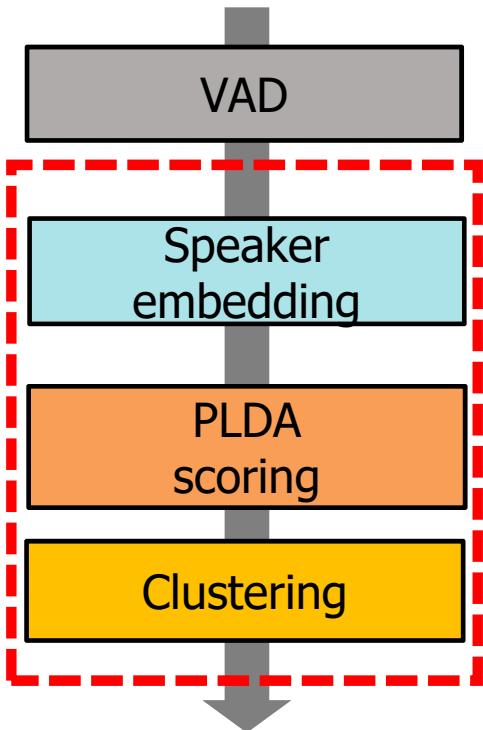
Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
0001		2.17	11.18	
0002		11.18	17.78	
0003		18.70	22.52	

# VAD

A screenshot of a terminal window with four tabs visible. The tabs are labeled from left to right: '» emacs', '⌘1', '» bash', '⌘2', '» bash', '⌘3', '» bash', and '⌘4'. The '⌘2' tab is currently active. The terminal prompt shows the user's path: 'shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b\_track2\$'. Below the path, the user has typed the command: '../run\_demo.sh --stage-name sad'. The terminal window has a dark background with light-colored text.

```
shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b_track2$ ../run_demo.sh --stage-name sad
```

# Diarization



```
./run_demo.sh --stage-name sad  
./run_demo.sh --stage-name diarize
```

Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
0001	2	2.170	3.295	
0002	3	3.295	17.780	
0003	4	18.700	22.52	

# Diarization

```
↪ emacs          #1 | ↪ bash          #2 | ↪ bash          #3 | ↪ bash          #4 | +  
shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b_track2$ ./run_demo.sh --stage-name diarize
```

# Diarization (after 30min.)

```
↳ emacs   #1 | ↳ bash   #2 | ↳ bash   #3 | ↳ bash   #4 | ↳ bash   #5 | +  
WARNING: Truncating turn overlapping non-scoring region. TURN: FILE: S02, SPEAKER: 3, ONSET: 54.850000, OFFSET: 68.350000, DUR: 13.500000  
WARNING: Truncating turn overlapping non-scoring region. TURN: FILE: S02, SPEAKER: 3, ONSET: 70.090000, OFFSET: 71.000000, DUR: 0.910000  
WARNING: Truncating turn overlapping non-scoring region. TURN: FILE: S02, SPEAKER: 3, ONSET: 73.390000, OFFSET: 74.050000, DUR: 0.660000  
WARNING: Truncating turn overlapping non-scoring region. TURN: FILE: S02, SPEAKER: 4, ONSET: 74.050000, OFFSET: 76.300000, DUR: 2.250000  
Checking for overlapping reference speaker turns...  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: P05  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: P06  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: P07  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: P08  
Checking for overlapping system speaker turns...  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: 2  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: 3  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: 4  
WARNING: Merging overlapping speaker turns. FILE: S02, SPEAKER: 5  
Scoring...  
WARNING: File "S01" missing in reference RTMs.  
WARNING: File "S01" missing in system RTMs.  
WARNING: File "S09" missing in reference RTMs.  
WARNING: File "S09" missing in system RTMs.  
WARNING: File "S21" missing in reference RTMs.  
WARNING: File "S21" missing in system RTMs.  

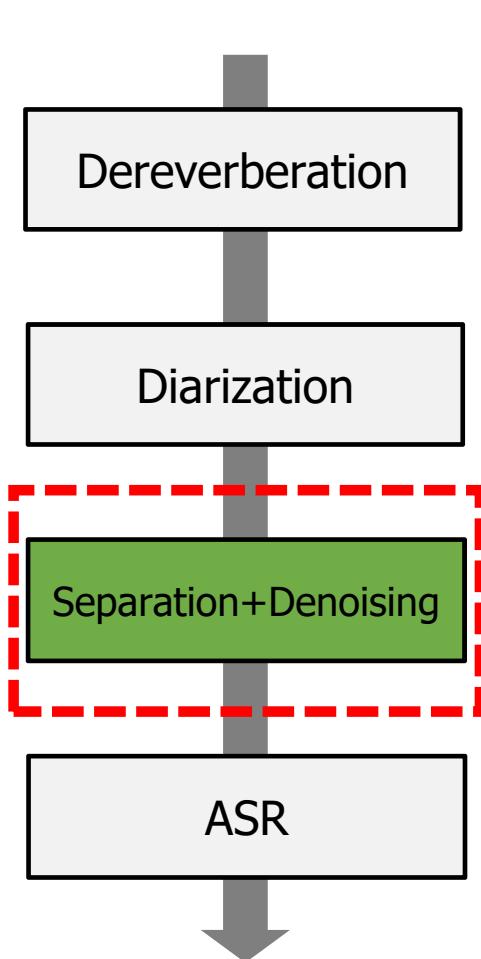

| File            | DER   | JER   | B3-Precision | B3-Recall | B3-F1 | GKT(ref, sys) | GKT(sys, ref) | H(ref sys) | H(sys ref) | MI   | NMI  |
|-----------------|-------|-------|--------------|-----------|-------|---------------|---------------|------------|------------|------|------|
| S01             | 0.00  | 0.00  | 1.00         | 1.00      | 1.00  | 1.00          | 1.00          | 0.00       | 0.00       | 0.00 | 1.00 |
| S02             | 50.83 | 57.10 | 0.41         | 0.56      | 0.47  | 0.46          | 0.32          | 2.23       | 1.49       | 1.25 | 0.40 |
| S09             | 0.00  | 0.00  | 1.00         | 1.00      | 1.00  | 1.00          | 1.00          | 0.00       | 0.00       | 0.00 | 1.00 |
| S21             | 0.00  | 0.00  | 1.00         | 1.00      | 1.00  | 1.00          | 1.00          | 0.00       | 0.00       | 0.00 | 1.00 |
| *** OVERALL *** | 50.83 | 57.10 | 0.85         | 0.89      | 0.87  | 0.86          | 0.82          | 0.56       | 0.37       | 2.31 | 0.83 |



```
local/demo/run_diarize.sh: Saved audacity labels to demo/S02.diar  
shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b_track2$
```


```

# Separation+Denoising



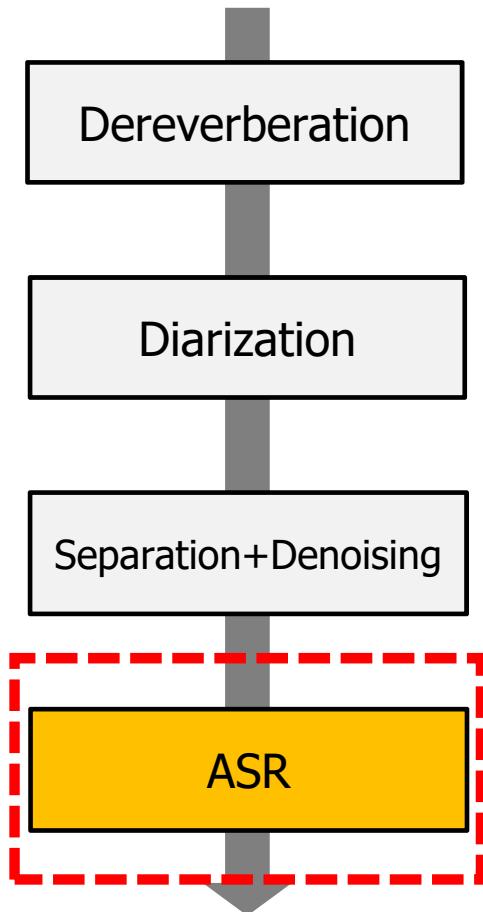
```
./run_demo.sh --stage-name sad  
./run_demo.sh --stage-name diarize  
./run_demo.sh --stage-name gss
```

# Separation+Denoising

```
↪ emacs          #1 | ↪ bash          #2 | ↪ bash          #3 | ↪ bash          #4 | +  
shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b_track2$ ./run_demo.sh --stage-name gss
```

# Separation+Denoising

# ASR



```
./run_demo.sh --stage-name sad  
./run_demo.sh --stage-name diarize  
./run_demo.sh --stage-name gss  
./run_demo.sh --stage-name asr
```

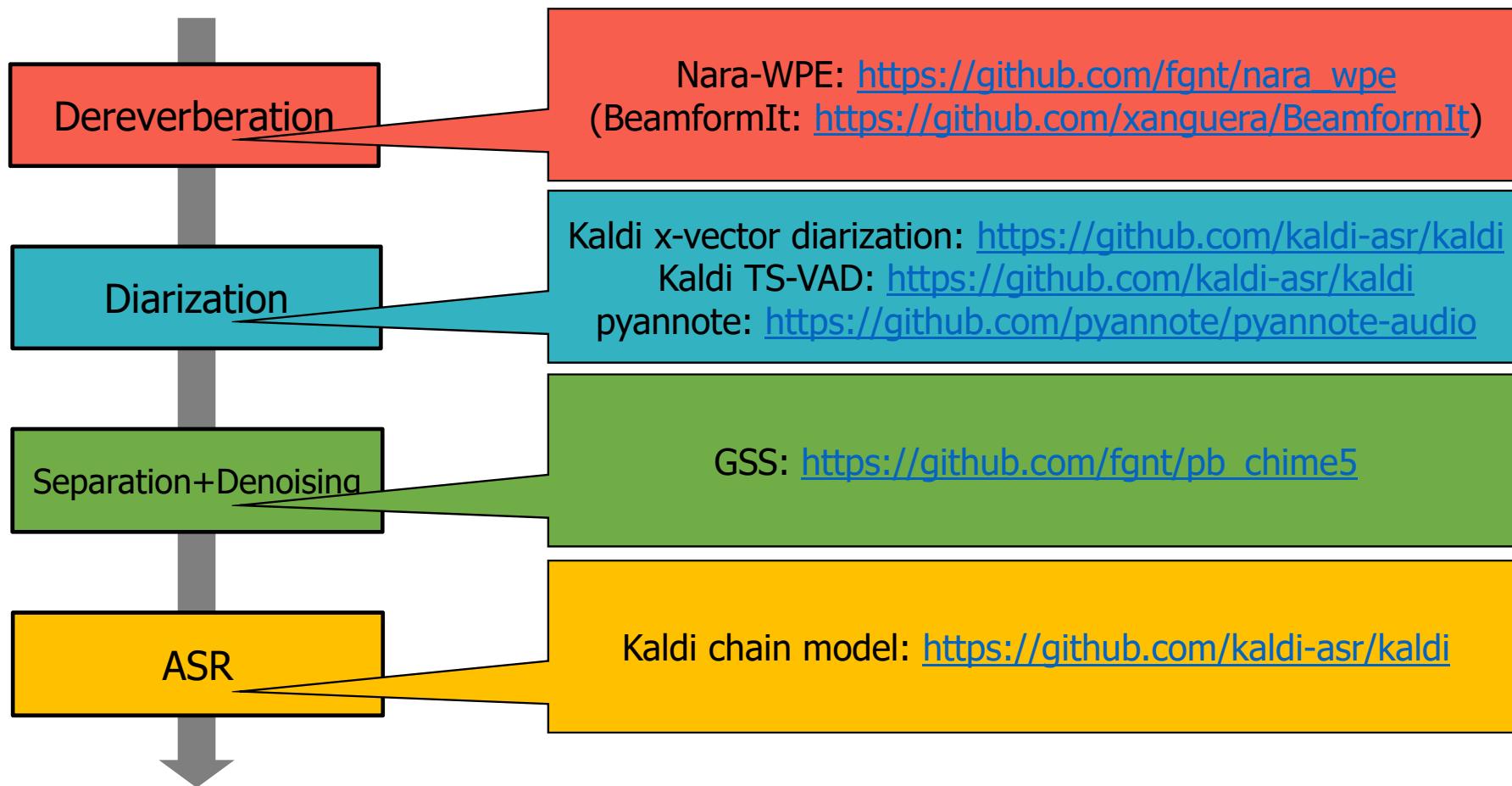
Segment ID $\tau$	Speaker Cluster $k$	Onset (sec) $b(\tau)$	Offset (sec) $e(\tau)$	Transcription $W_\tau$
0001	2	2.170	3.295	can i can do yours the pie
0002	3	3.295	17.780	yeah you have to whisk it though right
0003	4	18.700	22.52	[laughs]

# ASR

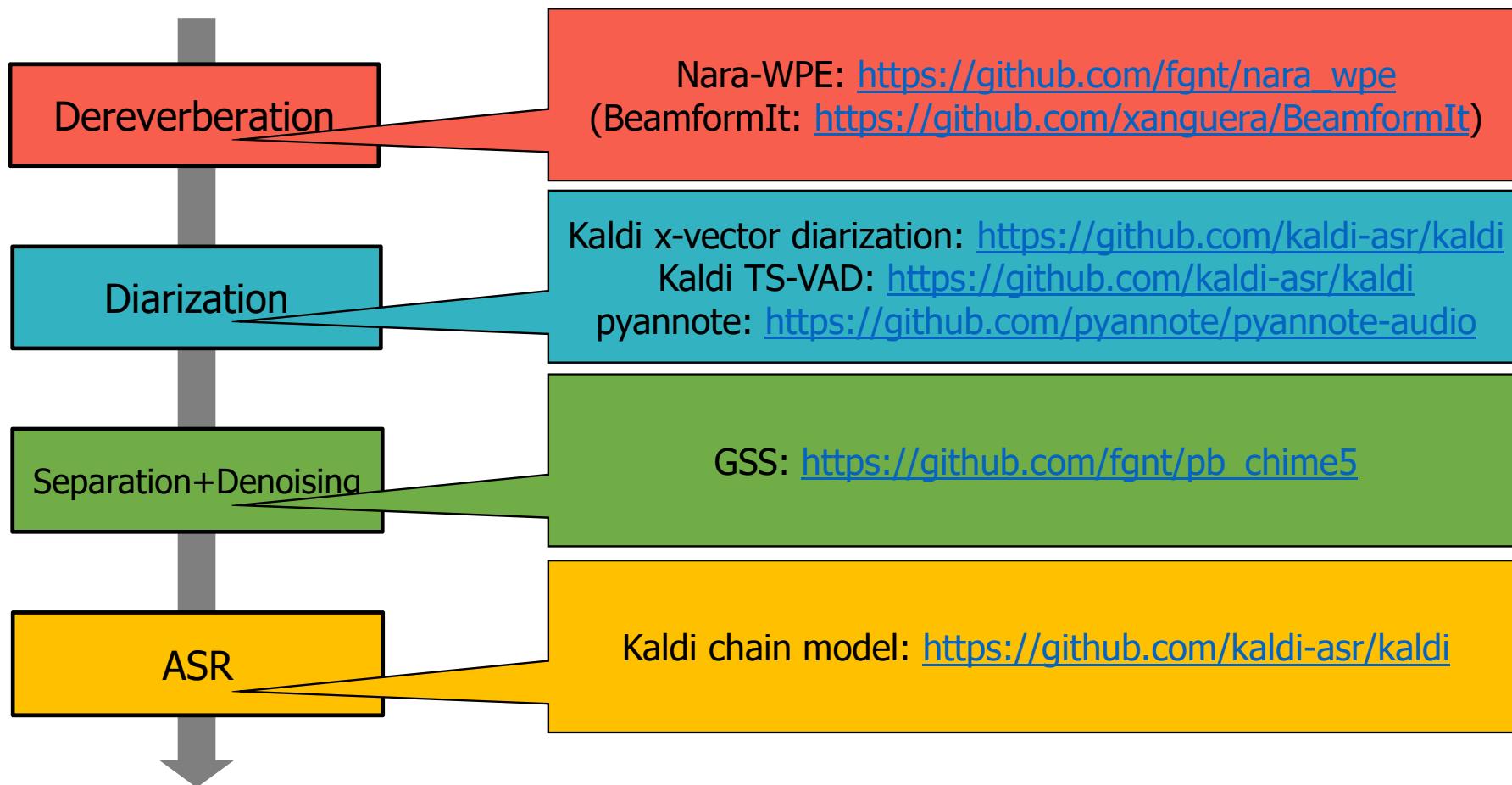
```
↪ emacs   #1 | ↪ bash   #2 | ↪ bash   #3 | ↪ bash   #4 | ↪ bash   #5 | +  
shinji@b08:/export/b08/shinji/202004jsalt20/kaldi/egs/chime6/s5b_track2$ ./run_demo.sh --stage-name asr
```



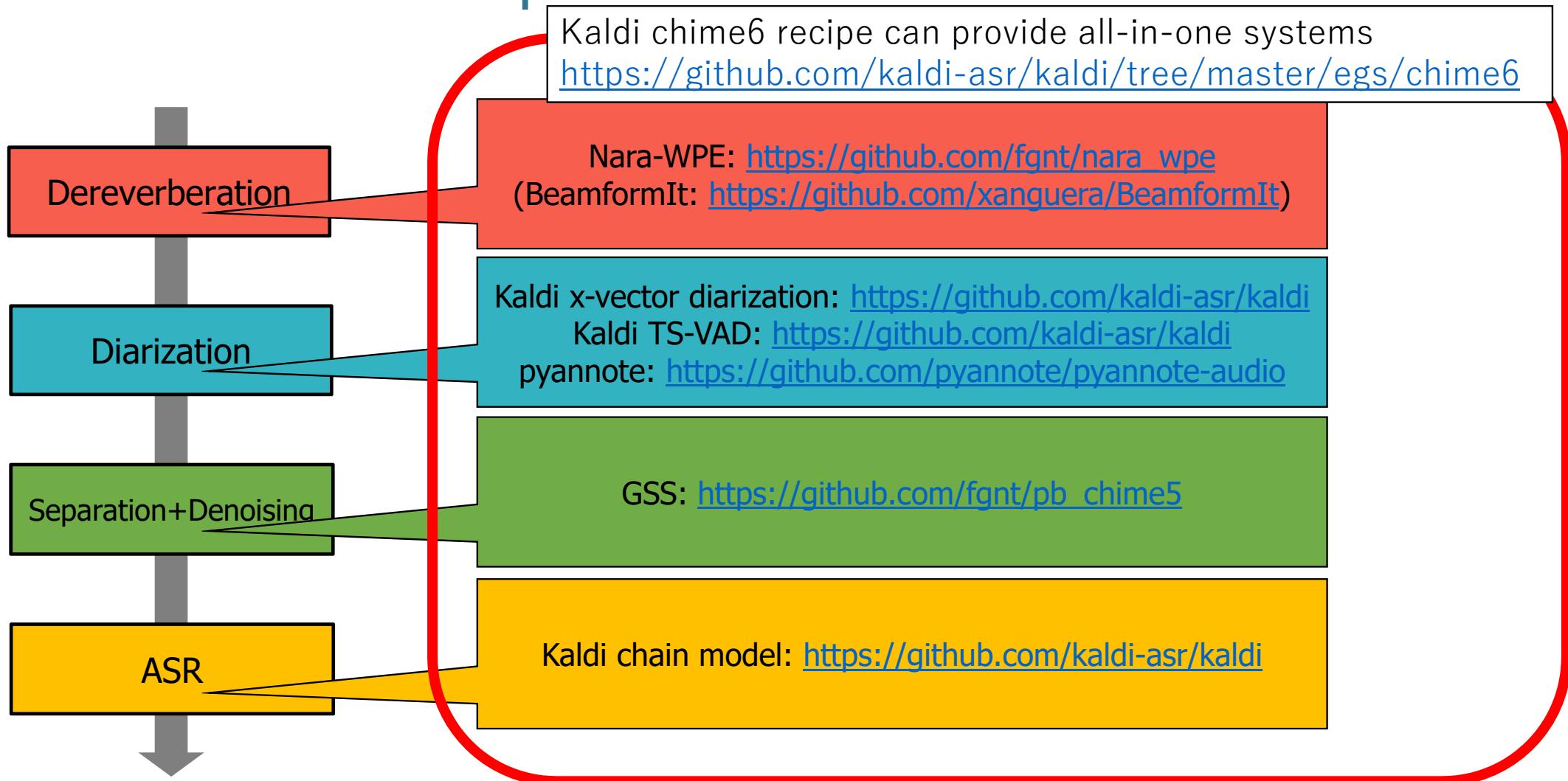
# All are based on open source



# All are based on open source



# All are based on open source



# Discussions

**We have a great step toward distant conversational speech recognition and analysis**

- The performance is significantly improved from the progress of each module
- Database and reproducible open-source systems (CHiME-6 as an example)

**The system is based on independent optimization**

- Each module is optimized with different models, objectives, and tools

**Jointly optimal systems**

# Agenda

## Part 1: Introduction

- 1.1. What is distant conversational ASR and analysis?
- 1.2. Why is it difficult?
- 1.3. Why is it important?
- 1.4. Its research history
- 1.5. Typical systems for distant conversational ASR and analysis

} 25min

} 5min Q&A

} 25min

**15min Q&A and Break**

} 25min + 5min Q&A

} 25min + 5min Q&A

} 25min + 5min Q&A

## Part 2: Current state-of-the-art systems

- 2.1. Descriptions of the techniques
- 2.2. Reproducible baselines

## Part 3: A new research trend: Jointly optimal systems

- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$  ( $x$ : other functionalities)
- 3.3. ASR +  $x$

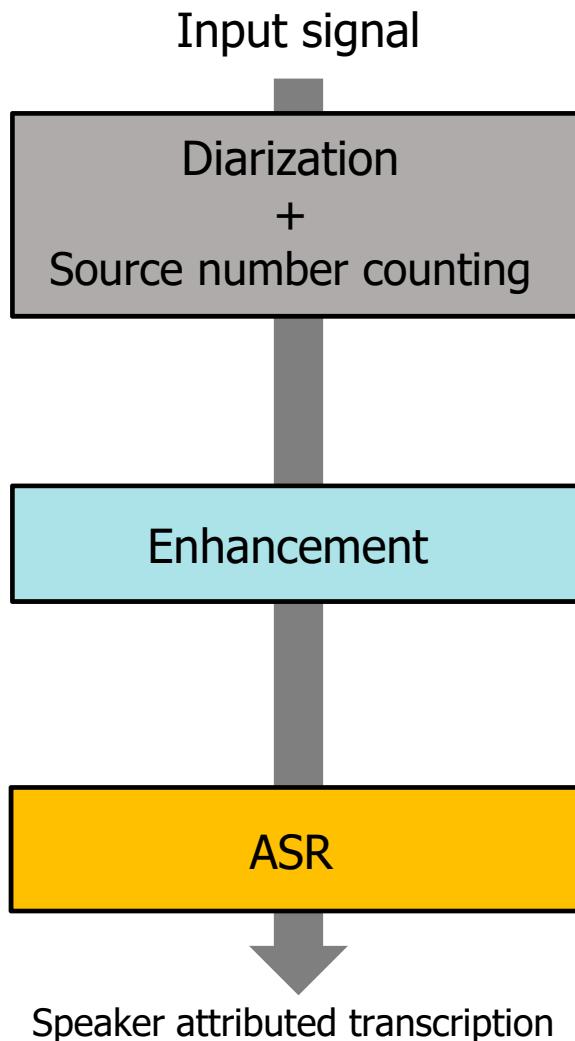
} 5min + ?-min Q&A

## Part 4: Summary and discussion

- 4.1. Strengths of each approach
- 4.2. Challenges and fundamental difficulties

### 3. A new research trend: Jointly optimal systems

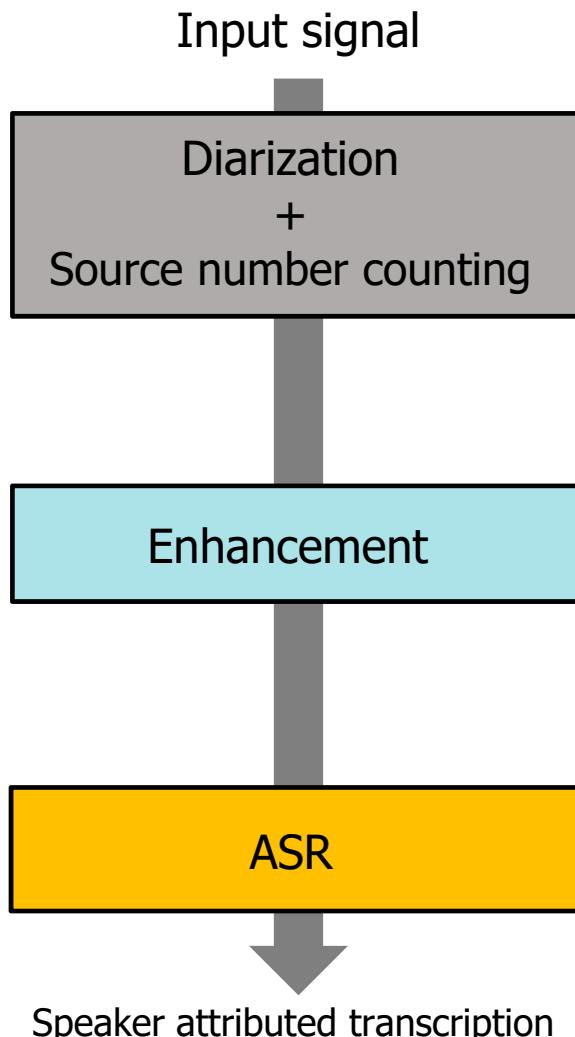
# A typical pipeline assumed in challenges like CHiME-6



- Works for real data  
- Large room for improvement

Can we do any better?

# A typical pipeline assumed in challenges like CHiME-6



All modules are optimized independently, and thus each module doesn't know what's going on in the other modules, i.e., the system is not jointly optimal.



Trend toward jointly optimal systems

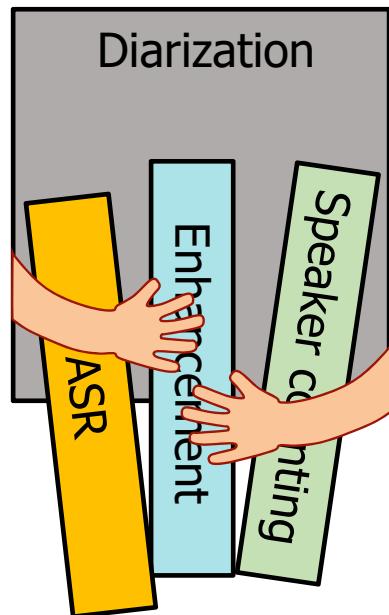
### 3. A new research trend: Jointly optimal systems

- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$
- 3.3. ASR +  $x$

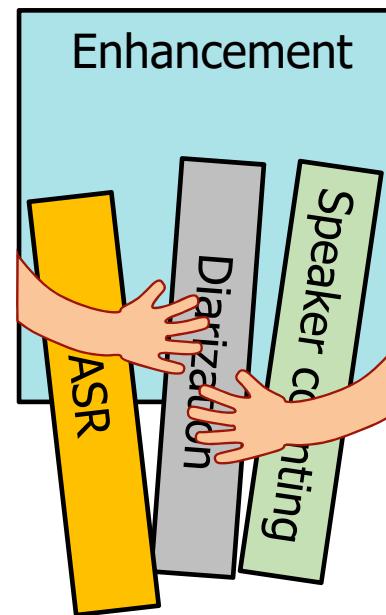
# Approaches for optimal distant conversational ASR and analysis

- Current approaches can be roughly **categorized** into the following three.
- Currently proposed joint systems can be seen as **extensions** of a core module.
- Each approach has **different strength**. (This will be discussed in Part 4.)

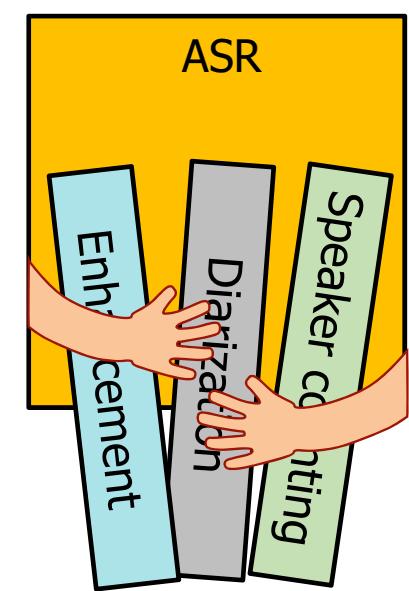
Diarization-originated approach



Enhancement-originated approach



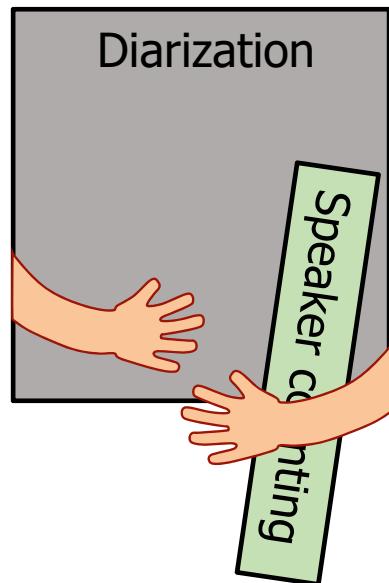
ASR-originated approach



# What has been achieved so far?

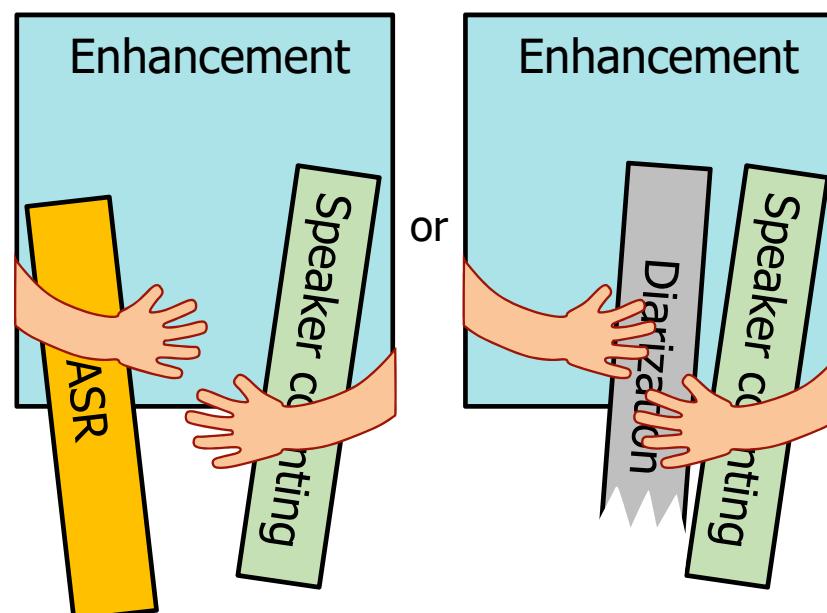
- None of them has yet achieved fully optimal distant conversational ASR and analysis.
- Their attainment levels are also different.
- Not yet clear which approach is most advantageous in the end.

Diarization-originated approach



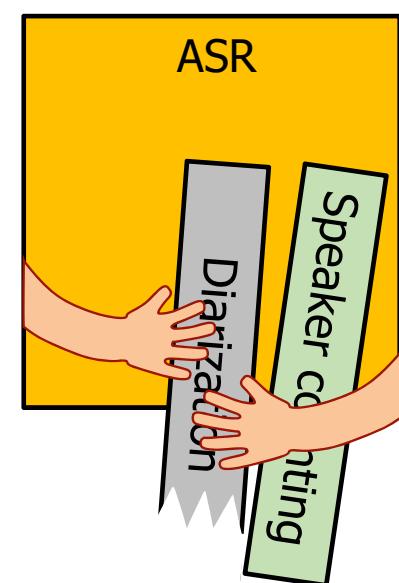
(Explained in 3.1)

Enhancement-originated approach



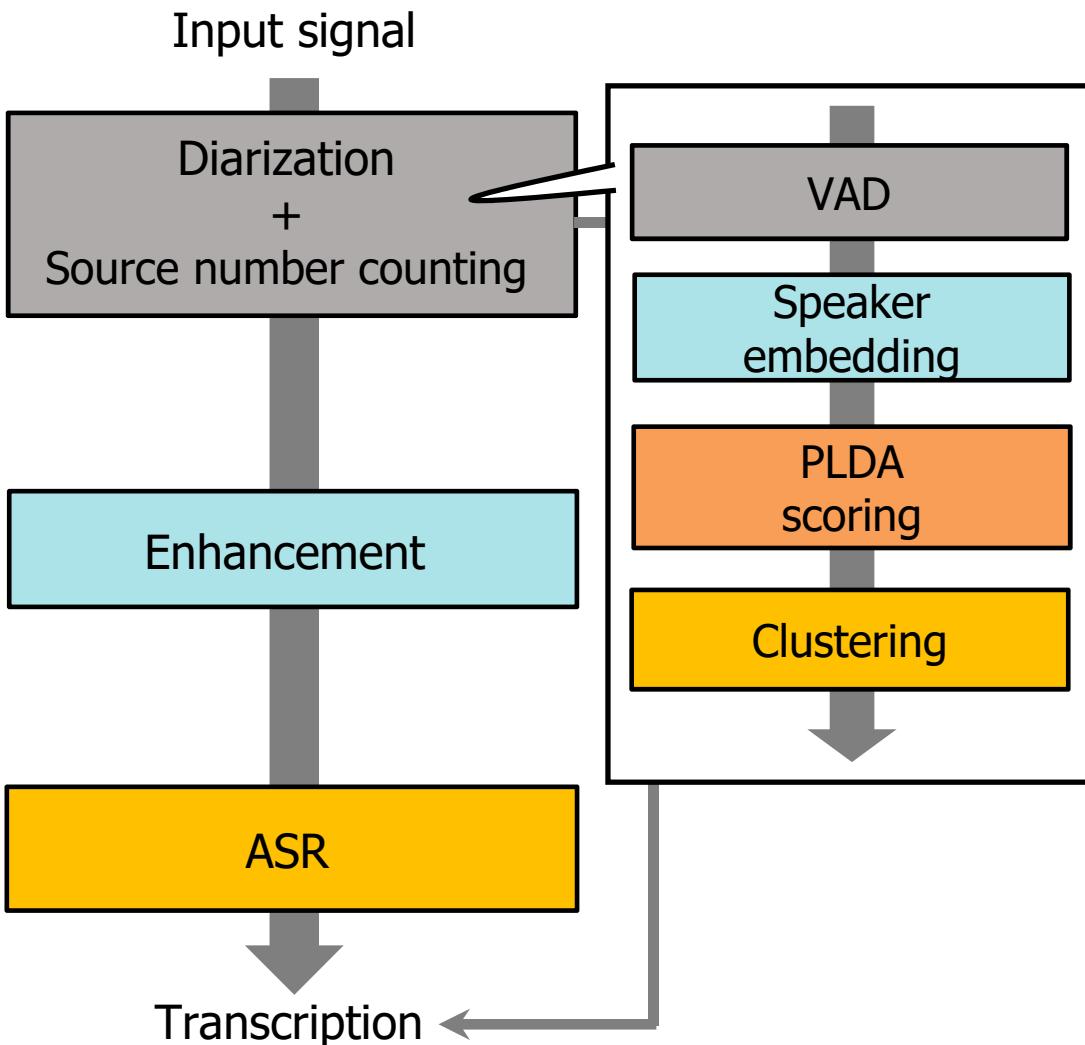
(Explained in 3.2)

ASR-originated approach



(Explained in 3.3)

## Section 3.1: Diarization + $x$



⌚ Before talking about the joint optimization, the problem is the **traditional diarization pipeline contains many independent modules.**

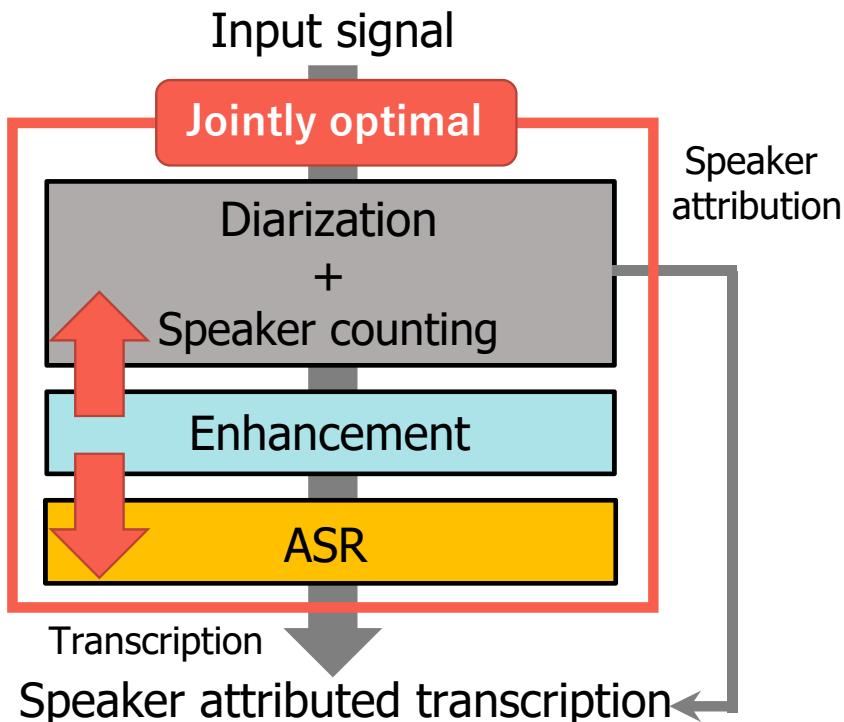
→ Hard to optimize diarization systems  
(a) for the diarization error minimization  
(b) for combining it with other modules.



A new trend towards neural diarization!

## Section 3.2: Enhancement + $x$

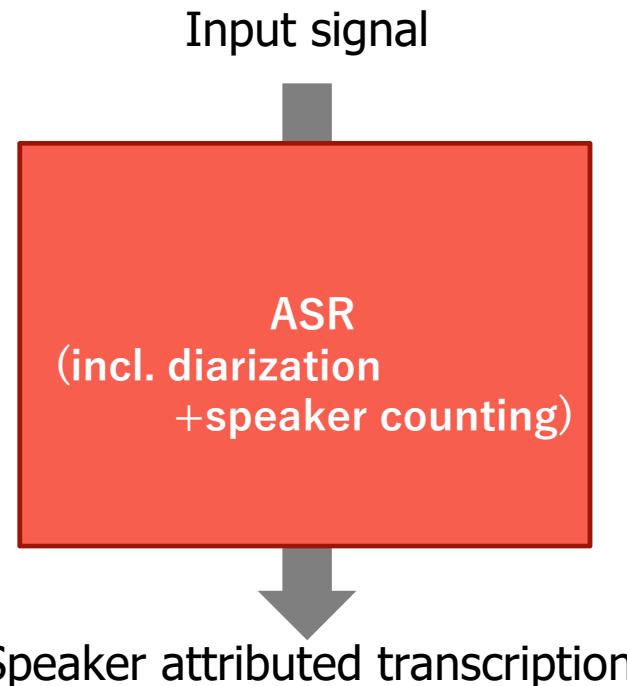
Trying to realize  
jointly optimal modular systems,  
by expanding enhancement module



- Core system: Neural separation system
- Integration of the neural separation with
  - (a) ASR
  - (b) Multi-channel enhancement + ASR
  - (c) Speaker counting + Diarization

## Section 3.3: ASR + $x$

Trying to realize  
a monolithic optimal system,  
by extending the ASR module



- Core system: End-to-end ASR
- Integration of the end-to-end ASR with
  - (a) Speaker counting
  - (b) Diarization

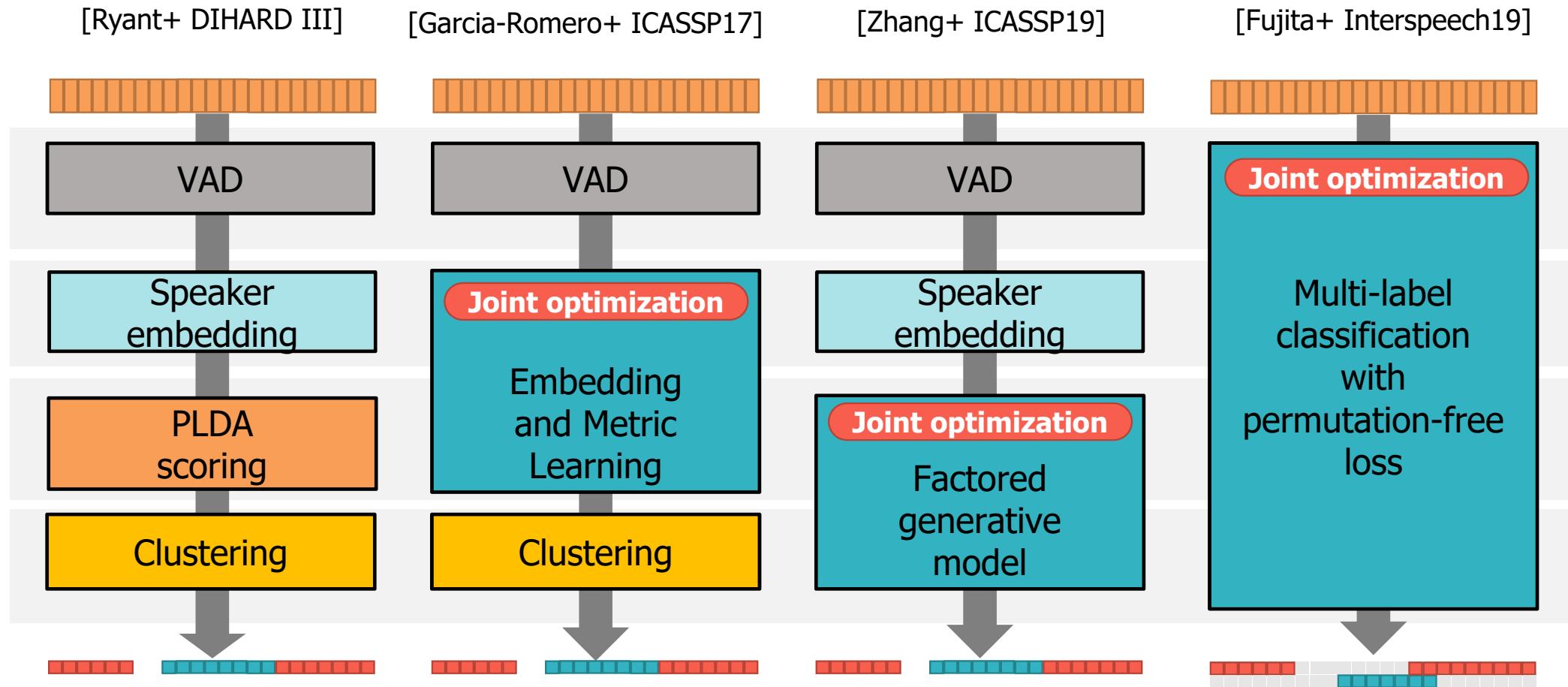
### 3. A new research trend: Jointly optimal systems

- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$
- 3.3. ASR +  $x$

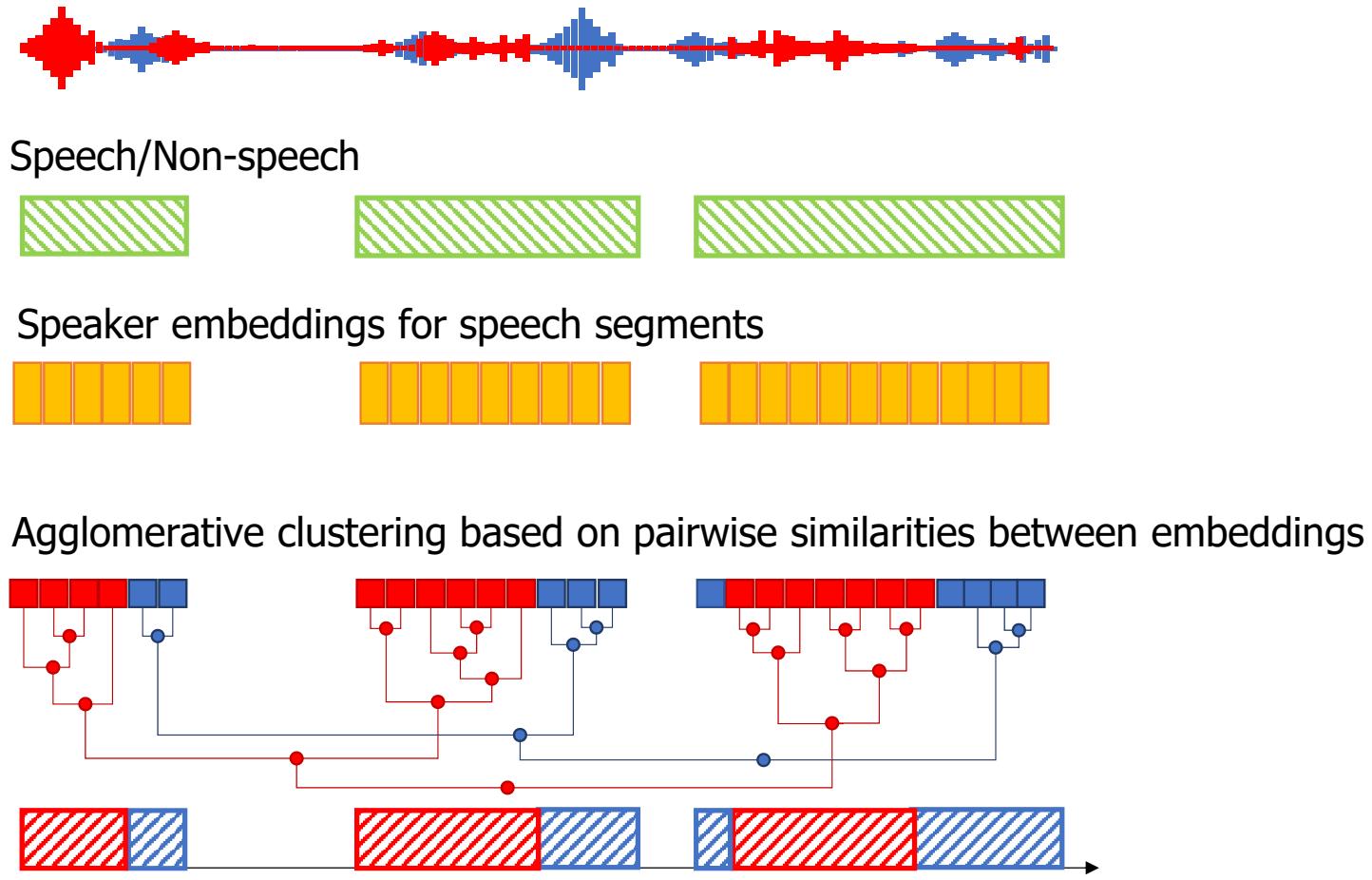
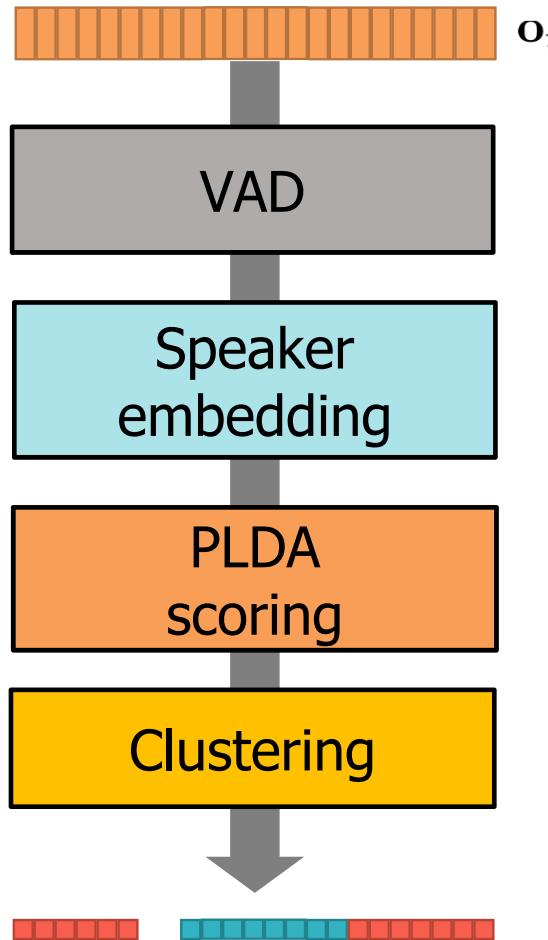
### 3.1 Diarization

From clustering-based pipeline to end-to-end optimal systems

# From Clustering-based to End-to-End

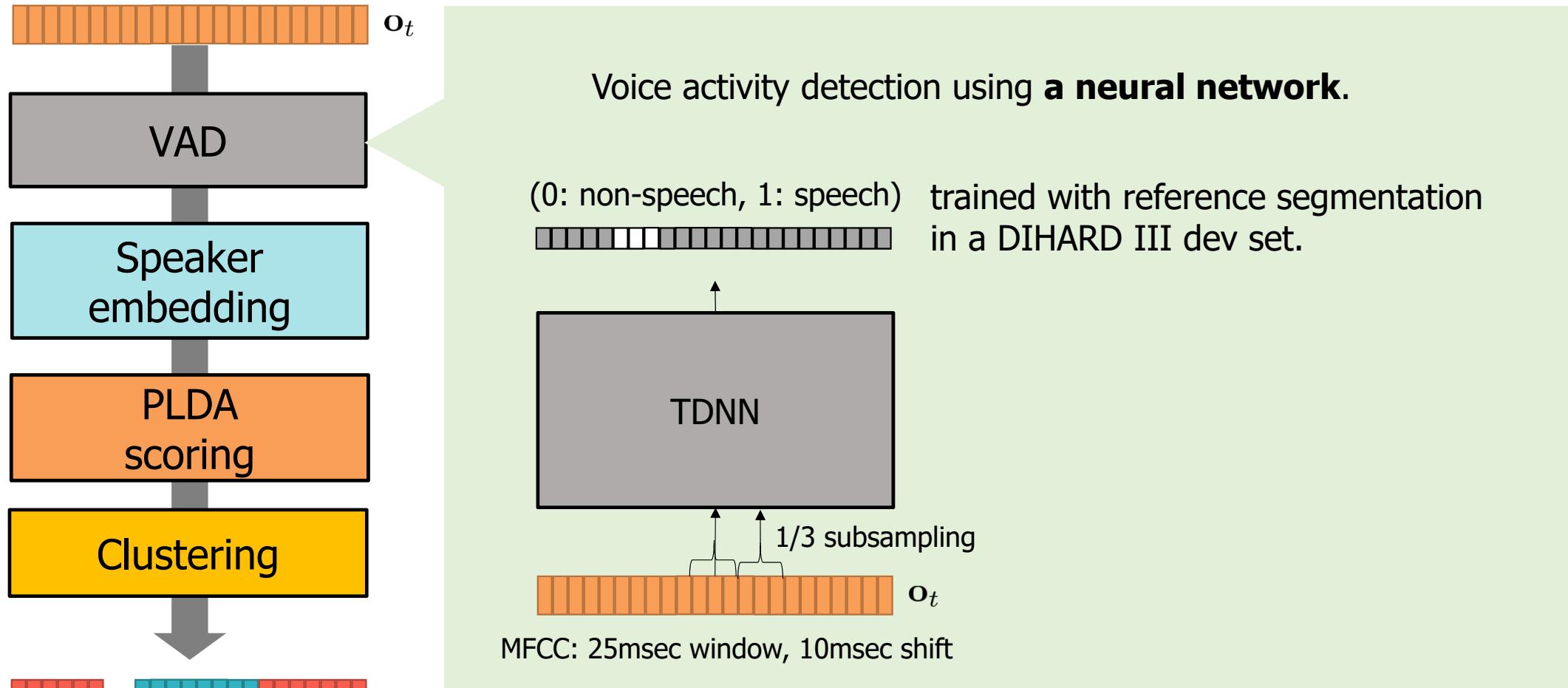


# Typical Clustering-based Diarization



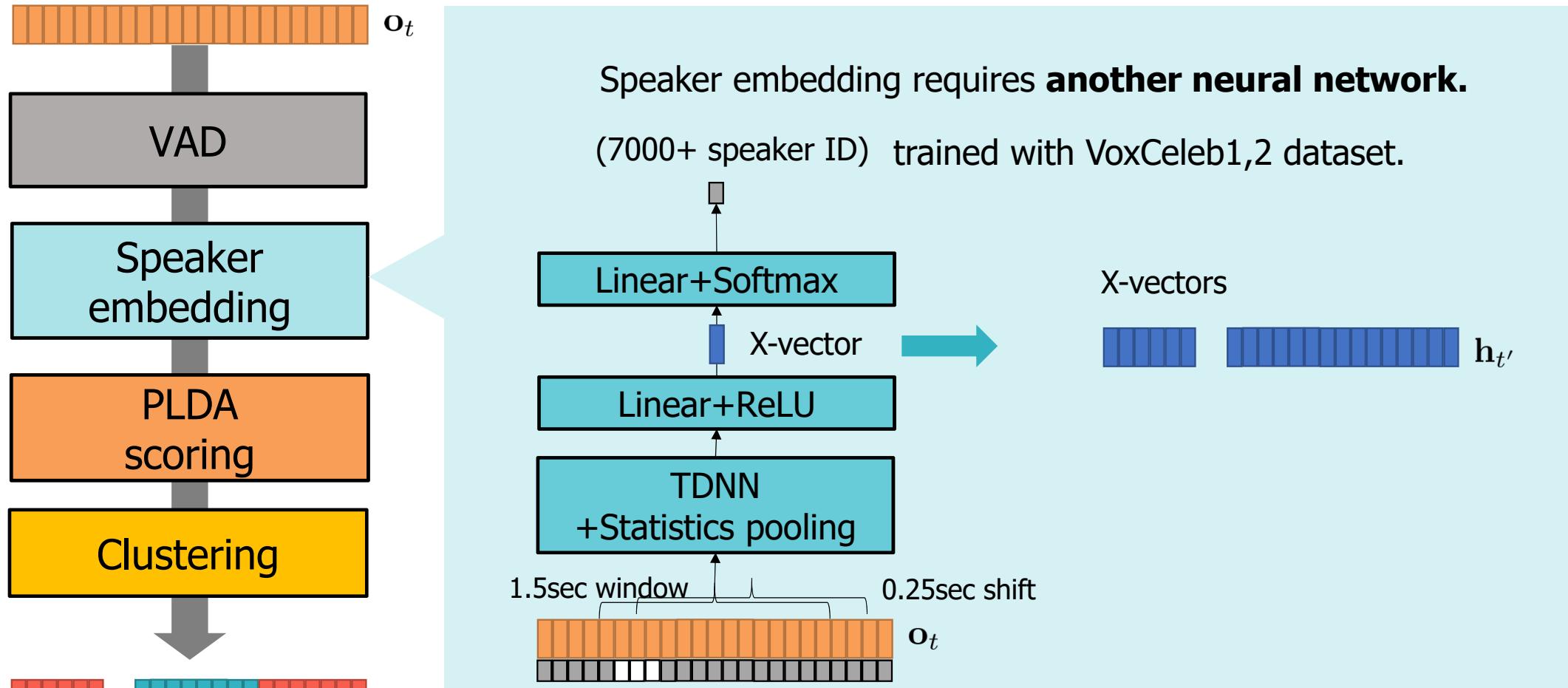
N. Ryant et al., "The Third DIHARD Diarization Challenge," arXiv 2021.

# Typical Clustering-based Diarization



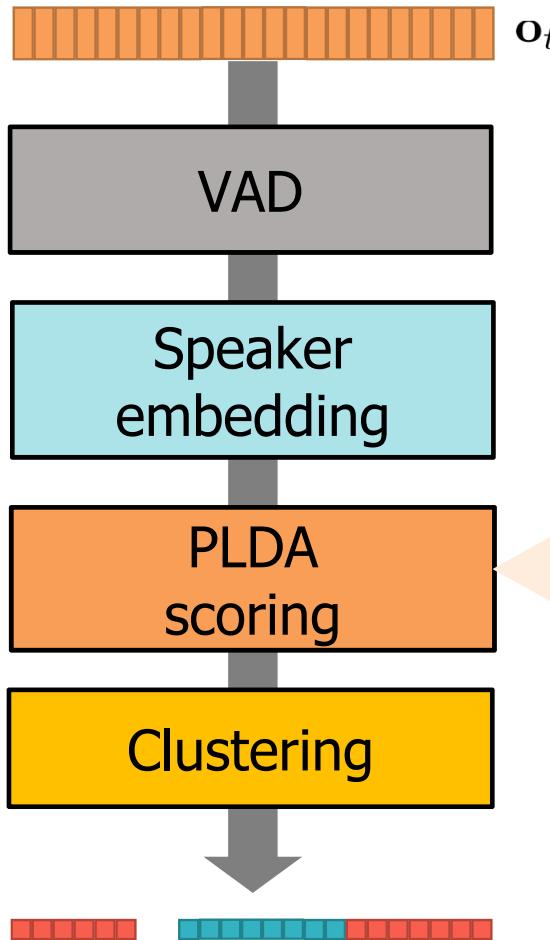
N. Ryant et al., "The Third DIHARD Diarization Challenge," arXiv 2021.

# Typical Clustering-based Diarization



N. Ryant et al., "The Third DIHARD Diarization Challenge," arXiv 2021.

# Typical Clustering-based Diarization



Further, the system requires **another scoring model to be trained.**



$$\text{LLR}(\mathbf{h}_{t_1}, \mathbf{h}_{t_2}) = \log \frac{p(\mathbf{h}_{t_1}, \mathbf{h}_{t_2} | H_{\text{same}})}{p(\mathbf{h}_{t_1}, \mathbf{h}_{t_2} | H_{\text{diff}})}$$

$H_{\text{same}}$  : Hypothesis that two embeddings belong to the same speaker

$$p(\mathbf{h}_{t_1}, \mathbf{h}_{t_2} | H_{\text{same}}) = \int_{\beta} p(\mathbf{h}_{t_1} | \beta) p(\mathbf{h}_{t_2} | \beta) d\beta$$

$H_{\text{diff}}$  : Hypothesis that two embeddings are from different speakers

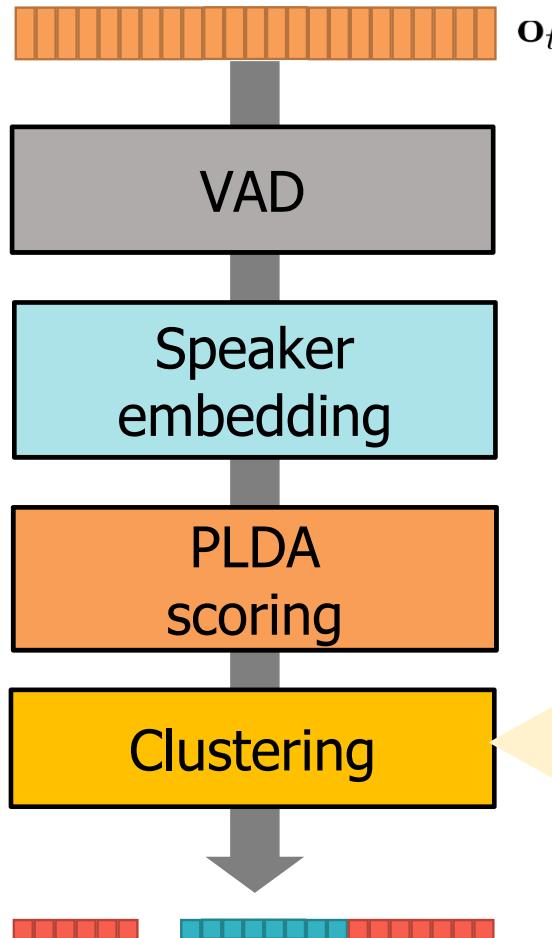
$$p(\mathbf{h}_{t_1}, \mathbf{h}_{t_2} | H_{\text{diff}}) = \int_{\beta_1} p(\mathbf{h}_{t_1} | \beta_1) d\beta_1 \int_{\beta_2} p(\mathbf{h}_{t_2} | \beta_2) d\beta_2$$

The PLDA model is trained with VoxCeleb1,2 dataset.

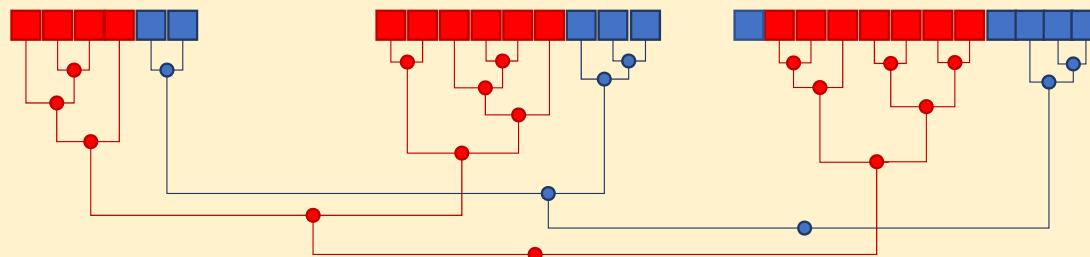
This log-likelihood ratio, known as **PLDA score**, is used for clustering.

N. Ryant et al., "The Third DIHARD Diarization Challenge," arXiv 2021.

# Typical Clustering-based Diarization



Lastly, the system uses **unsupervised method** for grouping speakers.

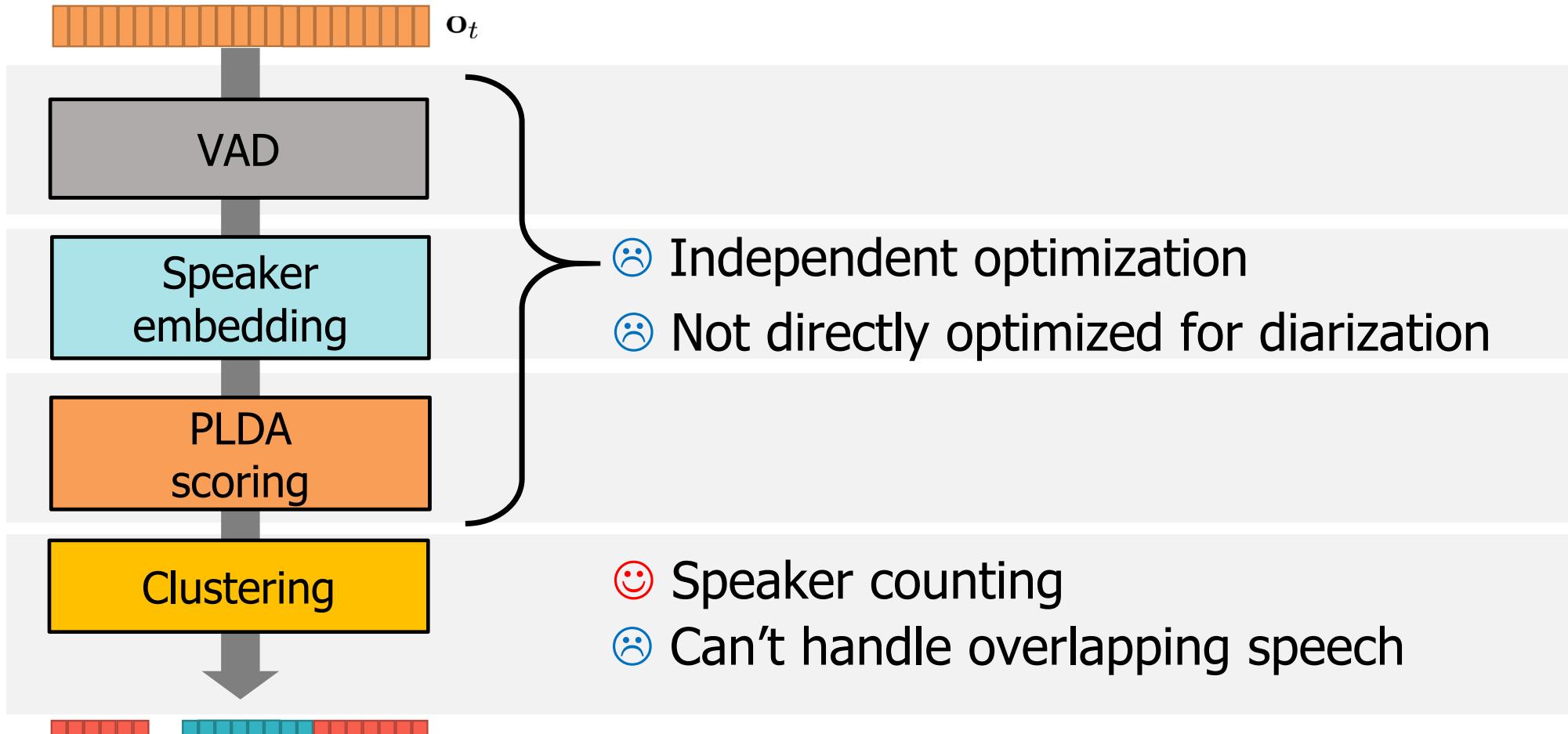


## Agglomerative Hierarchical Clustering

Iteratively merge a pair of clusters that has the highest PLDA score, until the score met the threshold.  
The threshold is tuned to minimize DER in DIHARD III dev set.

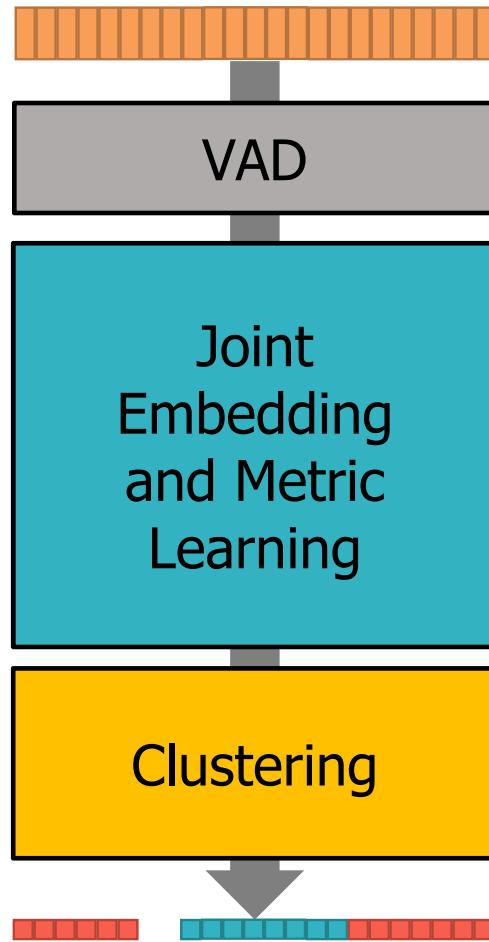
N. Ryant et al., “The Third DIHARD Diarization Challenge,” arXiv 2021.

# Typical Clustering-based Diarization



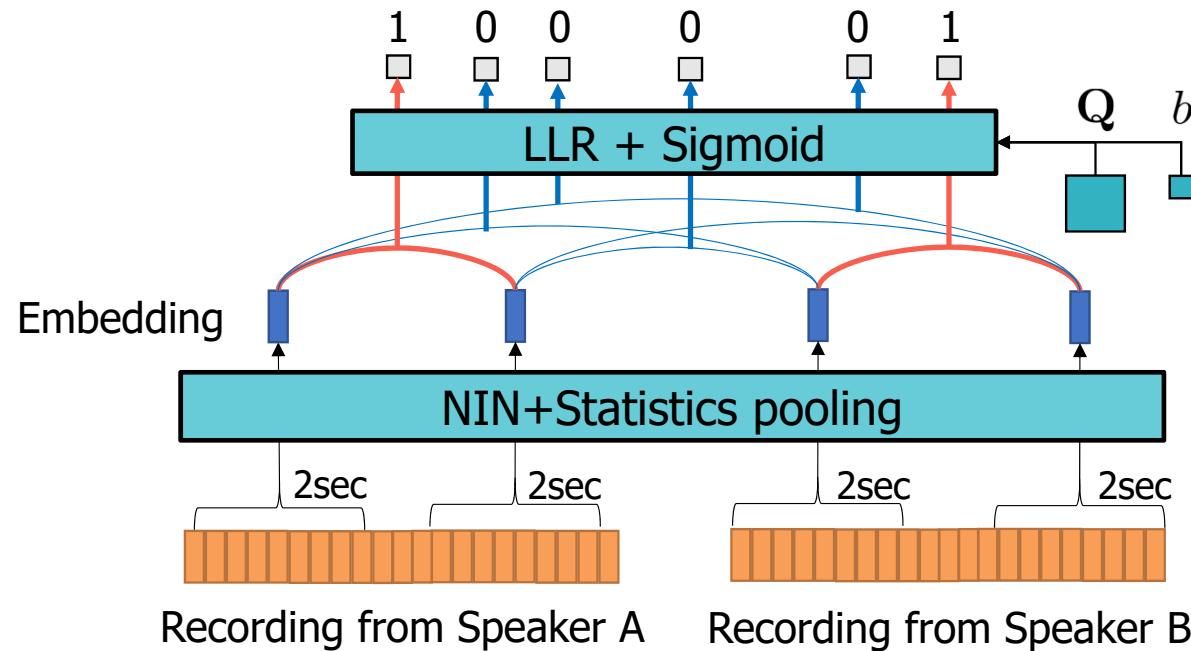
N. Ryant et al., "The Third DIHARD Diarization Challenge," arXiv 2021.

# Joint Embedding and Metric Learning



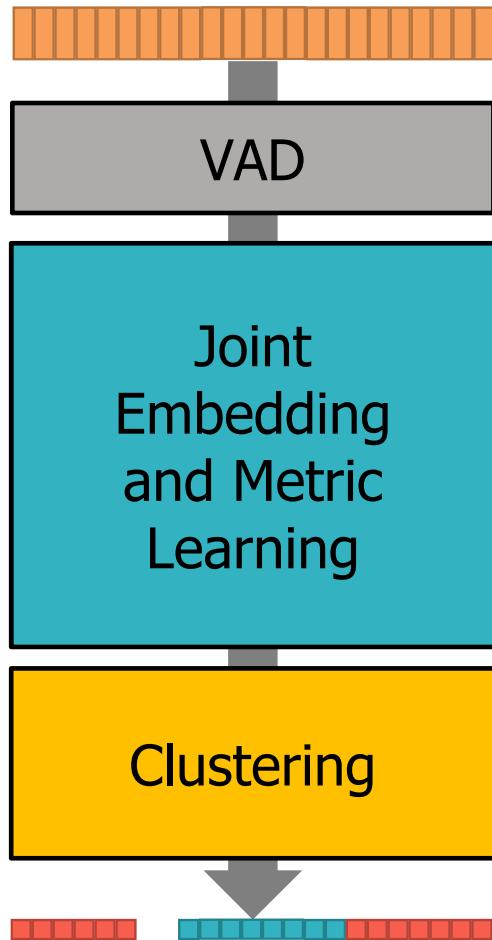
Speaker embedding and PLDA scoring matrix are jointly trained.

$$\text{LLR}(\mathbf{h}_{t_1}, \mathbf{h}_{t_2}) = \mathbf{h}_{t_1}^\top \mathbf{Q} \mathbf{h}_{t_1} + \mathbf{h}_{t_2}^\top \mathbf{Q} \mathbf{h}_{t_2} + \mathbf{h}_{t_1}^\top \mathbf{h}_{t_2} + b$$



D. Garcia-Romero et al., "Speaker diarization using deep neural network embeddings," ICASSP 2017.

# Joint Embedding and Metric Learning



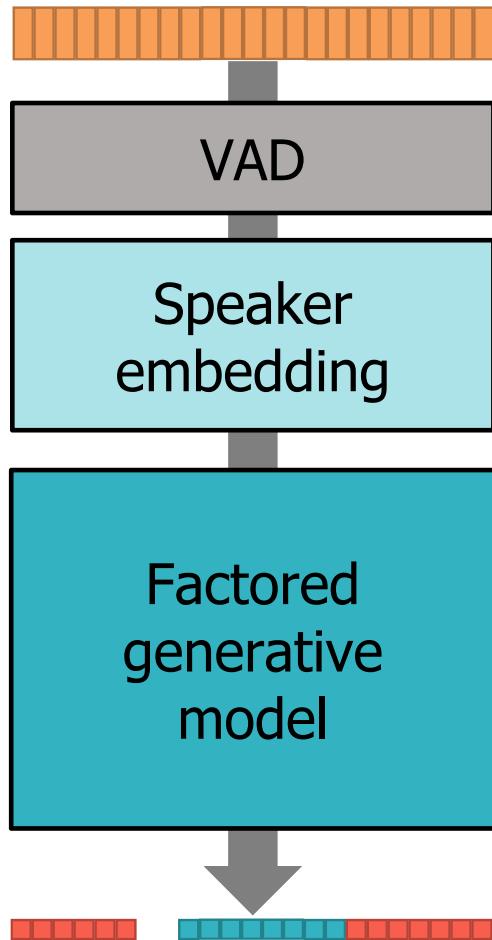
DERs on CALLHOME 2-7 speakers **without VAD or overlap errors.**

Speaker Embedding	Scoring	DER (%)
i-vector	PLDA	11.2
Senone DNN	PLDA	10.9
<b>Joint Embedding and Metric Learning</b>		<b>9.9</b>

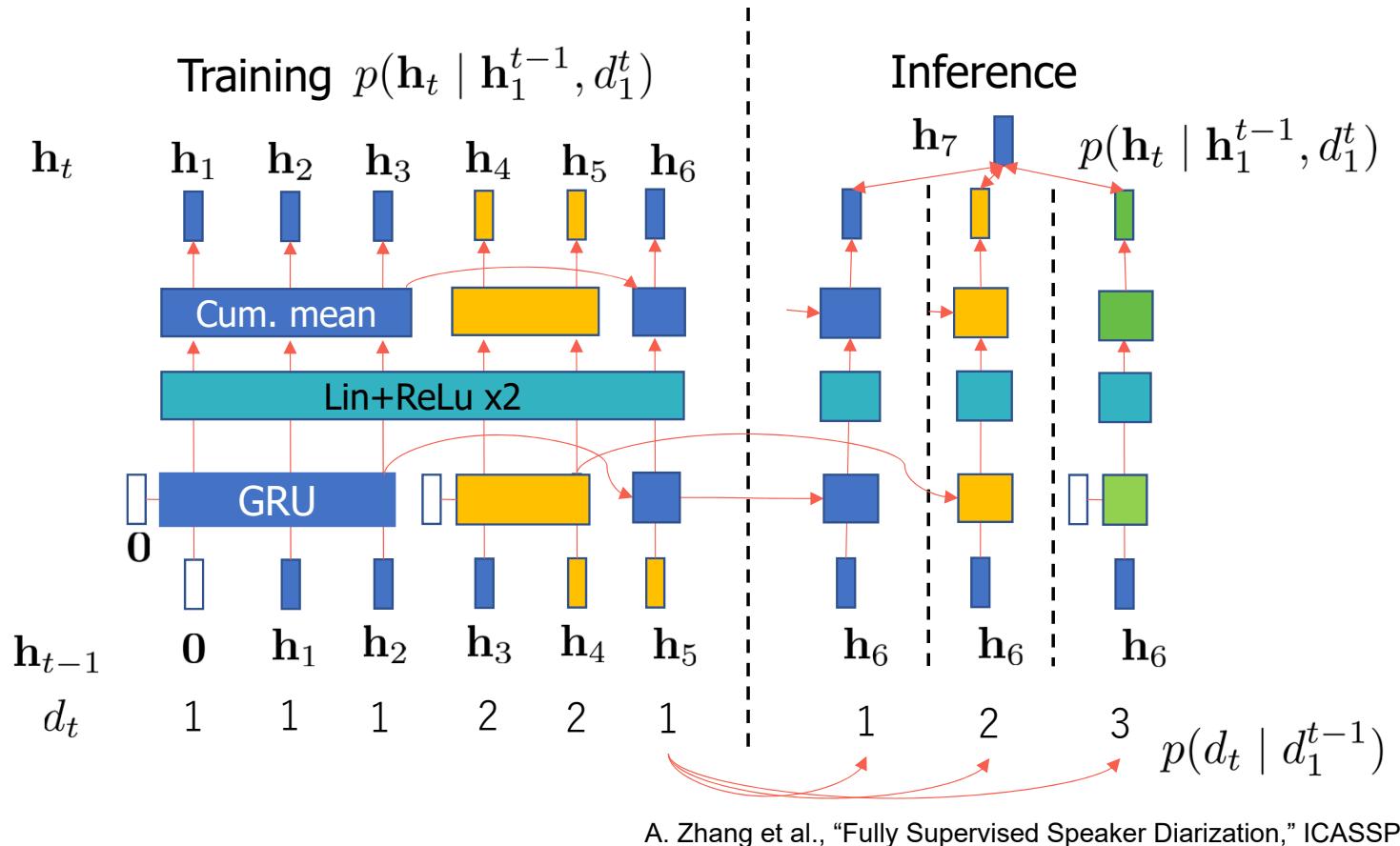
- 😊 Joint optimization
- 😊 Speaker counting
- 😢 Not directly optimized for diarization
- 😢 Can't handle overlapping speech

D. Garcia-Romero et al., "Speaker diarization using deep neural network embeddings," ICASSP 2017.

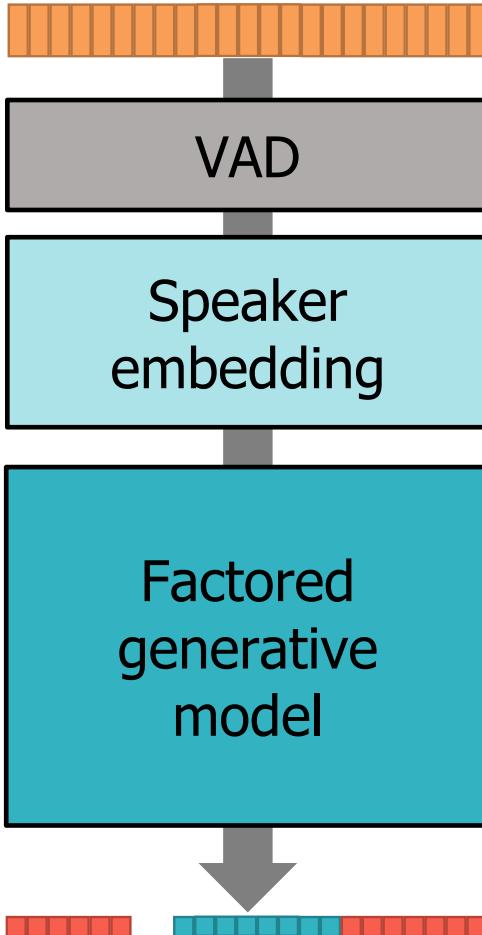
# Fully-supervised Speaker Diarization



UIS-RNN is fully-supervised with speaker embeddings (d-vectors) and corresponding diarization.



# Fully-supervised Speaker Diarization



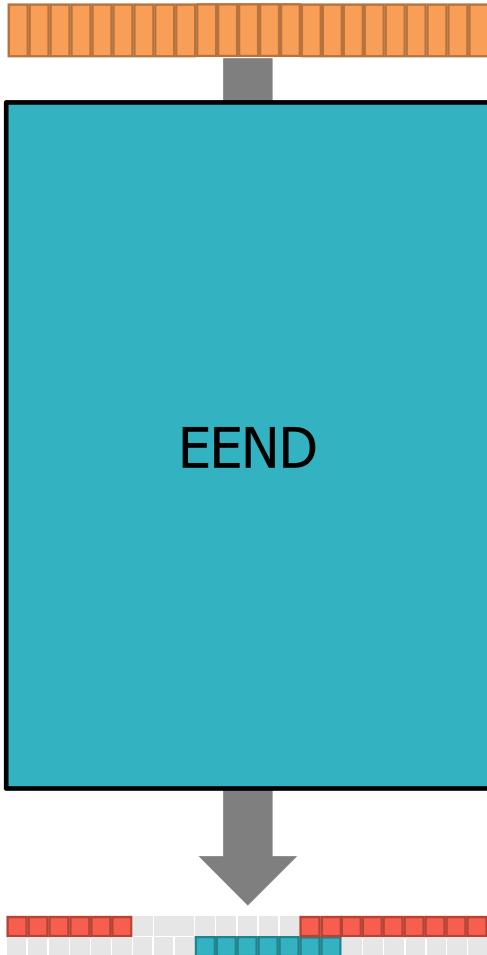
DERs on CALLHOME 2-7 speakers **without VAD or overlap errors.**

Speaker Embedding	Method	DER (%)
d-vector V3	K-means clustering	12.3
d-vector V3	Spectral clustering	8.8
<b>d-vector V3</b>	<b>UIS-RNN</b>	<b>7.6</b>

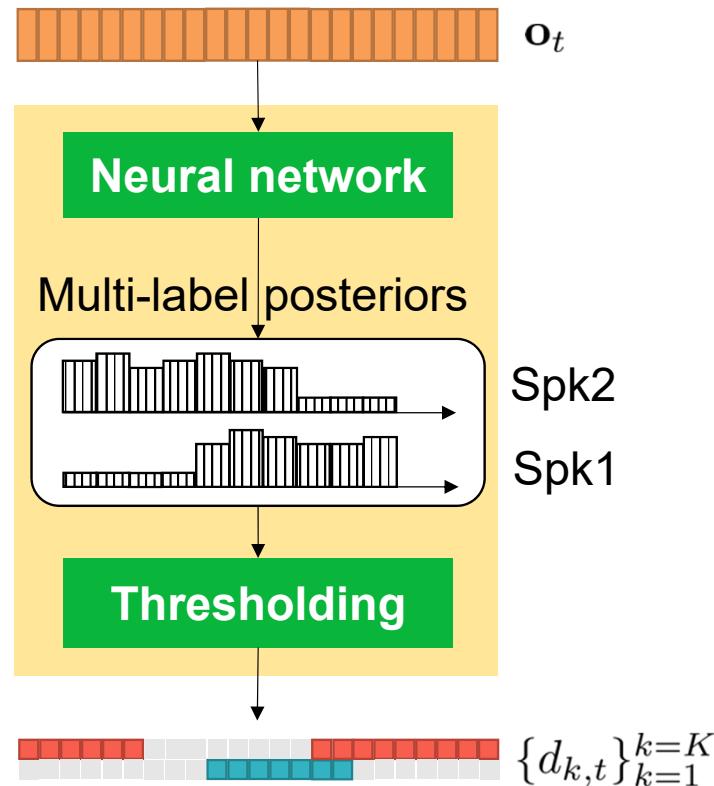
- 😊 Fully-supervised
- 😊 Speaker counting + Online
- 😢 Can't handle overlapping speech

A. Zhang et al., "Fully Supervised Speaker Diarization," ICASSP 2019.

# End-to-End Neural Diarization (EEND)

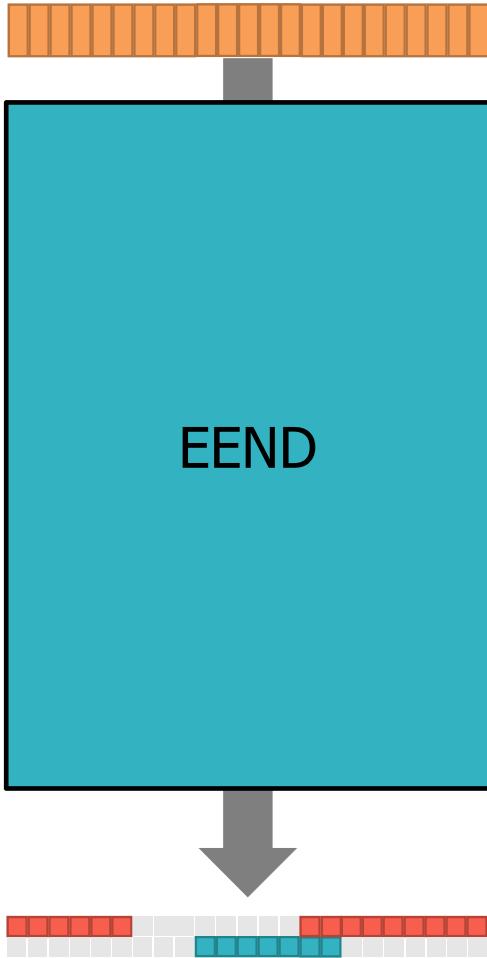


Direct estimation of diarization using **multi-label classification**



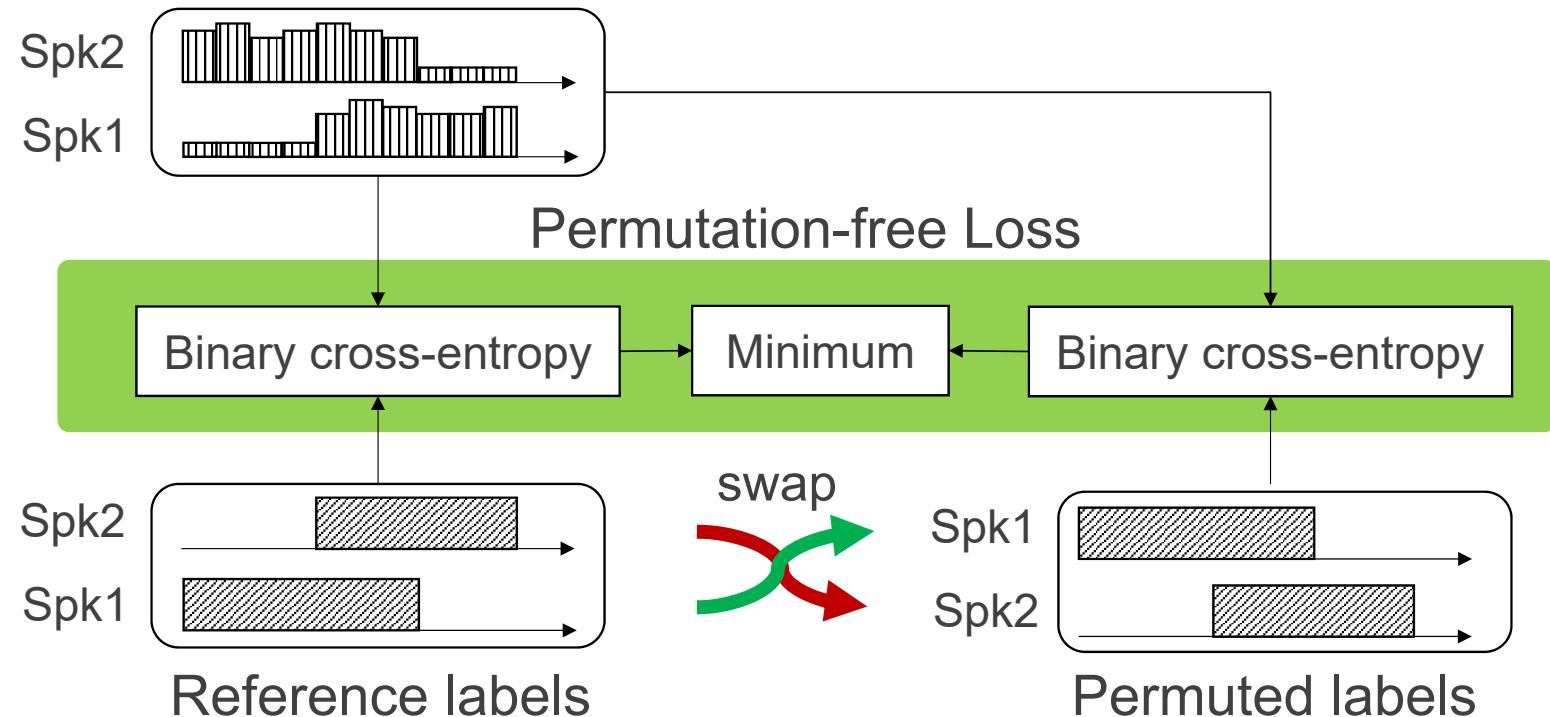
Fujita et al., "End-to-End Neural Speaker Diarization with Permutation-Free Objectives," Interspeech 2019.

# End-to-End Neural Diarization (EEND)



Permutation-free loss

Multi-label posteriors

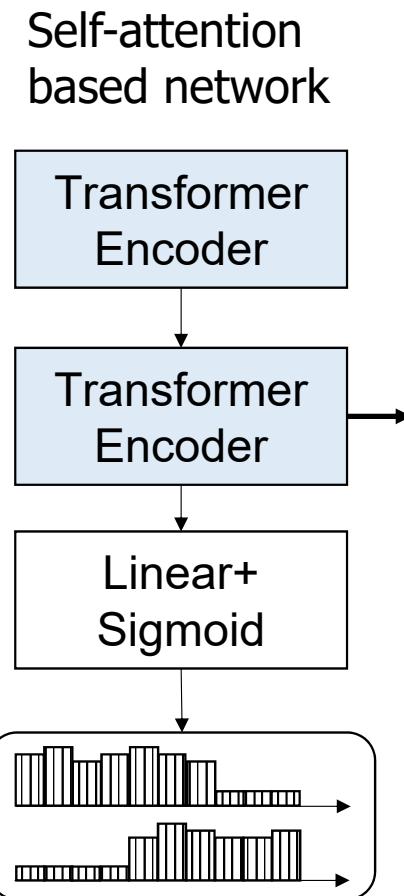
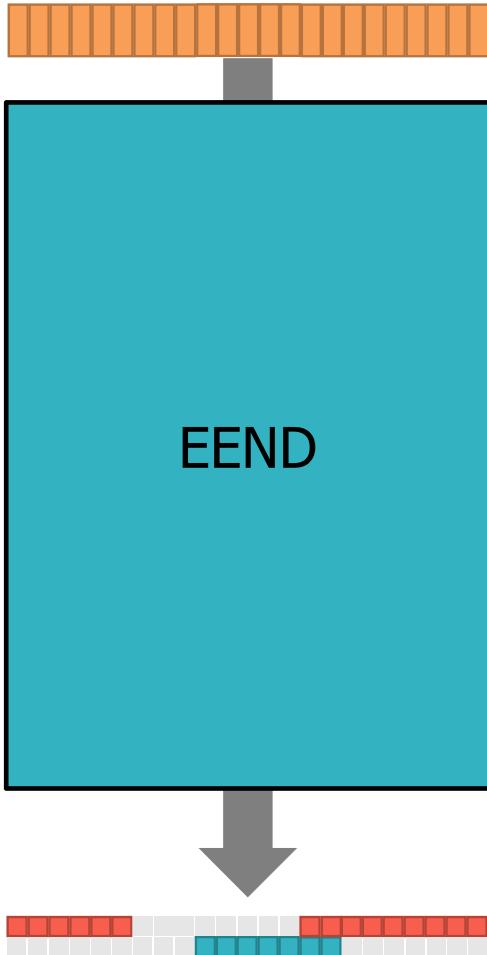


Reference labels

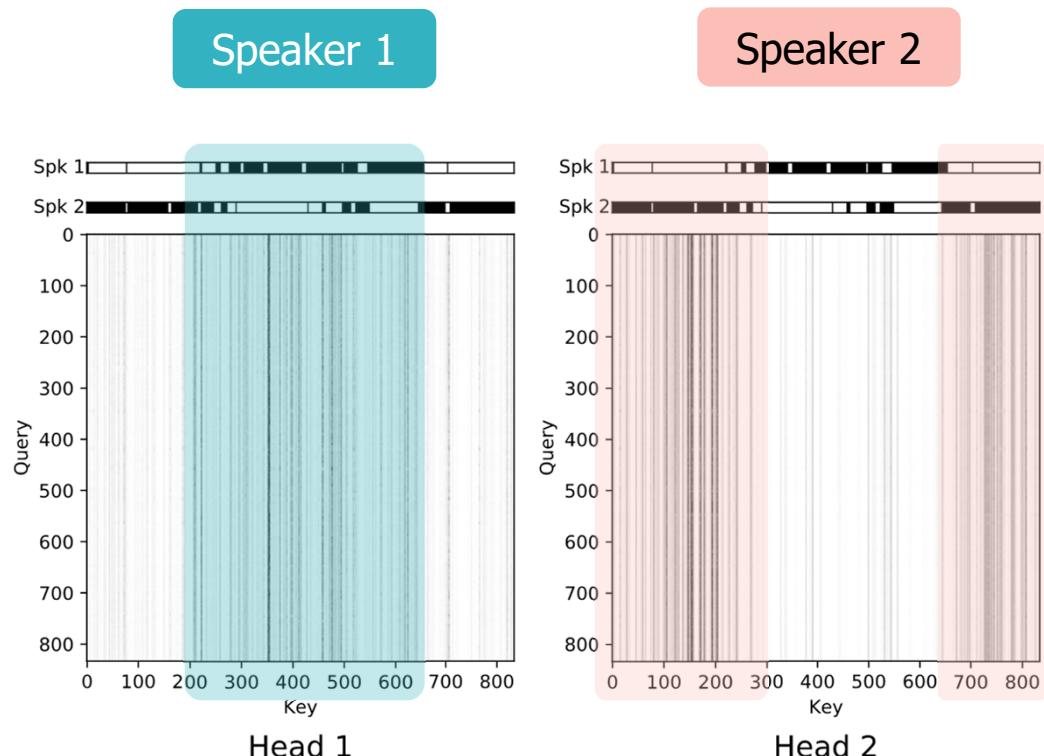
Permuted labels

Y. Fujita et al., "End-to-End Neural Speaker Diarization with Permutation-Free Objectives," Interspeech 2019.

# End-to-End Neural Diarization (EEND)

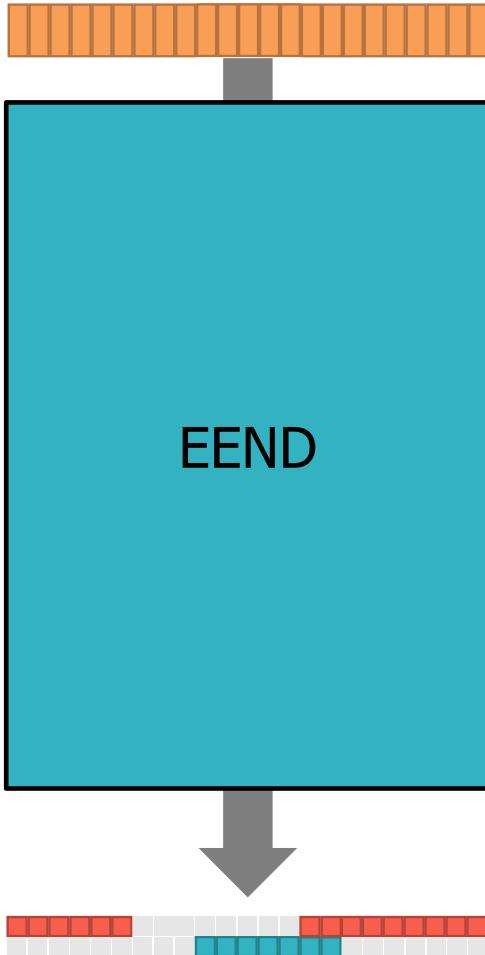


Attention weight matrix shows that one attention head focused on one speaker's activity.



Y. Fujita et al., "End-to-End Neural Speaker Diarization with Self-attention," ASRU 2019.

# End-to-End Neural Diarization (EEND)

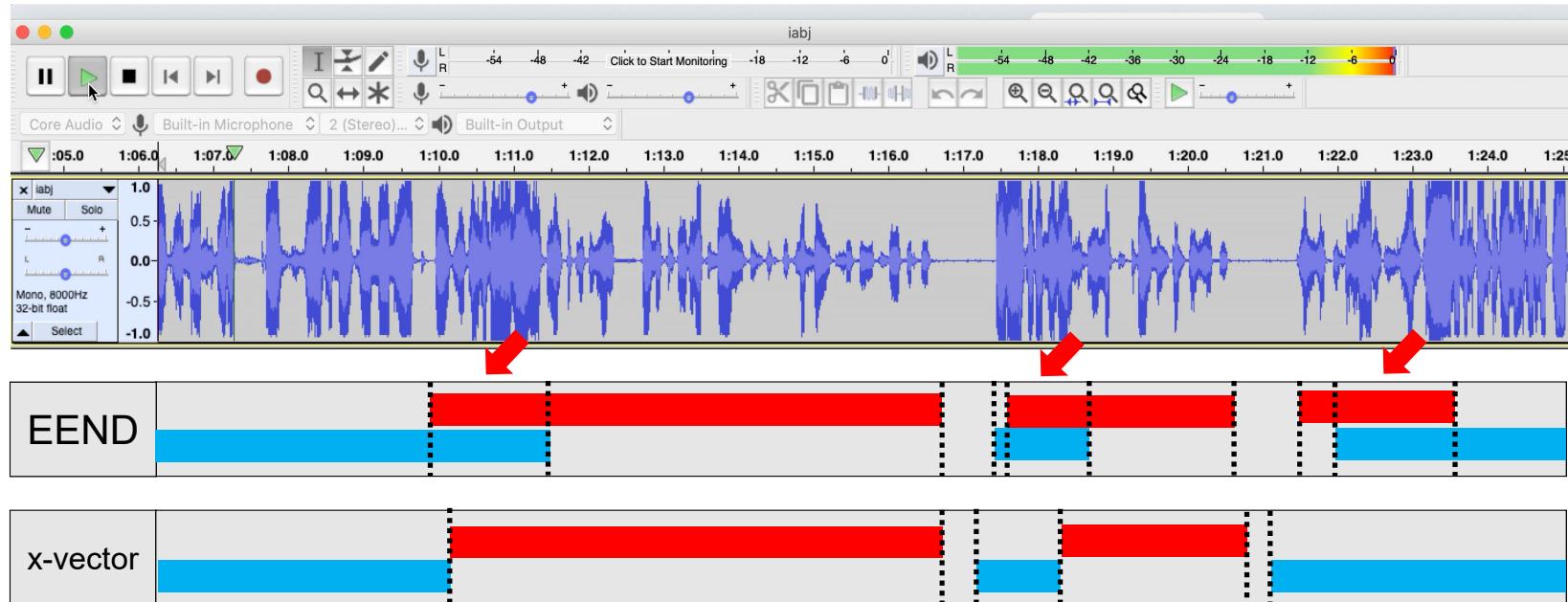
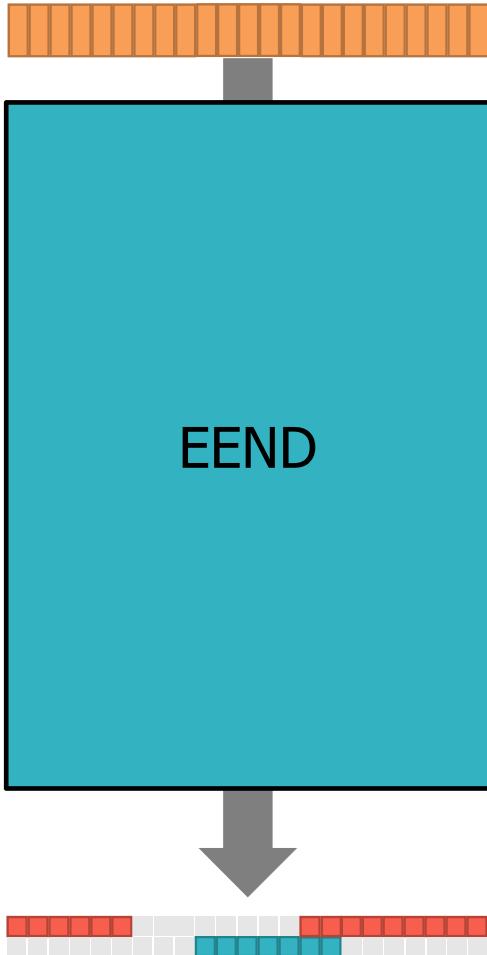


DERs on CALLHOME 2-speakers **with VAD and overlap errors.**

Method	Training Data	DER(%)
Clustering-based	i-vector	SWB2&SRE
	x-vector	SWB2&SRE
EEND	4-layer Self-attention	SimTrain (34.4%) → adaptation <b>9.54</b>

Y. Fujita et al., "End-to-End Neural Speaker Diarization with Self-attention," ASRU 2019.

# End-to-End Neural Diarization (EEND)

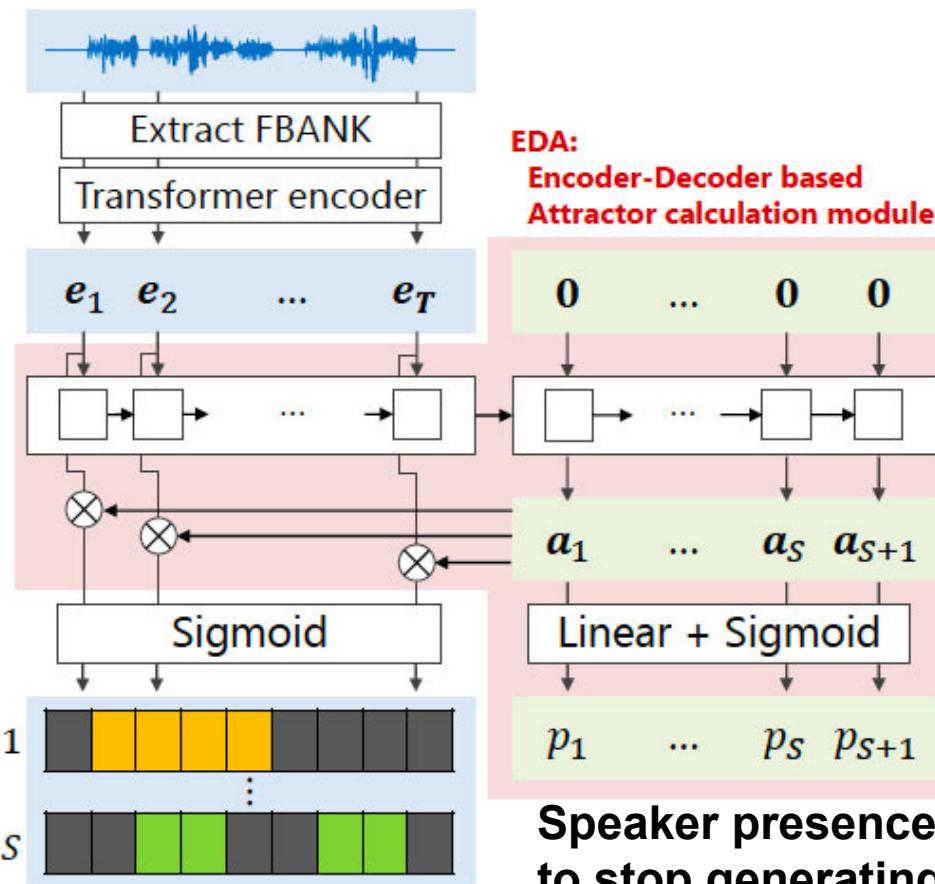
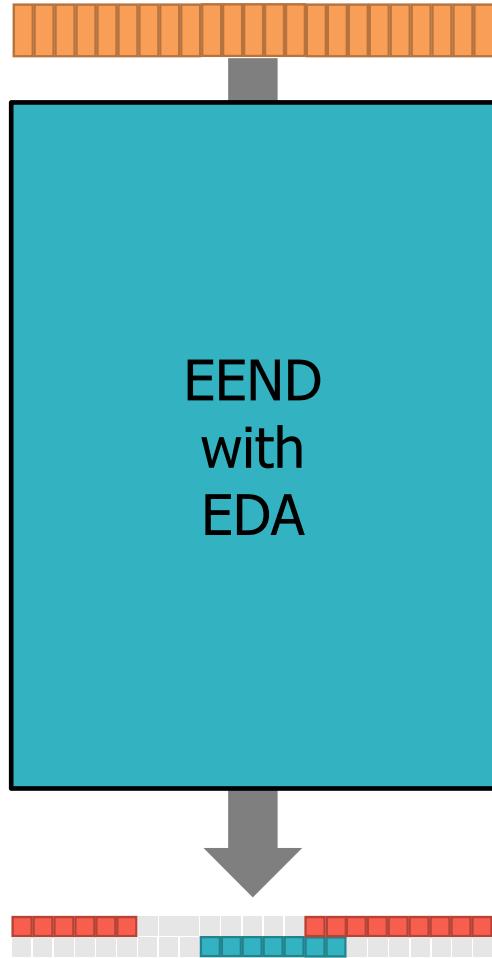


- 😊 End-to-end training
- 😊 Can handle overlapping speech
- 😢 Speaker counting

Top: waveform  
Middle: EEND  
Bottom: x-vector clustering

Y. Fujita et al., "End-to-End Neural Speaker Diarization with Self-attention," ASRU 2019.

# EEND with Encoder-Decoder Attractor (EDA)

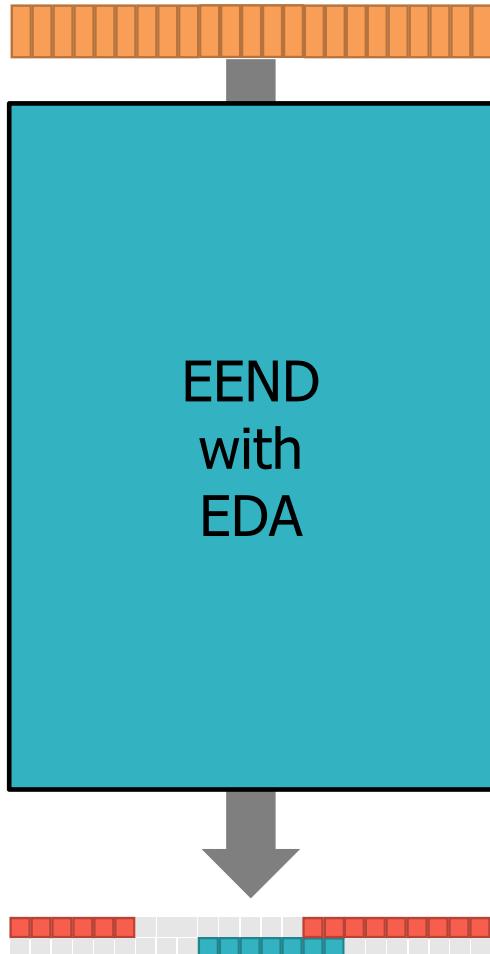


**Speaker-wise generation of attractor (centroid) vectors**

**Speaker presence prob.  
to stop generating new speakers**

S. Horiguchi et al., "End-to-End Speaker Diarization for an Unknown Number of Speakers with Encoder-Decoder Based Attractors," Interspeech 2020.

# EEND with Encoder-Decoder Attractor (EDA)



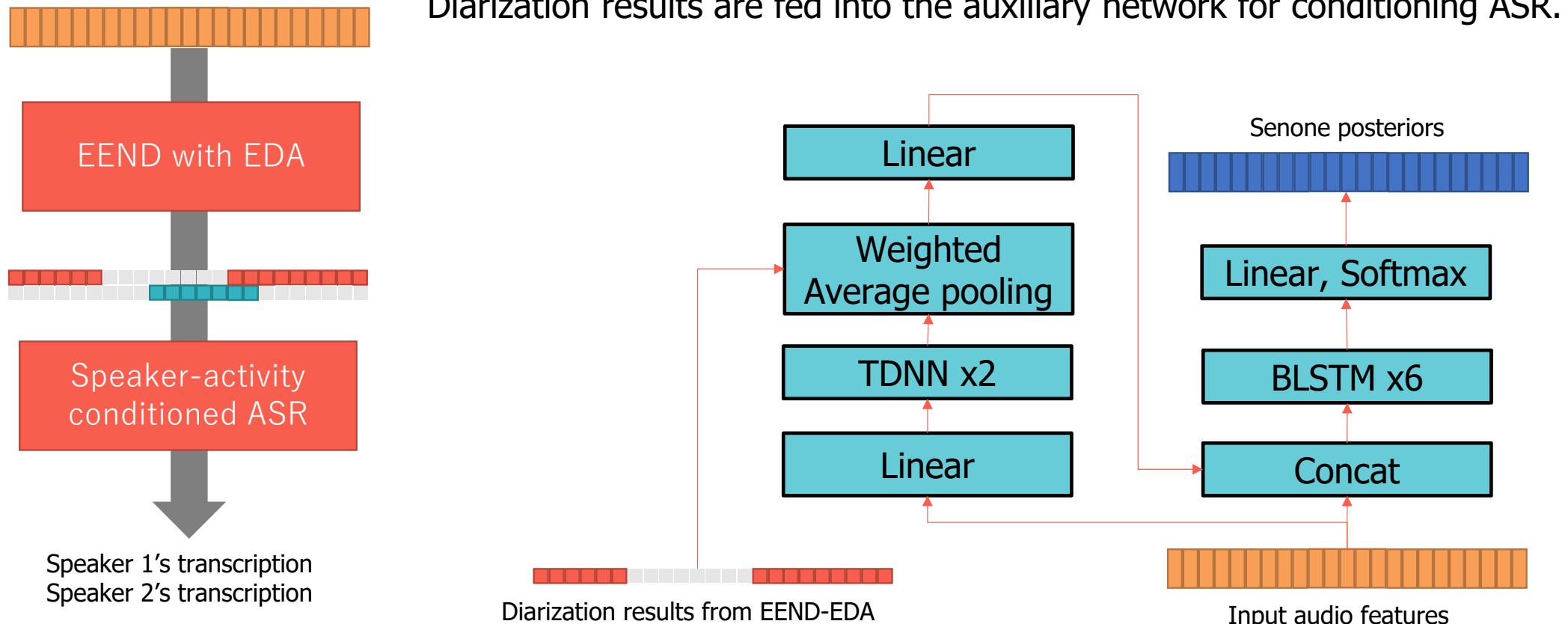
DERs on CALLHOME **2-6 speakers with VAD and overlap errors**

Model	Number of speakers					
	2	3	4	5	6	All
X-vector	15.45	18.01	22.68	<b>31.40</b>	<b>34.27</b>	19.43
<b>EEND-EDA</b>	<b>8.50</b>	<b>13.24</b>	<b>21.46</b>	33.16	40.29	<b>15.29</b>

- 😊 End-to-end training
- 😊 Can handle overlapping speech
- 😊 Speaker counting

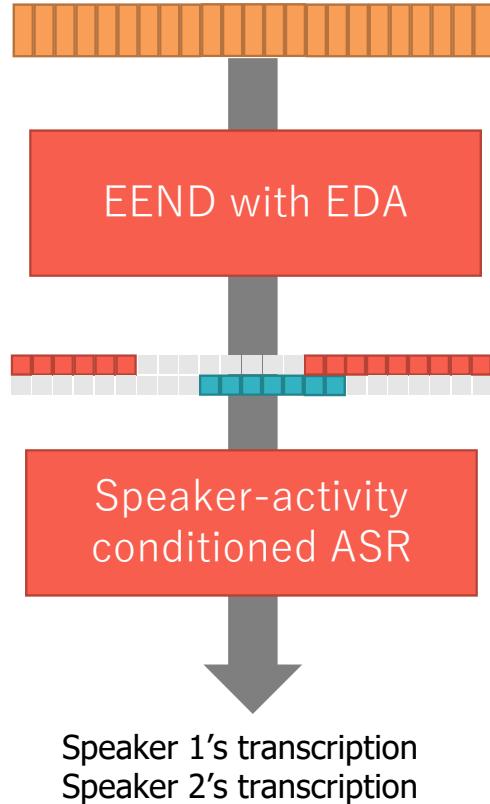
S. Horiguchi et al., "End-to-End Speaker Diarization for an Unknown Number of Speakers with Encoder-Decoder Based Attractors," Interspeech 2020

# Diarization-assisted ASR



S. R. Chetupalli et al., "Speaker diarization assisted ASR for multi-speaker conversations," arXiv 2021.

# Diarization-assisted ASR



SWERs on Hub 5 English evaluation speech

Model	w/ Ground-truth activity	w/ EEND-EDA
Single-speaker ASR	38.3	37.7
<b>Diarization-assisted ASR</b>	<b>25.8</b>	<b>26.5</b>

S. R. Chetupalli et al., "Speaker diarization assisted ASR for multi-speaker conversations," arXiv 2021.

# DIHARD III challenge

The third speaker diarization challenge focusing on **“hard” diarization**.  
Evaluation sets are drawn from a diverse conditions including  
broadcast interviews, meeting speech, speech in restaurants, and YouTube videos.

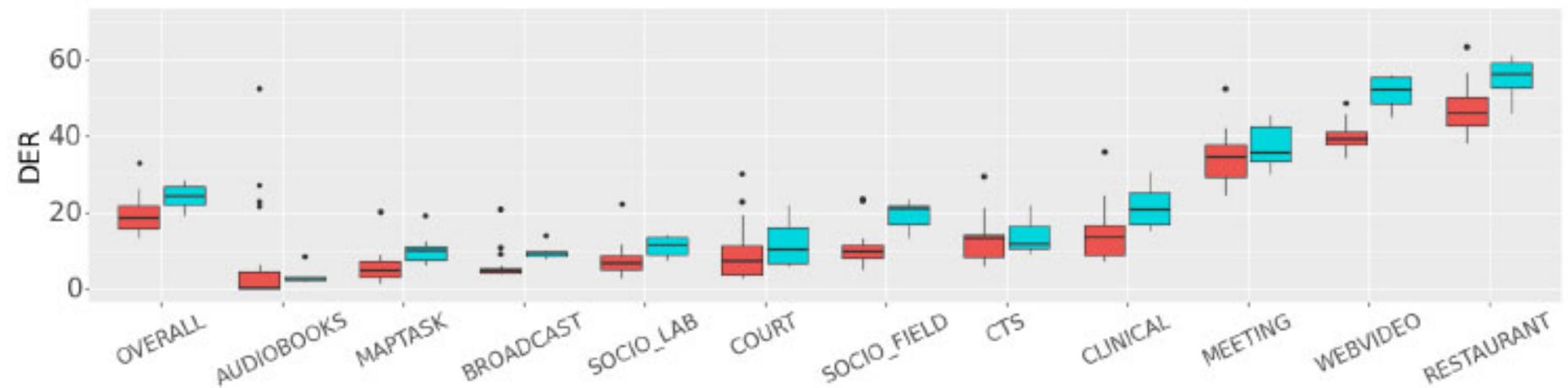
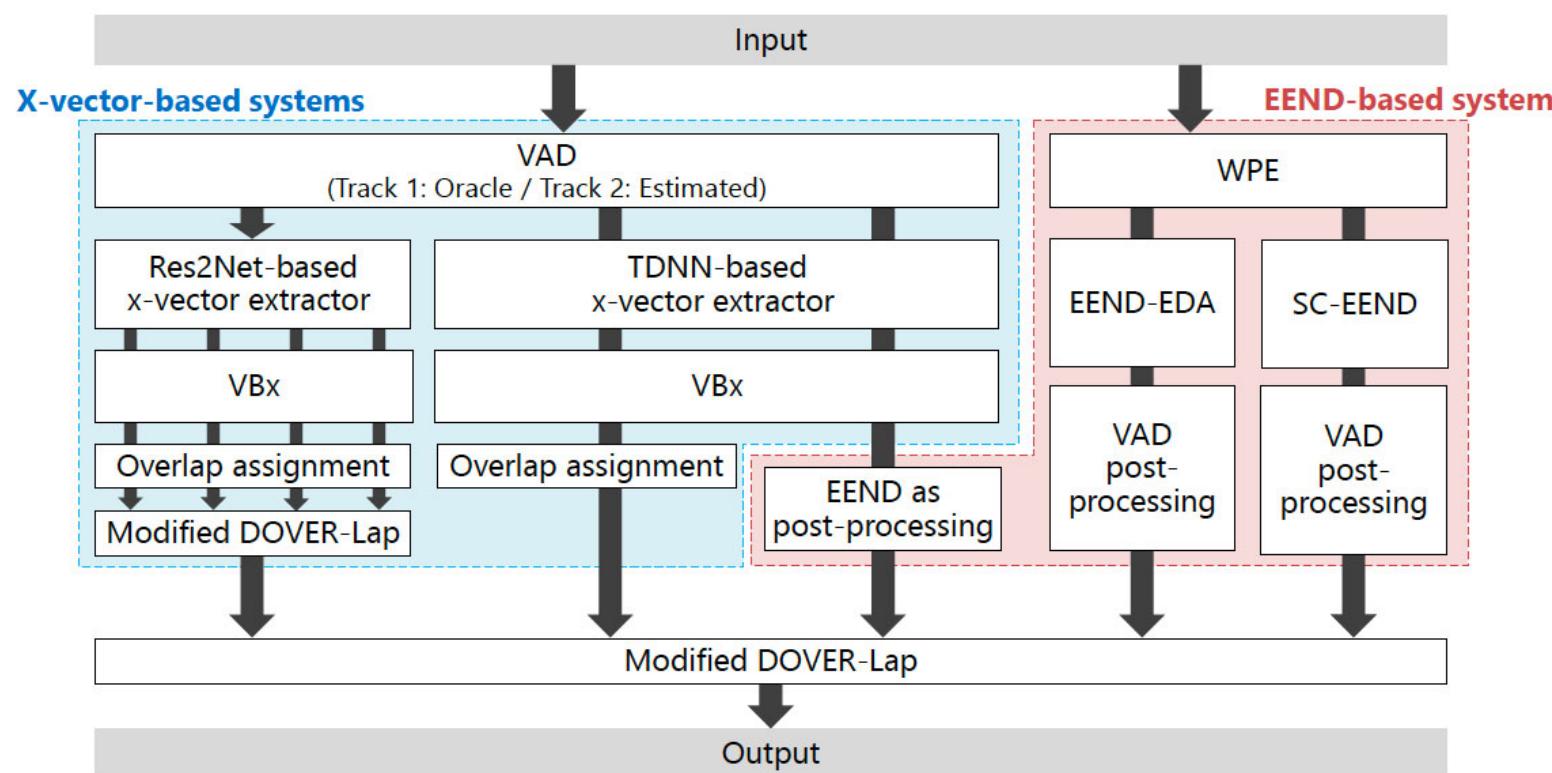


Figure 1: *Track 1 (red) and track 2 (blue) DER of primary submissions by domain on the core EVAL set.*

N. Ryant et al., “The Third DIHARD Diarization Challenge,” arXiv 2021.

# DIHARD III challenge

Hitachi/JHU system combines **clustering-based** and **end-to-end neural** diarization subsystems.



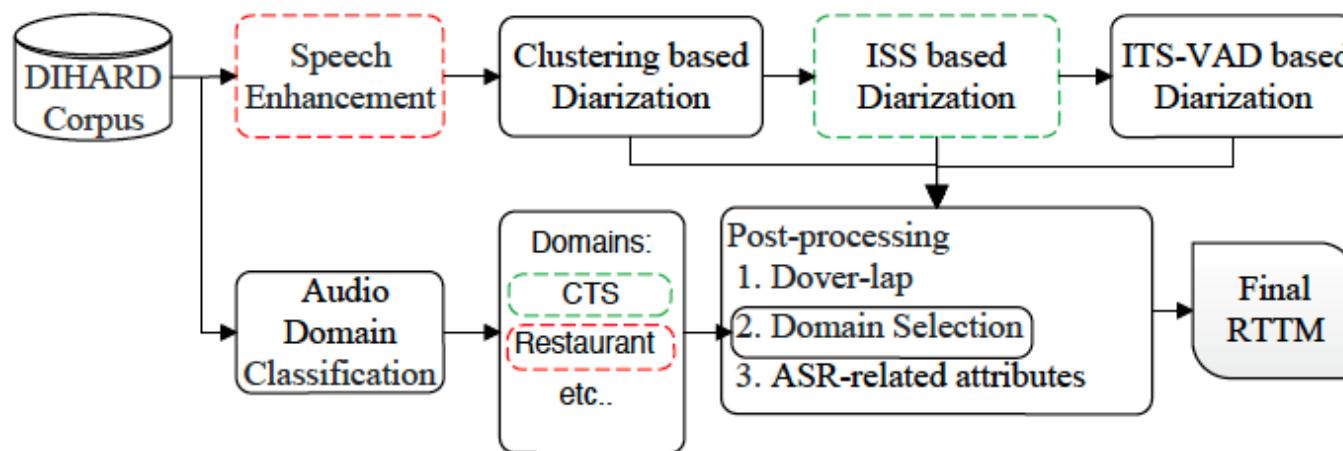
DERs on core set (Task 2)  
with **VAD and overlap errors**

Model	Dev	Eval
DIHARD III baseline	22.28	27.34
Res2Net x-vector clustering	18.39	24.64
EEND-EDA	18.50	22.84
...		
<b>DOVER-Lap of 5 systems</b>	<b>15.81</b>	<b>20.01</b>

S. Horiguchi et al., "The Hitachi-JHU DIHARD III System: Competitive End-to-End Neural Diarization and X-Vector Clustering Systems Combined by DOVER-Lap," DIHARD III workshop, 2021

# DIHARD III challenge

USTC-NELSLIP system trains **TS-VAD** and **source separation** models initialized by **clustering-based** diarization.



DERs on core set (Task 2)  
with **VAD and overlap errors**

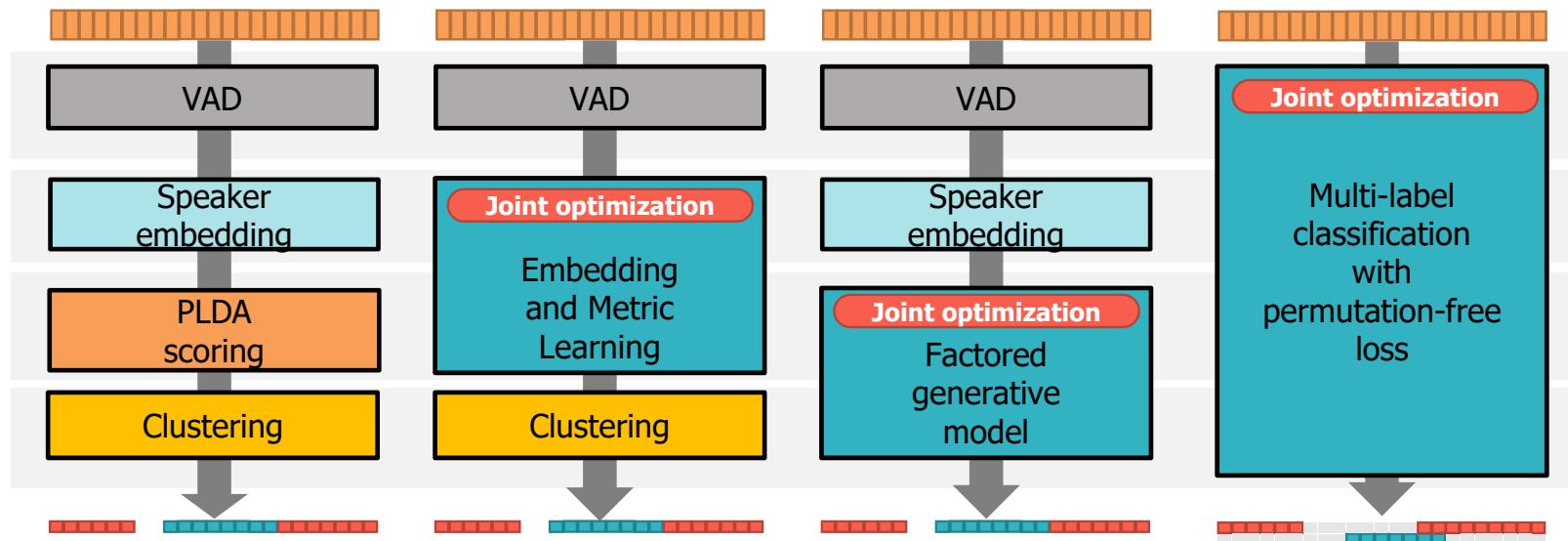
Model	Dev	Eval
DIHARD III baseline	22.28	27.34
<b>USTC-NELSLIP</b>	<b>14.49</b>	<b>19.37</b>

Domain Selection: Performance comparison among three methods.

Method	AUDIOBOOK	CTS	Restaurant	WebVideo	Others
Clustering	-	:(sad)	:(smile)	:(smile)	:(sad)
Source separation	-	:(smile)	:(sad)	:(sad)	:(sad)
TS-VAD	-	:(smile)	:(sad)	:(sad)	:(smile)

Y. Wang et al., "USTC-NELSLIP System Description for DIHARD III Challenge," DIHARD III workshop, 2021

# Short summary of “Diarization + X”

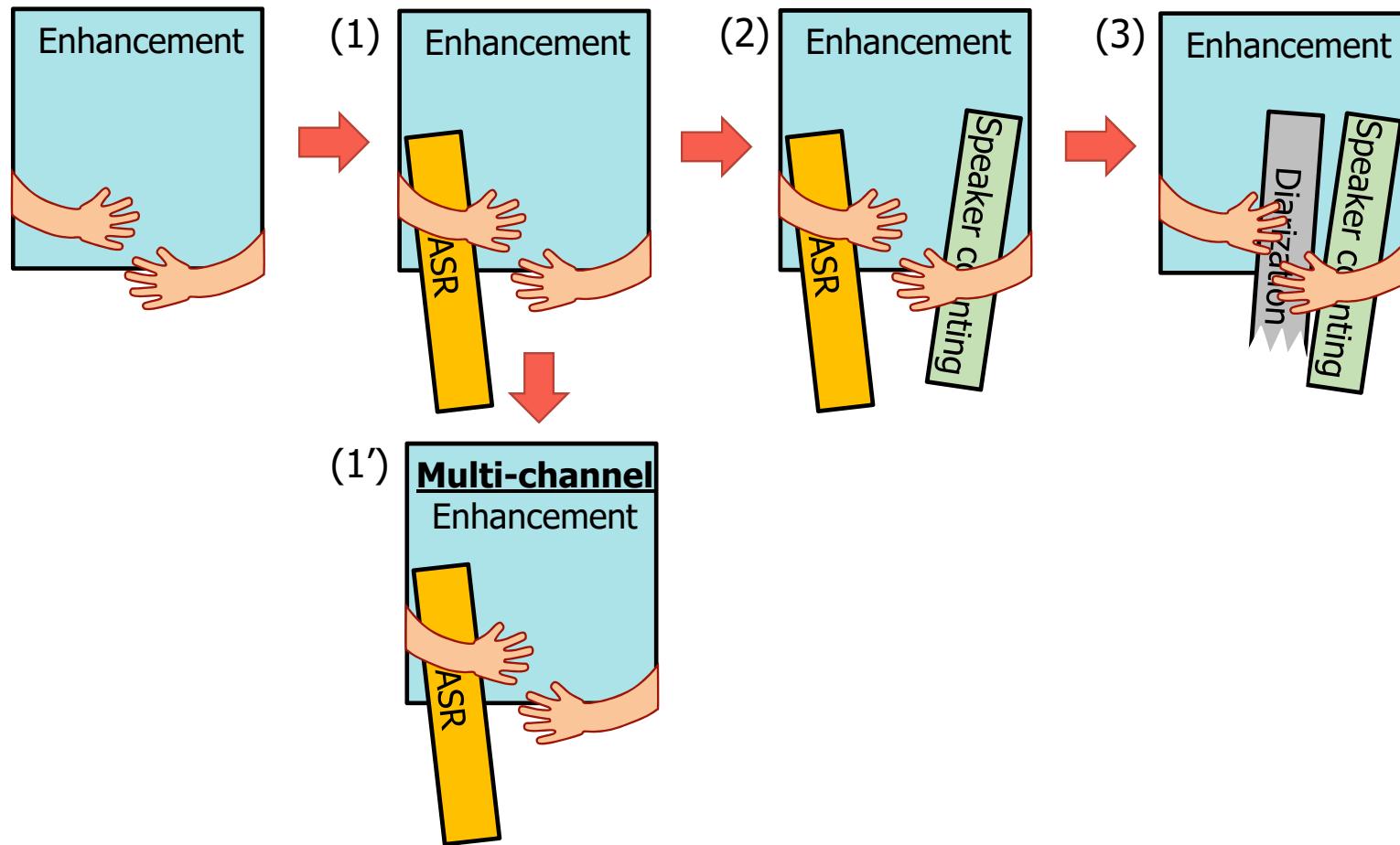


- Reviewed recent techniques focusing on joint optimization in diarization systems.
- End-to-end training was made possible by replacing clustering module with a neural network.
- End-to-end systems exhibit strong performance in various domains of DIHARD III challenge.
- Problems on large number of speakers with frequent overlaps remain far from solved.
- Diarization→ASR: further joint optimal systems are possible.

### 3. A new research trend: Jointly optimal systems

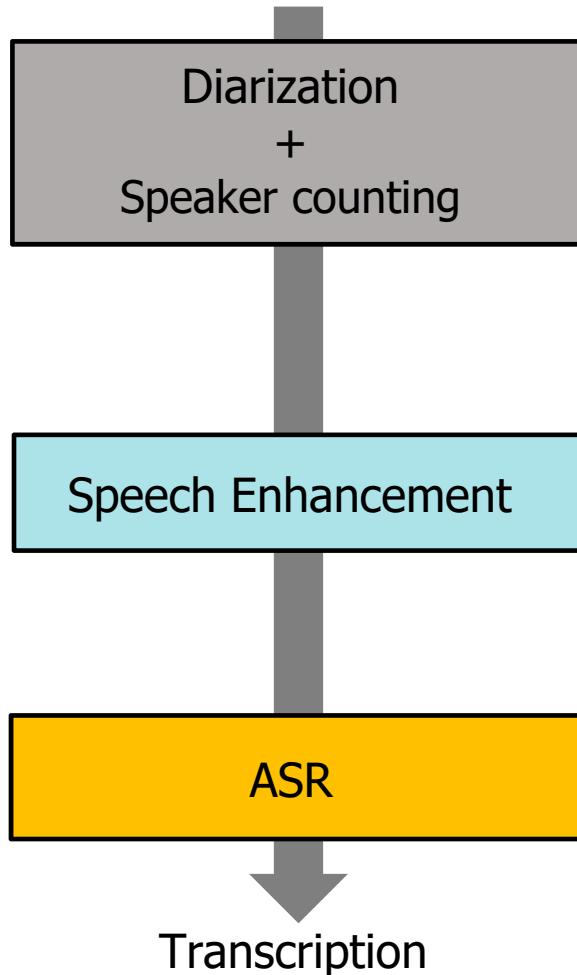
- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$
- 3.3. ASR +  $x$

# What has been achieved in “Enhancement + $x$ ”

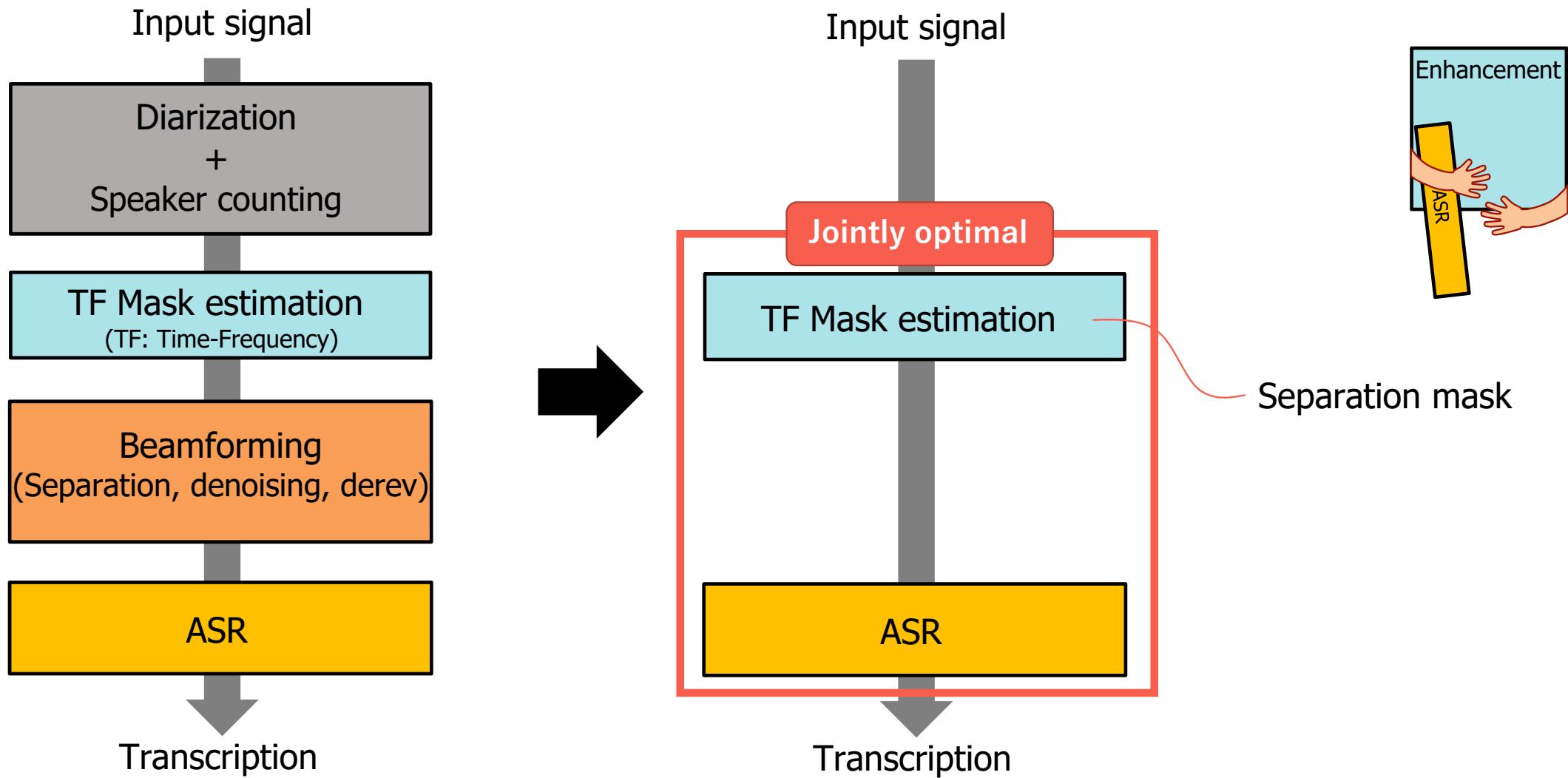


# (1) TF Mask estimation + ASR

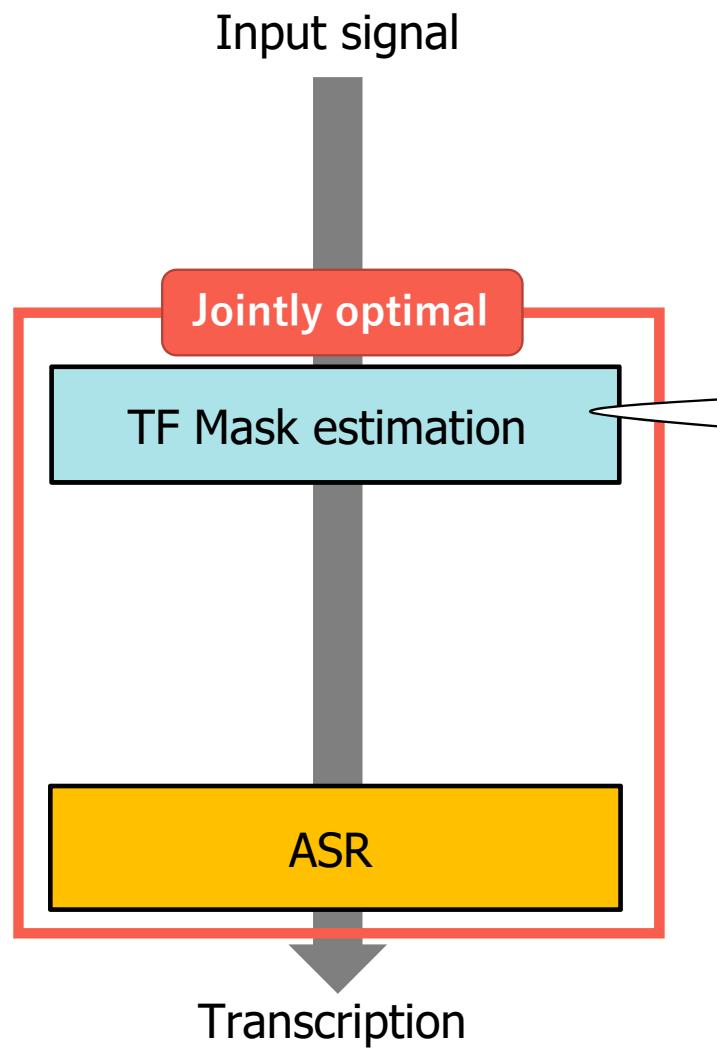
Input signal



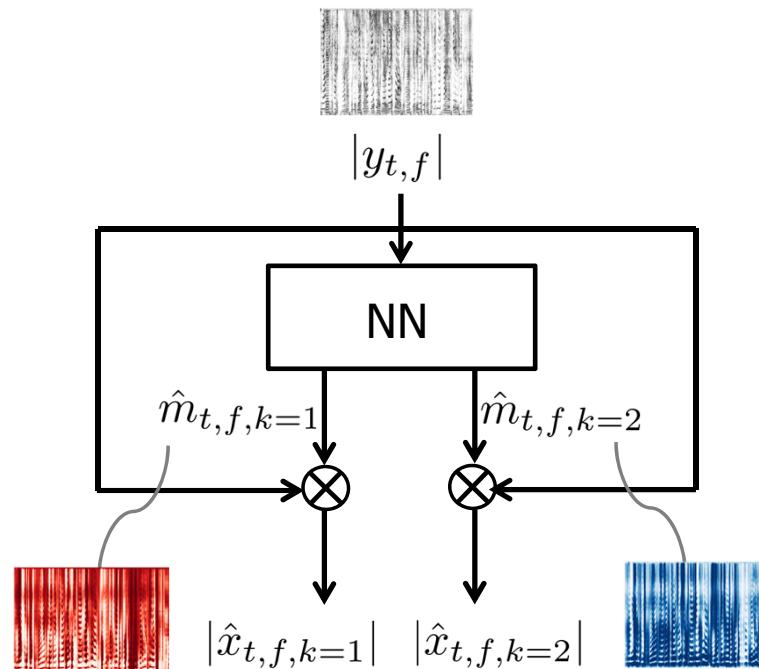
# (1) TF Mask estimation + ASR



# (1) TF Mask estimation + ASR: NN-based mask estimator



- Estimating TF masks (i.e., STFT-domain approach) [Yu+, 2017], or separated waveforms (i.e., time-domain approach) [Luo+, 2018]  
→ NN-based source separation



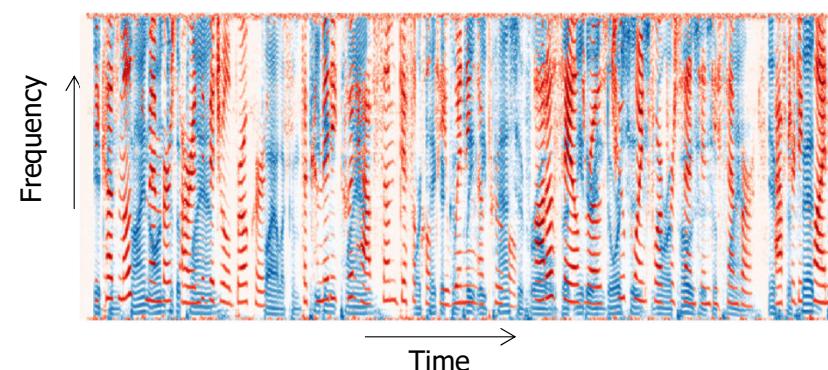
D. Yu et al., Permutation invariant training of deep models for speaker-independent multi-talker speech separation, ICASSP, 2017  
Y. Luo et al., TaSNet: Time-Domain Audio Separation Network for Real-Time, Single-Channel Speech Separation," ICASSP, 2018

# (1) TF Mask estimation + ASR: NN-based mask estimator

Input signal

- Estimating TF masks (i.e., STFT-domain approach) [Yu+, 2017], or separated waveforms (i.e., time-domain approach) [Luo+, 2018]  
→ NN-based source separation

Speech signals are **sparse**, and rarely overlap each other.

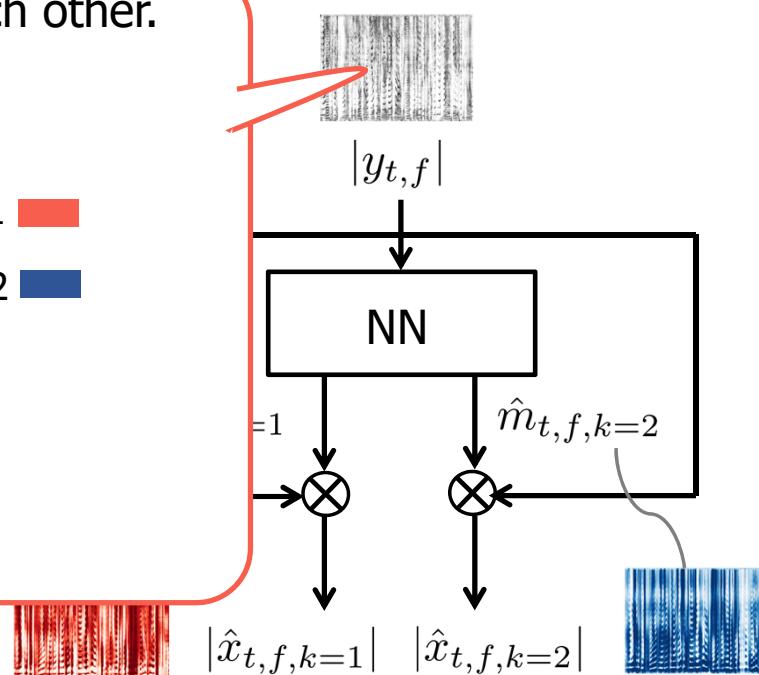


Speaker 1

Speaker 2

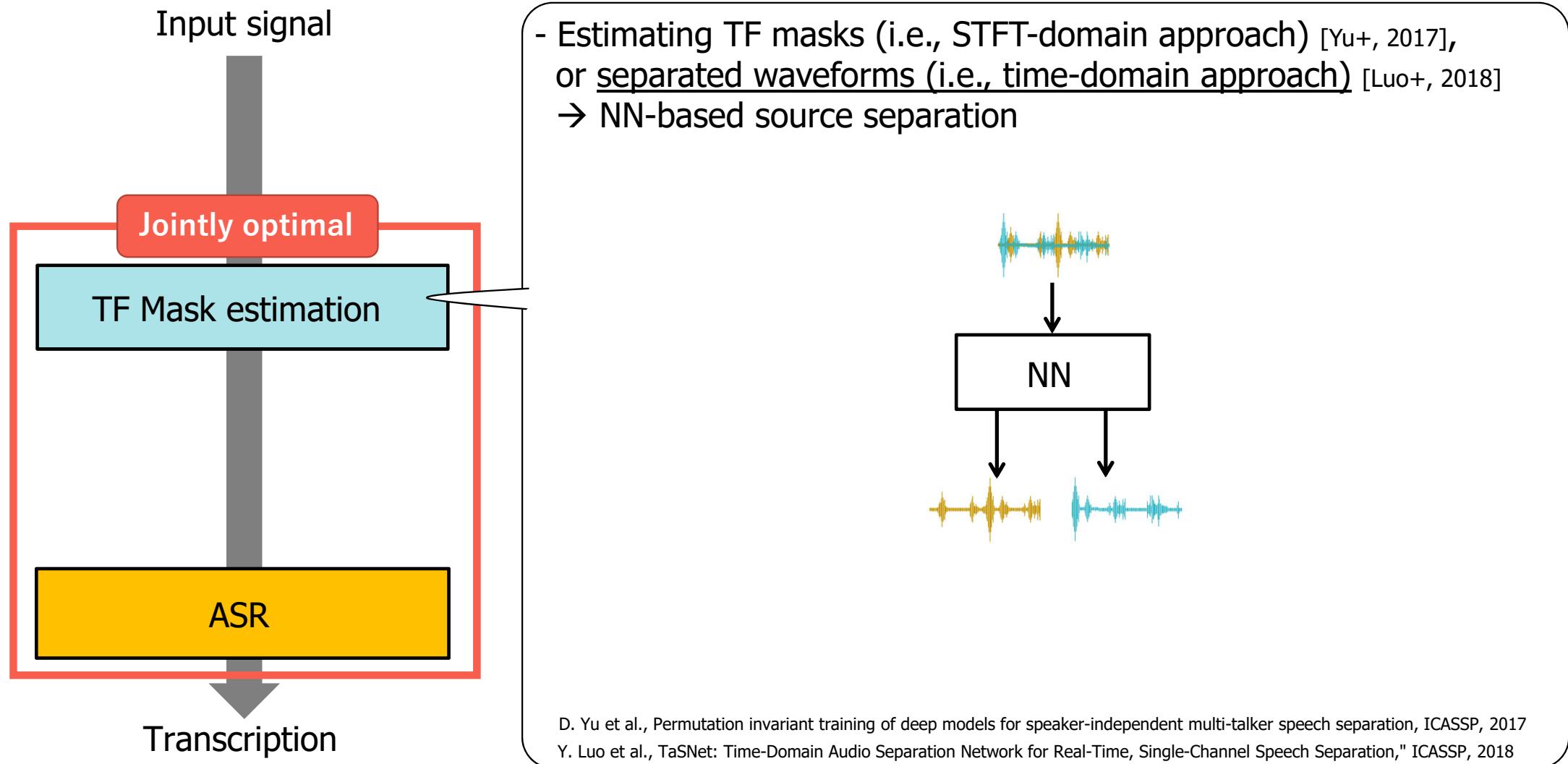
ASR

Transcription

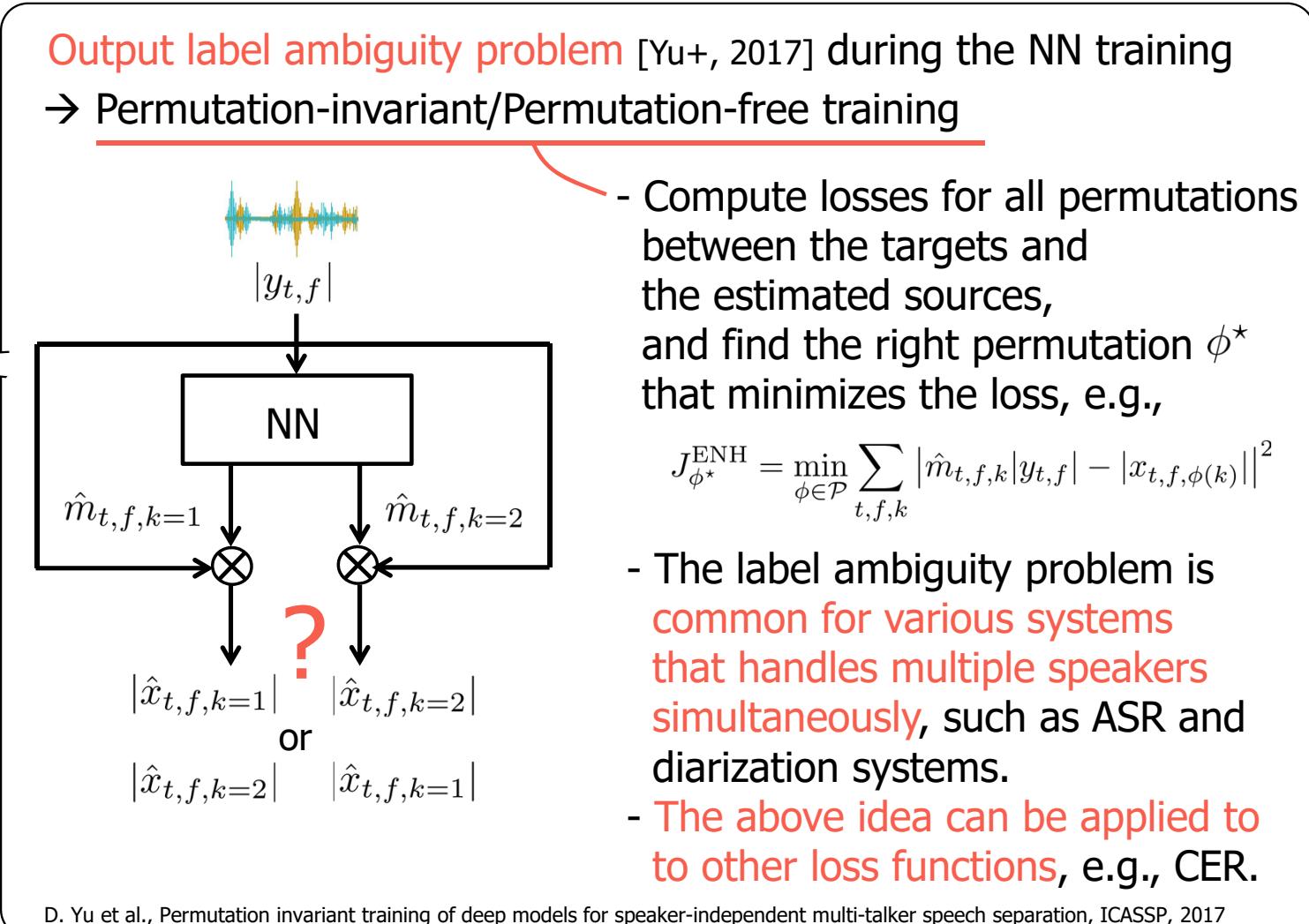
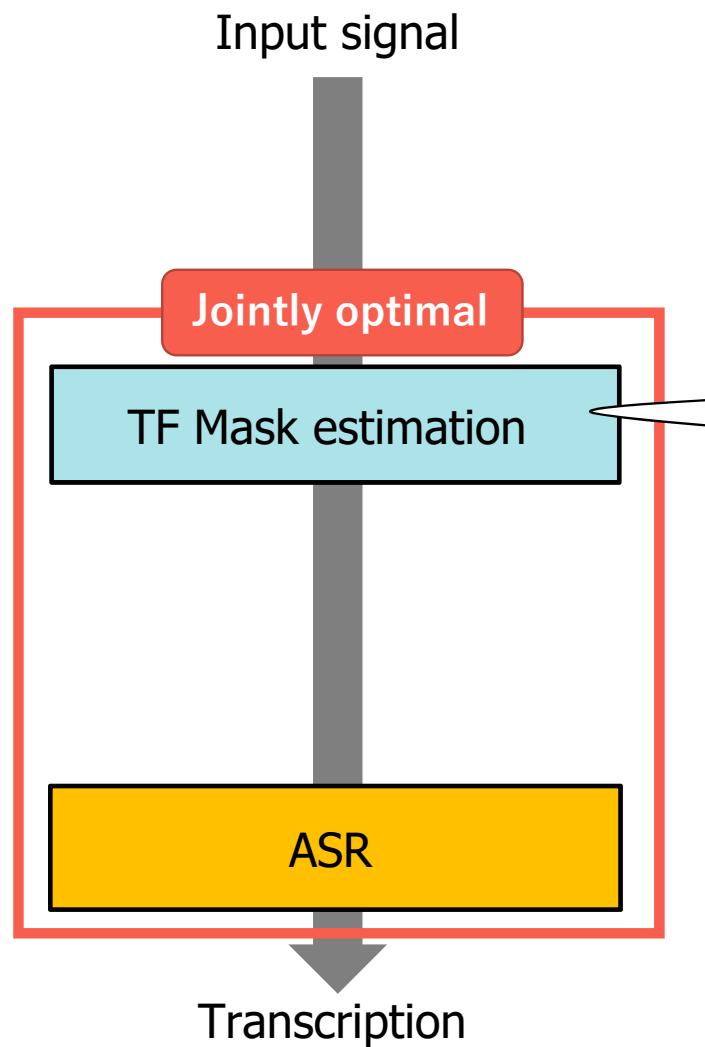


D. Yu et al., Permutation invariant training of deep models for speaker-independent multi-talker speech separation, ICASSP, 2017  
Y. Luo et al., TaSNet: Time-Domain Audio Separation Network for Real-Time, Single-Channel Speech Separation," ICASSP, 2018

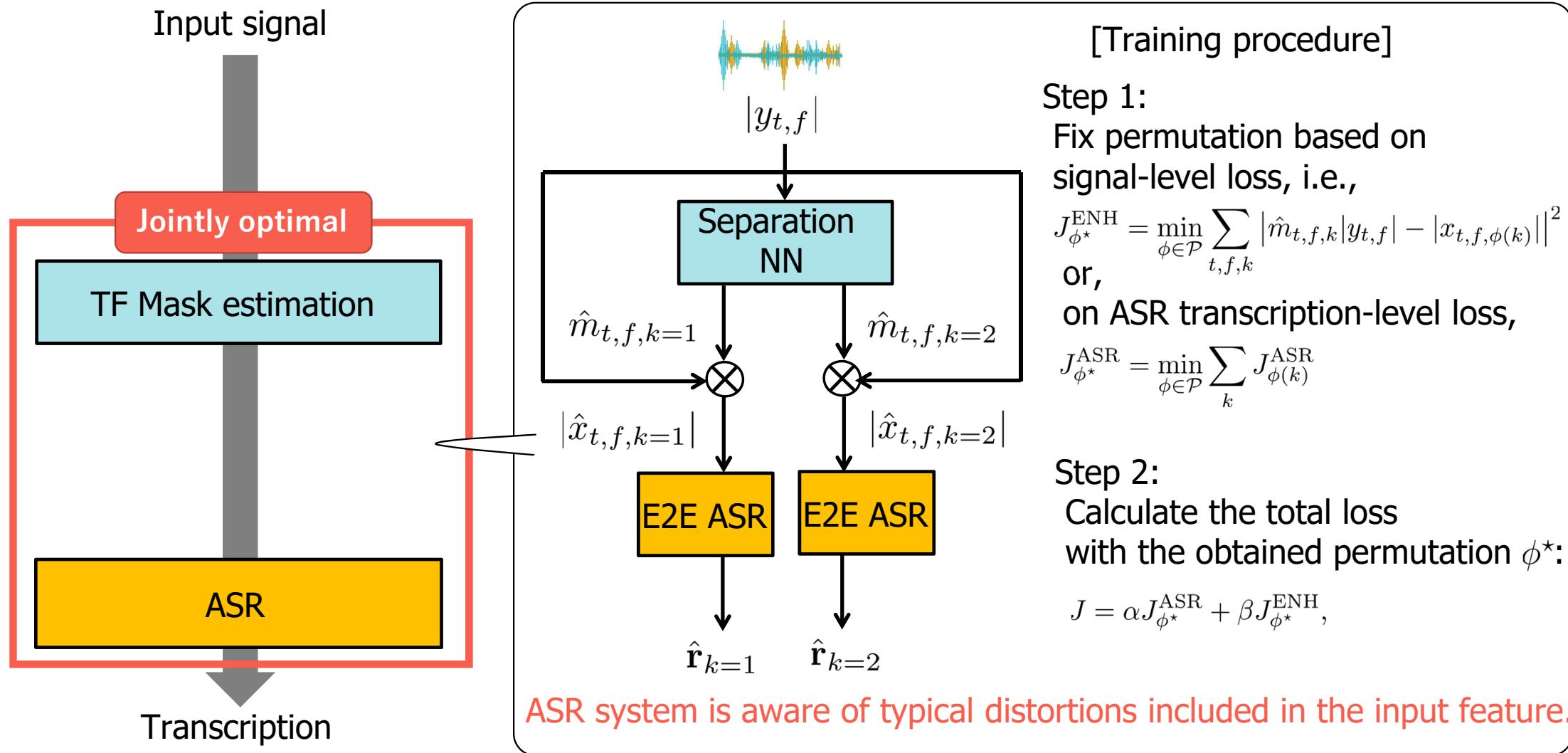
# (1) TF Mask estimation + ASR: NN-based mask estimator



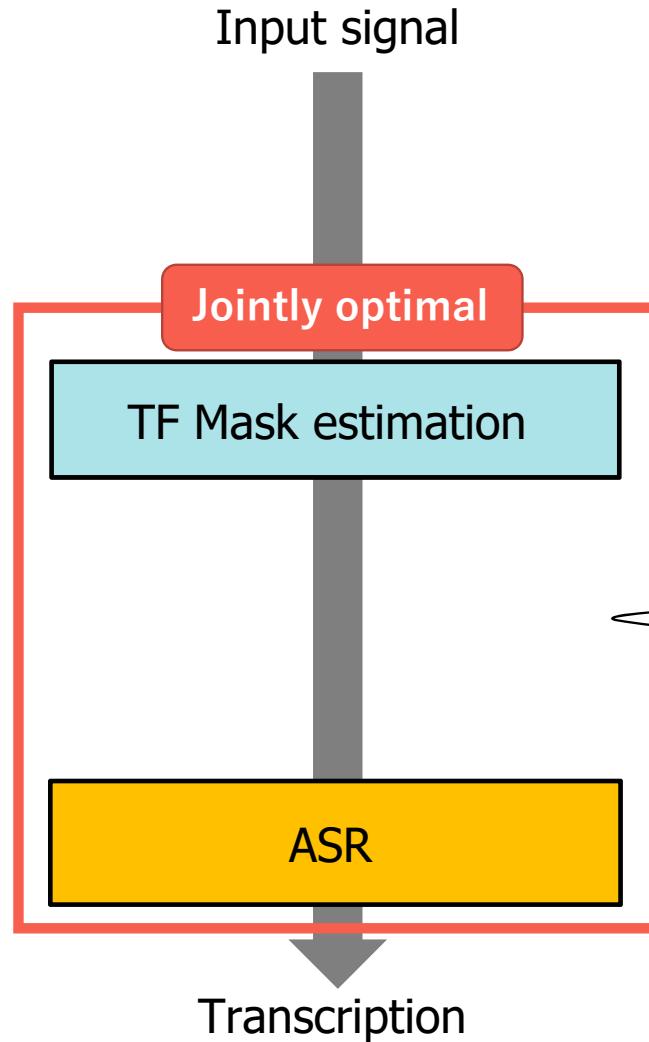
# (1) TF Mask estimation + ASR: Label ambiguity problem



# (1) TF Mask estimation + ASR: Label ambiguity problem



# (1) TF Mask estimation + ASR: Effectiveness



- Training and test data: WSJ-2mix (2 speaker full-overlap, noiseless, anechoic)

	Optimized part in joint system		SWER(%)
	front-end	back-end	
No proc.	-	-	79.1
STFT-domain separation	-	-	23.1
STFT-domain separation	-	✓	17.9
STFT-domain separation	✓	✓	13.2
Time-domain separation	-	-	22.9
Time-domain separation	-	✓	11.7
Time-domain separation	✓	✓	11.0

STFT-domain separation: Deep clustering+Mask inference [Settle+, 2018]

Time-domain separation: Conv-TasNet [von Neumann+, 2020]

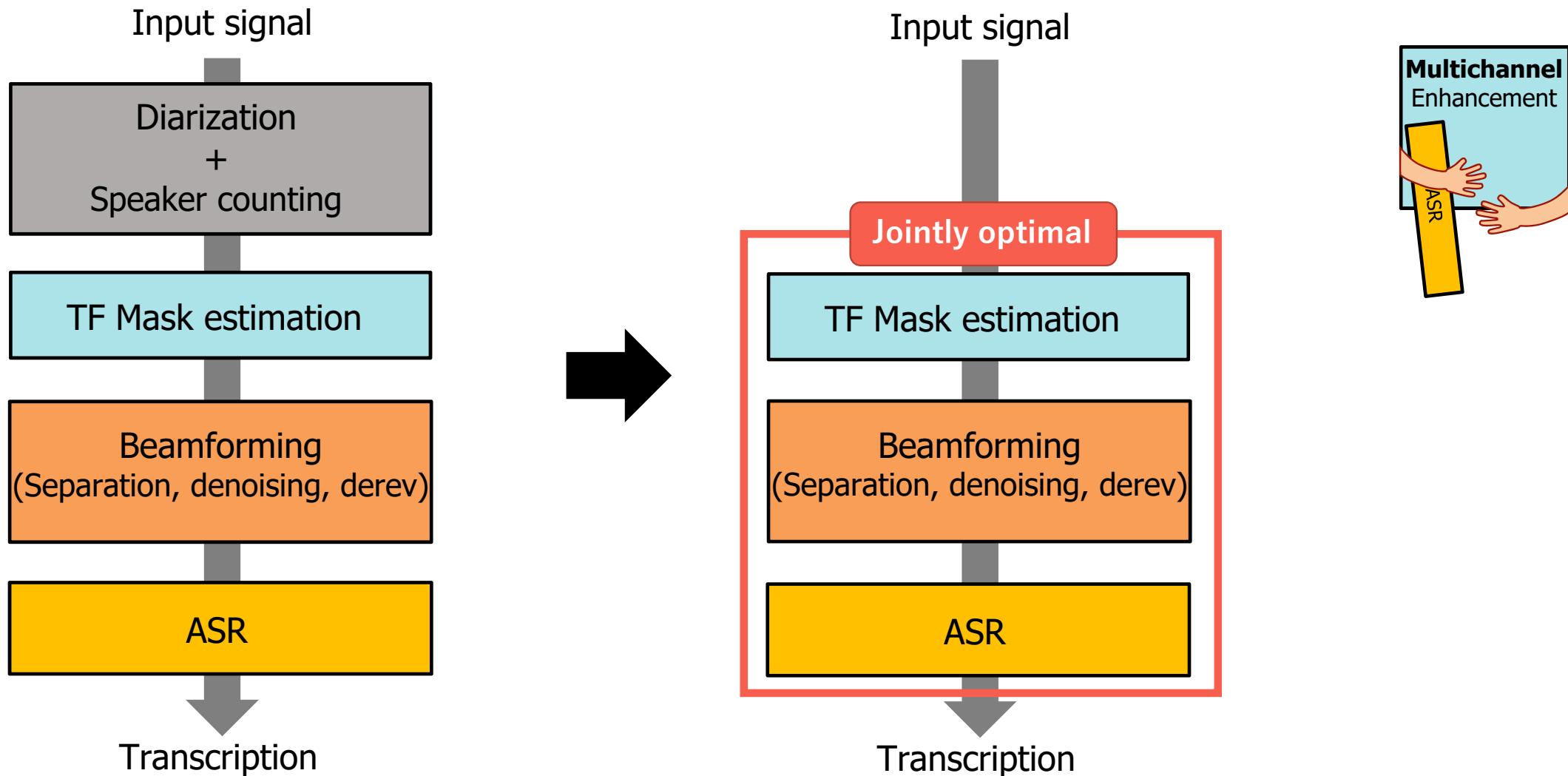
- Best results are obtained by jointly optimizing both front-end and back-end

✓ **Jointly optimal: 1ch separation + ASR**

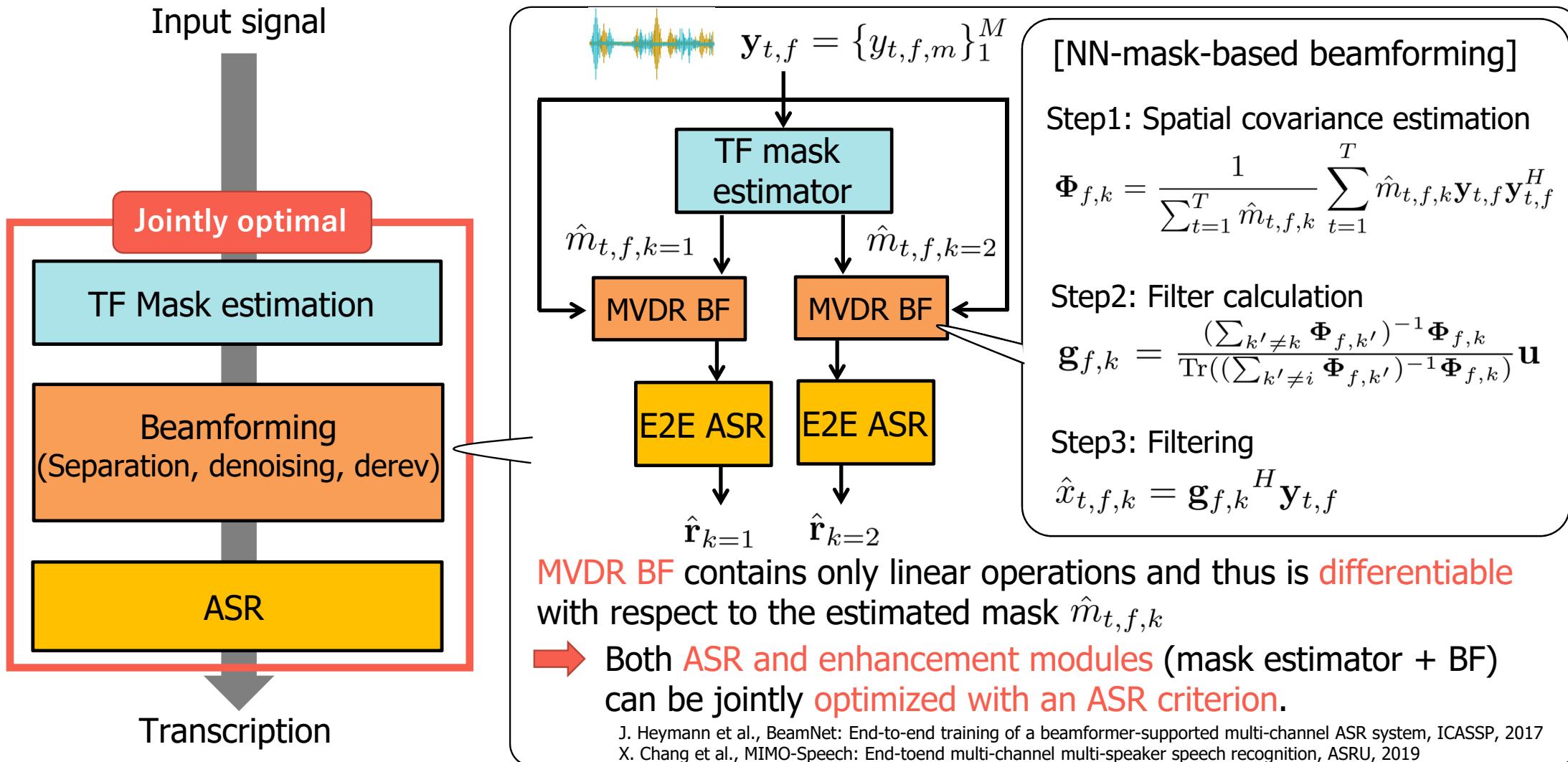
S. Settle et al., End-to-End Multi-Speaker Speech Recognition, ICASSP, 2018

T. von Neumann et al., End-to-end training of time domain audio separation and recognition, ICASSP, 2020

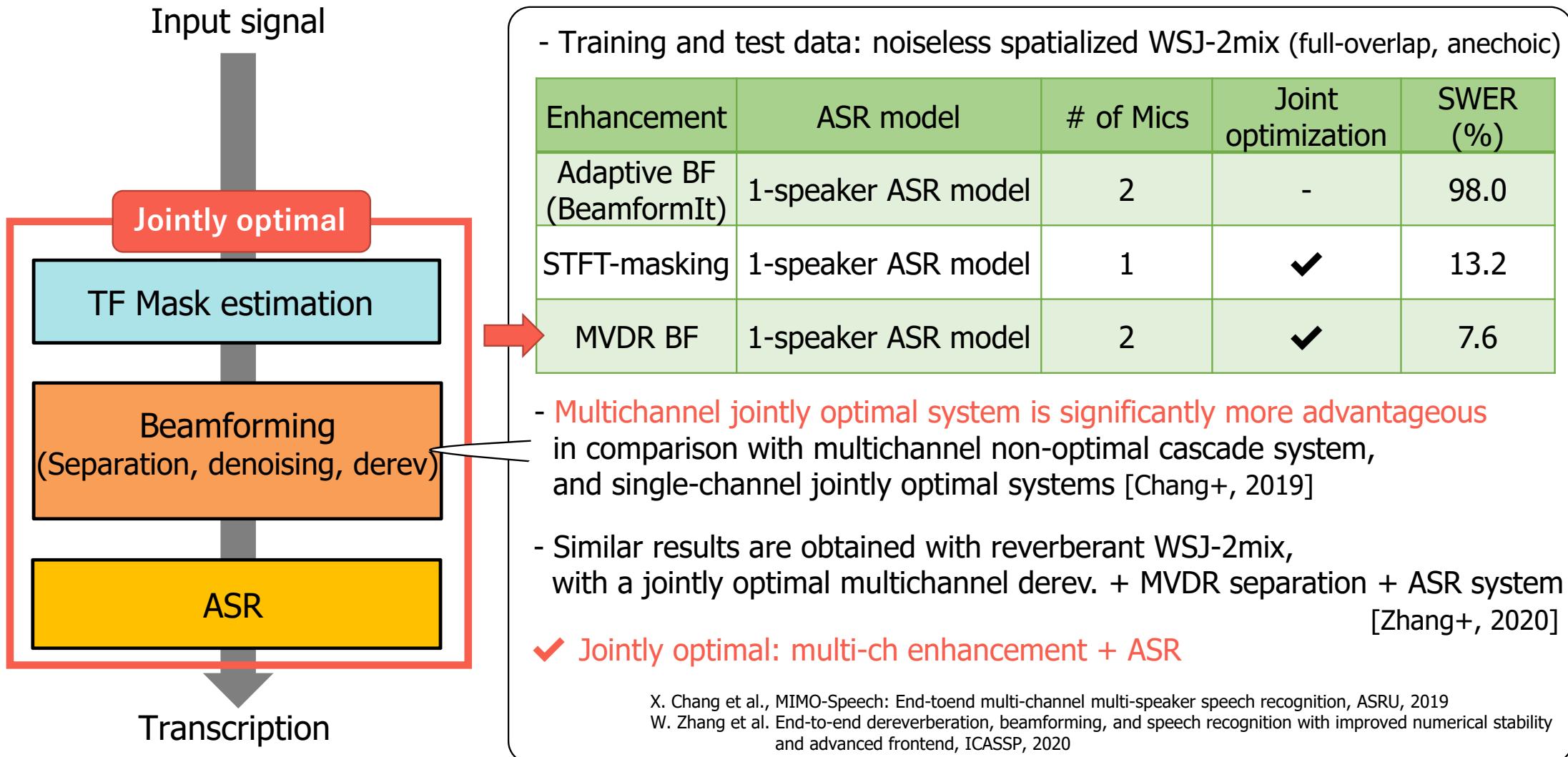
# (1') TF Mask estimation + Beamforming + ASR



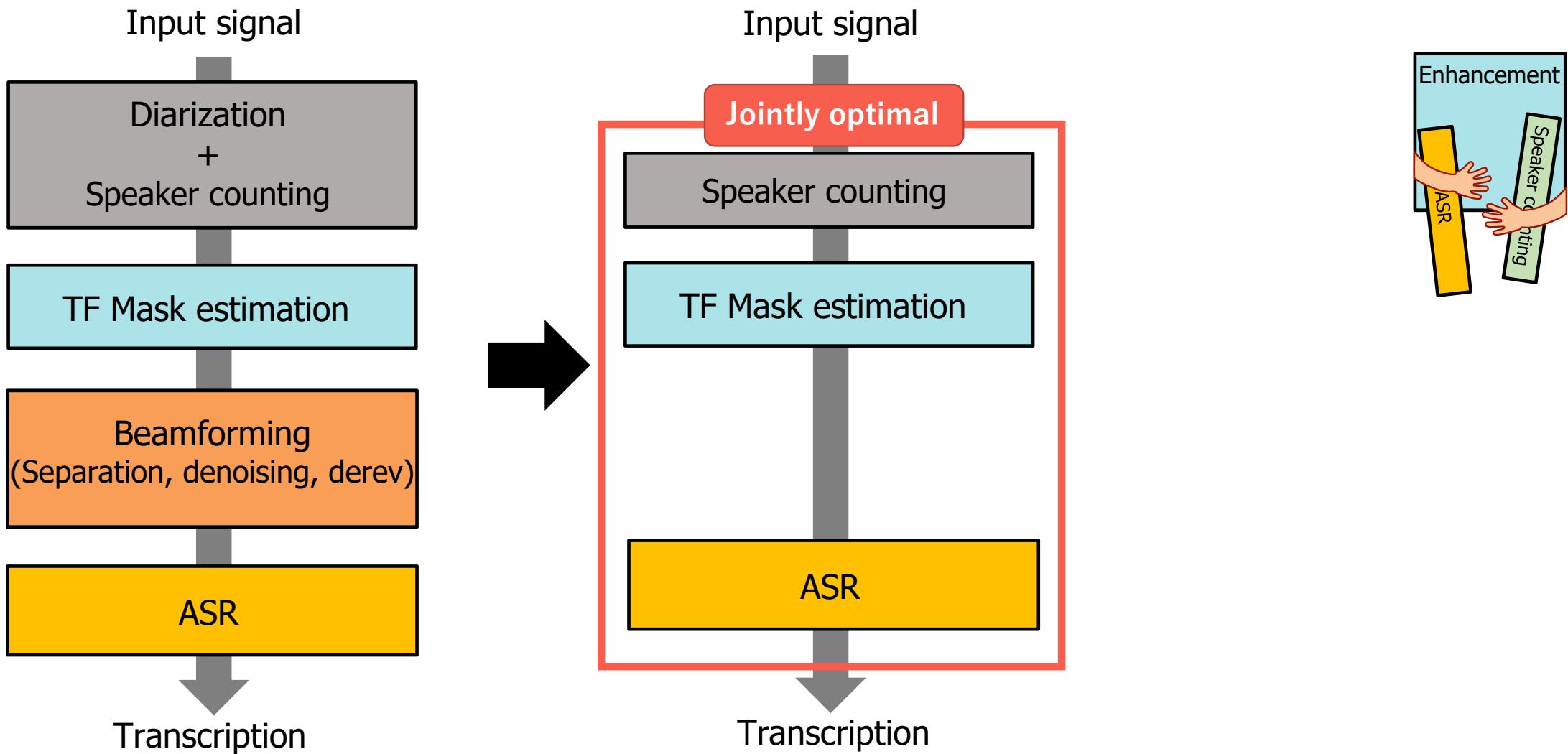
# (1') TF Mask estimation + Beamforming + ASR



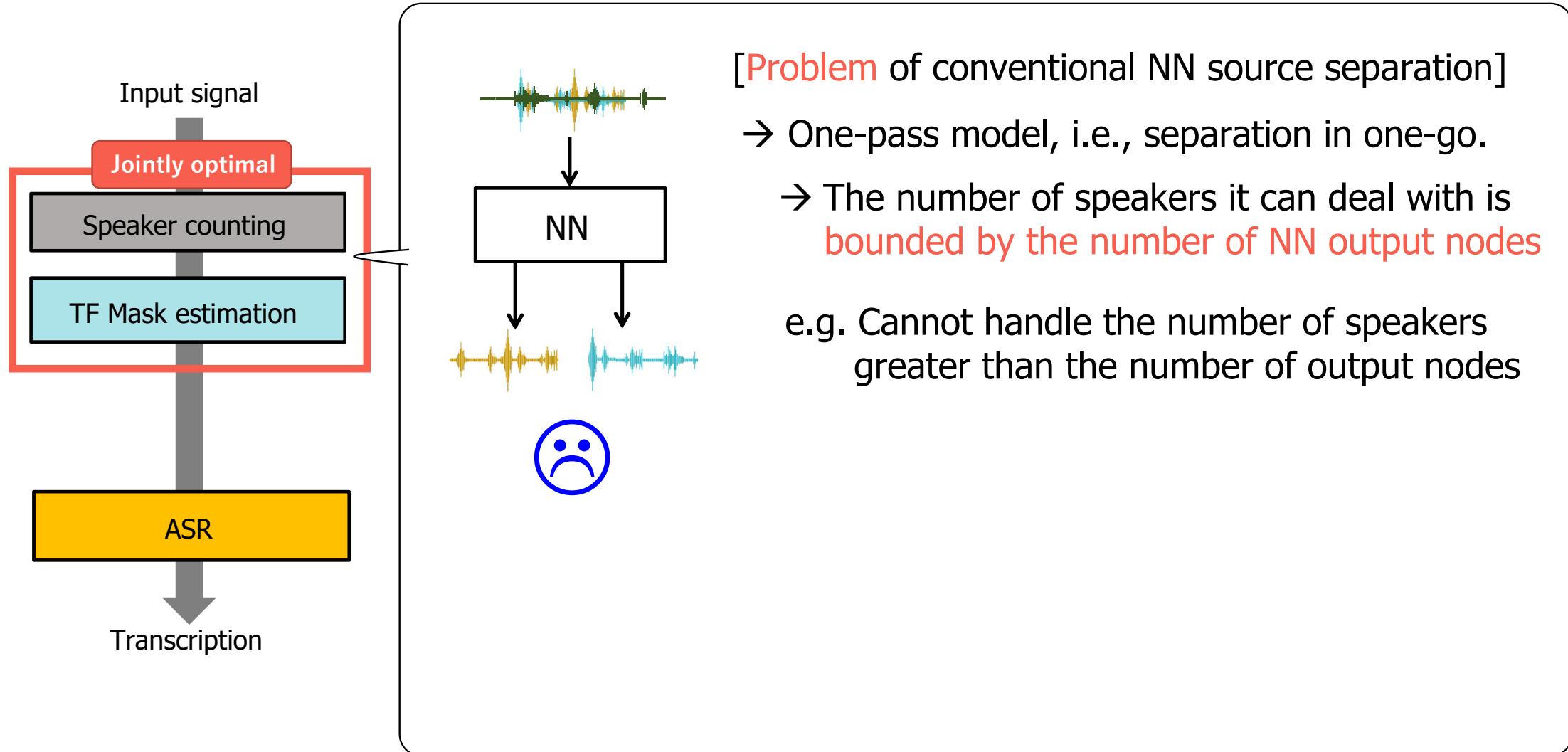
# (1') TF Mask estimation + Beamforming + ASR



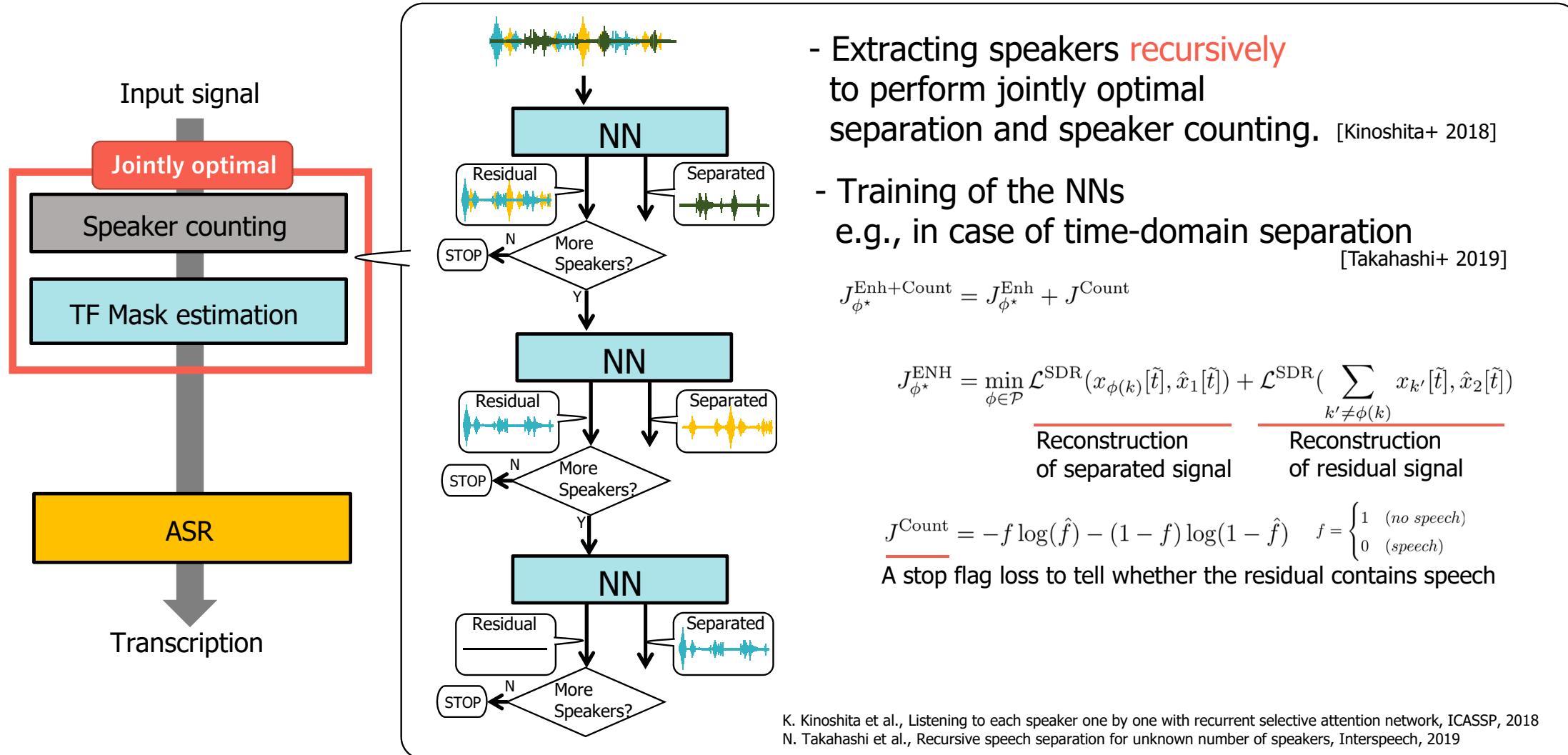
## (2) Speaker counting + TF Mask estimation + ASR



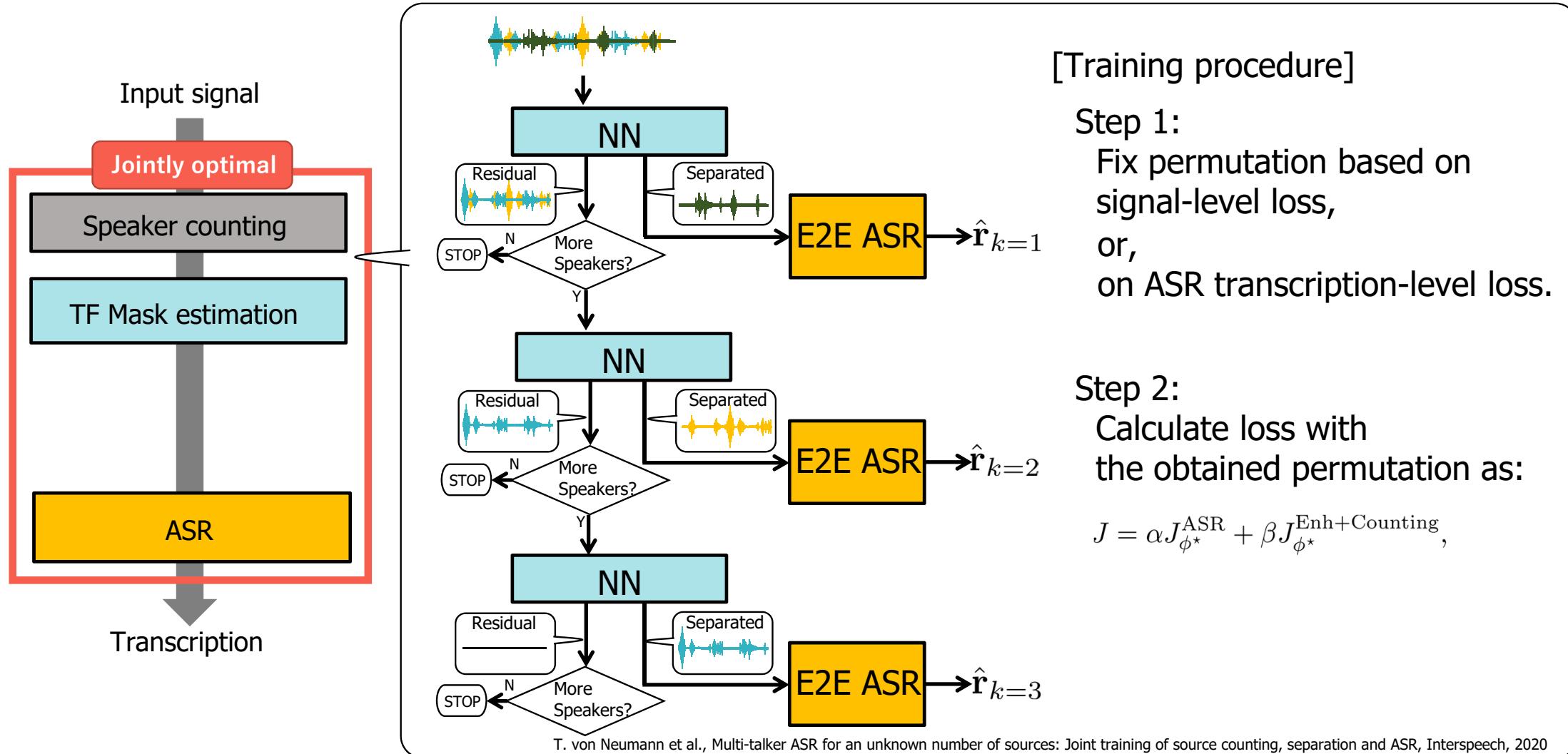
## (2) Speaker counting + TF Mask estimation + ASR



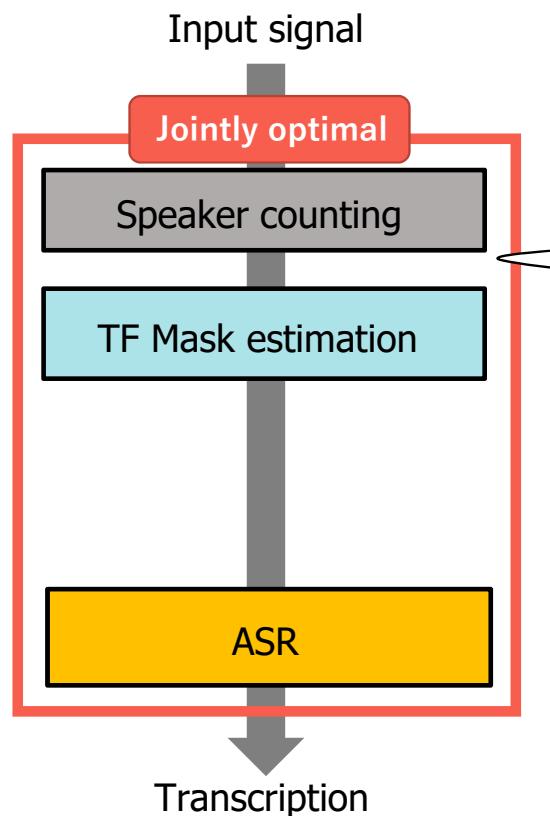
## (2) Speaker counting + TF Mask estimation + ASR



## (2) Speaker counting + TF Mask estimation + ASR



## (2) Speaker counting + TF Mask estimation + ASR



- Data: simulated mixture. 2- to 4-speaker fully overlapped speech.  
No noise and reverb.

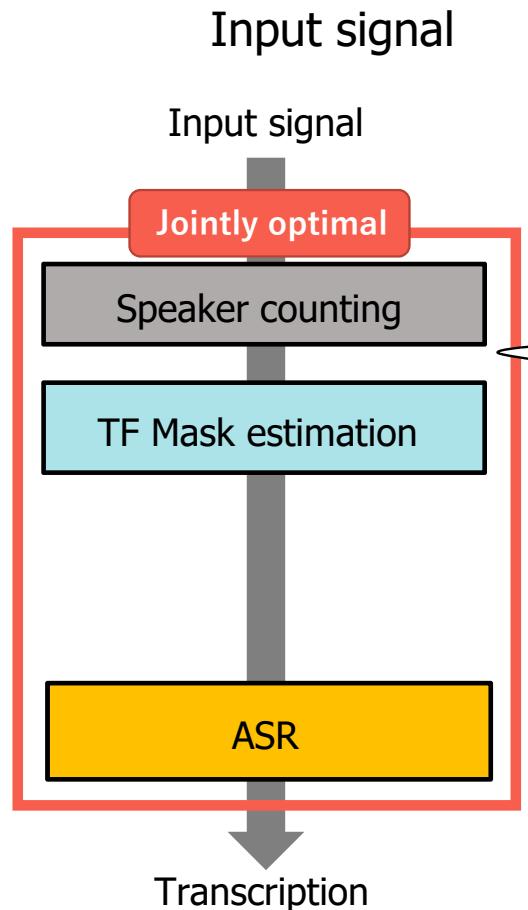
[Source counting accuracy (%)]

Model	Training data	Number of speakers			
		1	2	3	4
One-pass model (3 speaker output)	2 to 3 speakers	0.0	99.9	99.4	-
Recursive model	2 to 3 speakers	100.0	100.0	96.7	91.9

- Recursive separation
  - **Accurate counting accuracy** for seen cases i.e., 1- and 3-speaker cases
  - Could also **handle unseen cases** (4-speaker cases)

T. von Neumann et al., Multi-talker ASR for an unknown number of sources: Joint training of source counting, separation and ASR, Interspeech, 2020

## (2) Speaker counting + TF Mask estimation + ASR



- Data: simulated mixture. 2- to 4-speaker fully overlapped speech.  
No noise and reverb.

[ASR performance (SWER (%))]

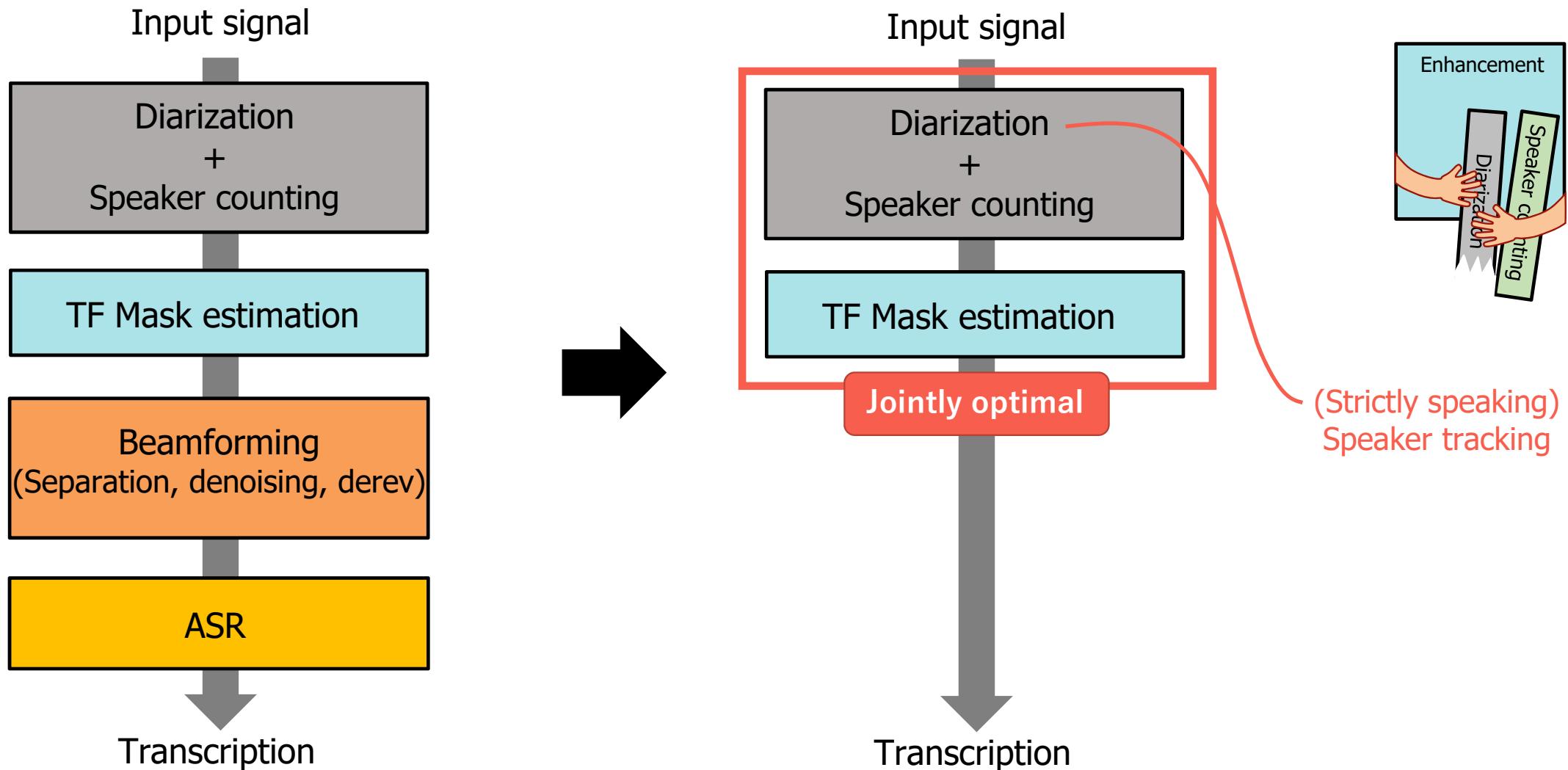
Model	Training data	Number of speakers		
		2	3	4
One-pass model (3 speaker output)	2 to 3 speakers	10.5	34.9	-
Recursive model	2 to 3 speakers	8.7	20.5	45.2

- ASR, separation, and speaker counting functionalities in one model.

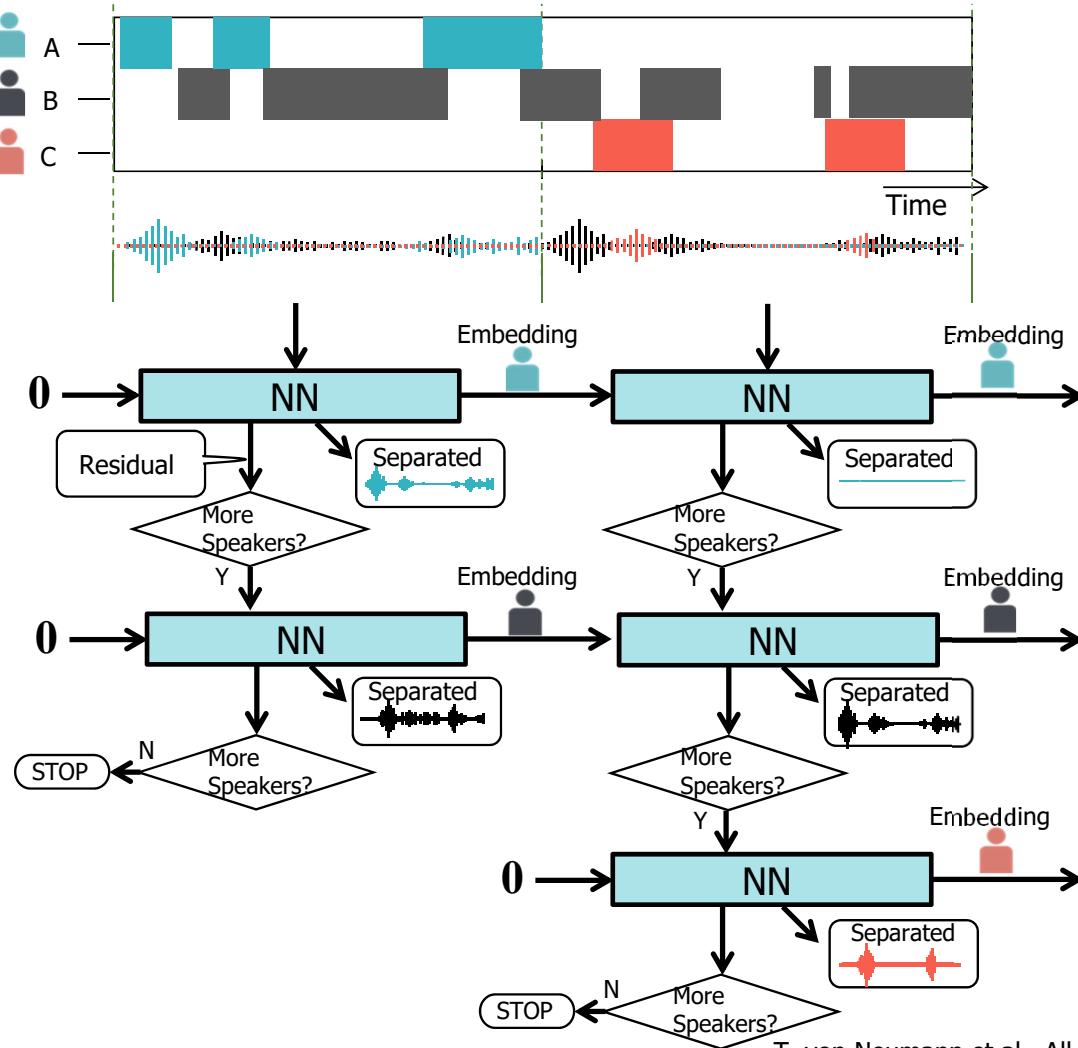
☺ Jointly optimal: 1-ch enhancement + speaker counting + ASR

T. von Neumann et al., Multi-talker ASR for an unknown number of sources: Joint training of source counting, separation and ASR, Interspeech, 2020

### (3) Diarization + Speaker counting + TF Mask estimation



### (3) Diarization + Speaker counting + TF Mask estimation



- Extension of recursive source separation [von Neumann+ 2019]
- At each block, perform recursive separation + embedding estimation
- Diarization can be performed with simple VAD
- An online model of an entire meeting
- Can be trained in a similar way as recursive source separation, i.e.,

$$J_{\phi^*}^{\text{Enh+Count}} = J_{\phi^*}^{\text{Enh}} + J^{\text{Count}}$$

T. von Neumann et al., All-neural online source separation, counting, and diarization for meeting analysis, ICASSP, 2019

### (3) Diarization + Speaker counting + TF Mask estimation

Input signal

Diarization  
+  
Speaker counting

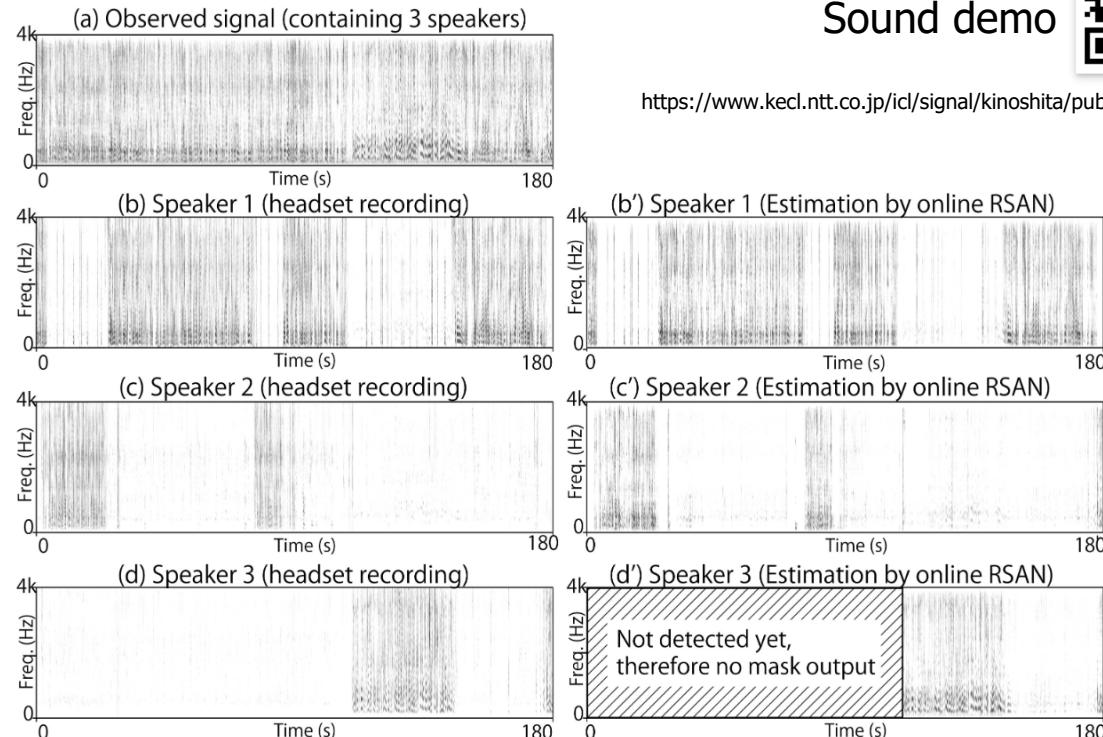
TF Mask estimation

Jointly optimal

Transcription

Data: Real meetings (4 to 6 speakers, about 15 min.),  
Significant noise and reverberation, overlap-ratio

[Source separation + speaker tracking results]



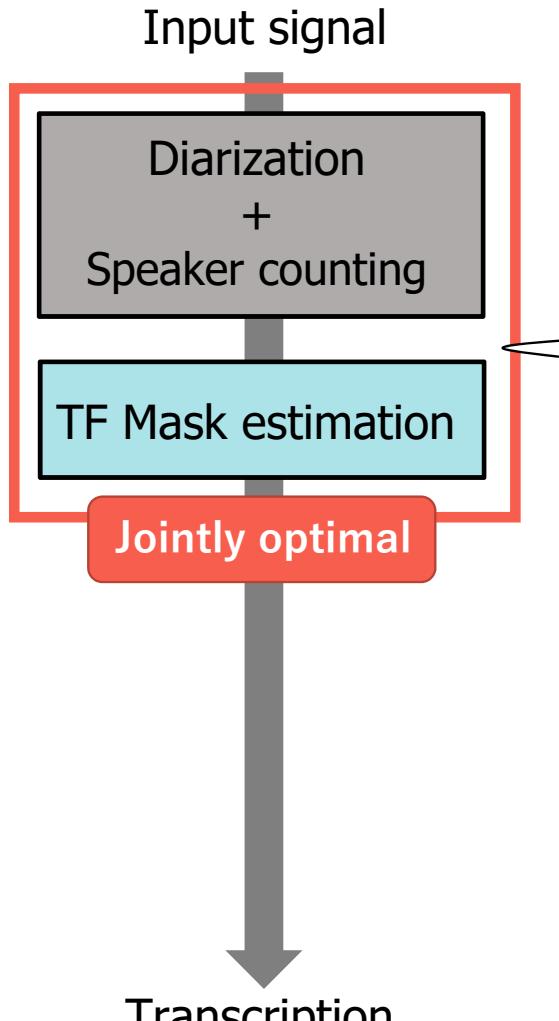
Sound demo



<https://www.kecl.ntt.co.jp/icl/signal/kinoshita/publications/ICASSP20/onlineRSAN/index.html>

K. Kinoshita et al., Tackling real noisy reverberant meetings with all-neural source separation, counting, and diarization system, ICASSP, 2020

## (2') Diarization + Speaker counting + TF Mask estimation



Data: Real meetings (4 to 6 speakers, each meeting is about 15 min.),  
Significant noise and reverberation

[Diarization results]

Method	DER (%)
X-vector clustering	65.0
Separation+Speaker tracking	51.4

☺ Outperformed x-vector clustering system

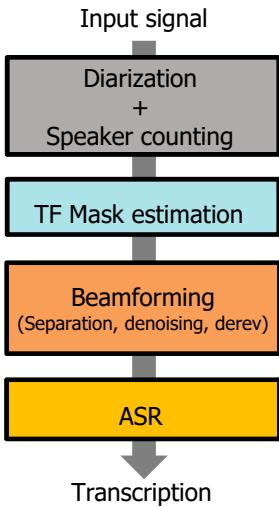
☹ The model is too big. Not very computationally efficient.

☺ Jointly optimal: 1-ch enhancement + speaker counting +speaker tracking  
(similar to diarization)

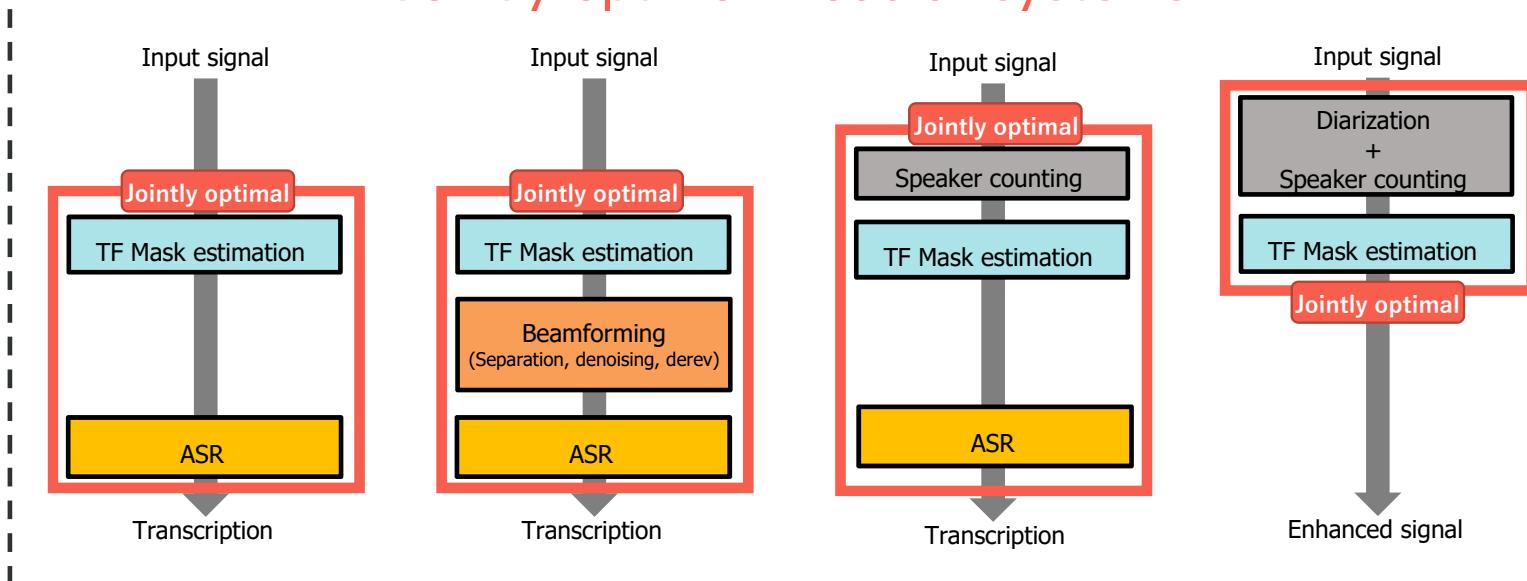
K. Kinoshita et al., Tackling real noisy reverberant meetings with all-neural source separation, counting, and diarization system, ICASSP, 2020

# Short summary of “Enhancement + X”

## Baseline system



## Jointly optimal modular systems

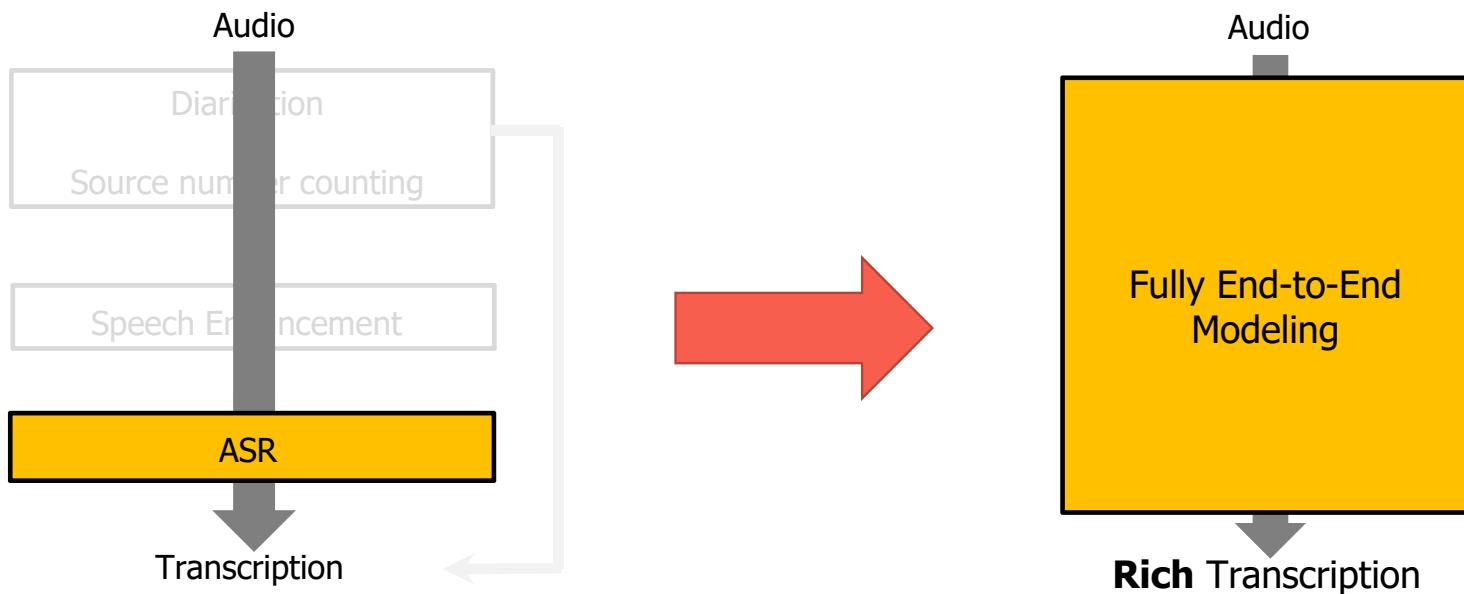


- 😊 Jointly optimal systems successfully outperformed cascaded systems.
- 😢 Each module is relatively large and sometimes computationally inefficient, which makes it difficult to train the system with large amount of data.  
→ Need more efficient training schemes, or more powerful computation resources.

### 3. A new research trend: Jointly optimal systems

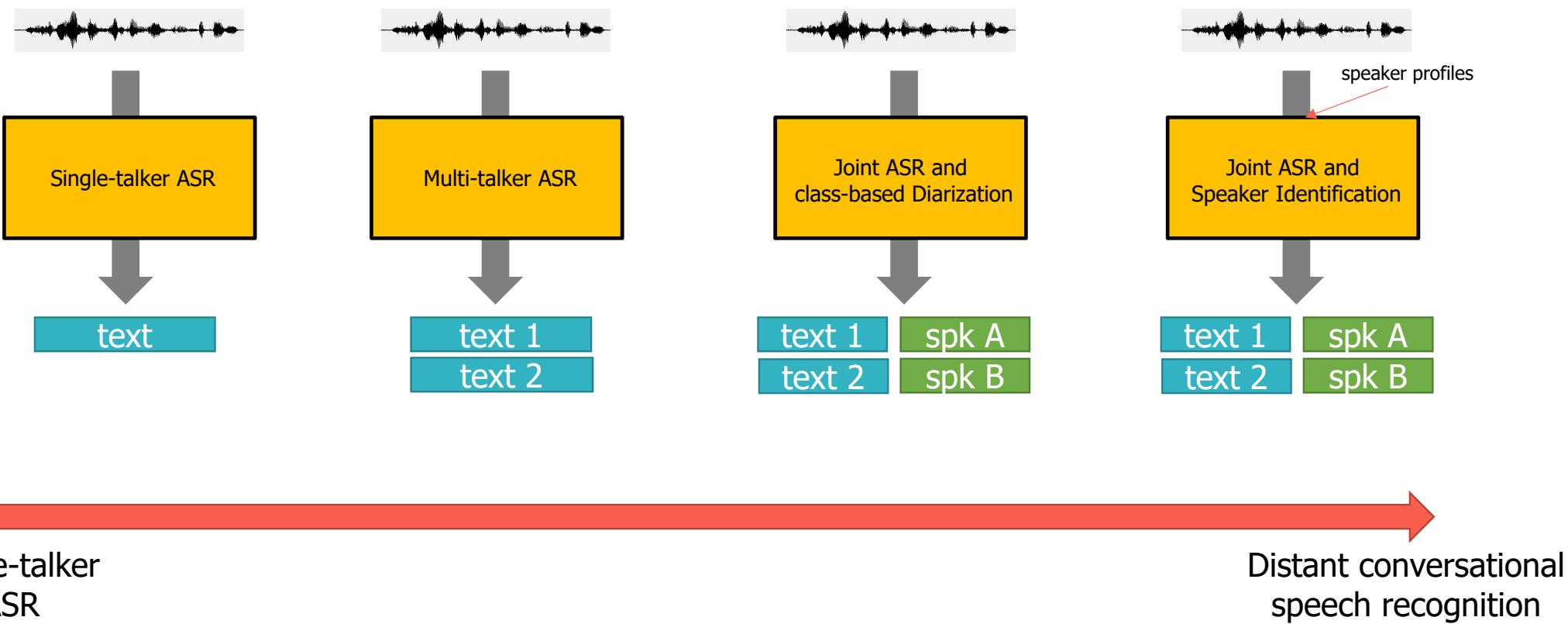
- 3.1. Diarization +  $x$
- 3.2. Enhancement +  $x$
- 3.3. ASR +  $x$

# ASR + $x$ : Enriching ASR System towards Rich Transcription

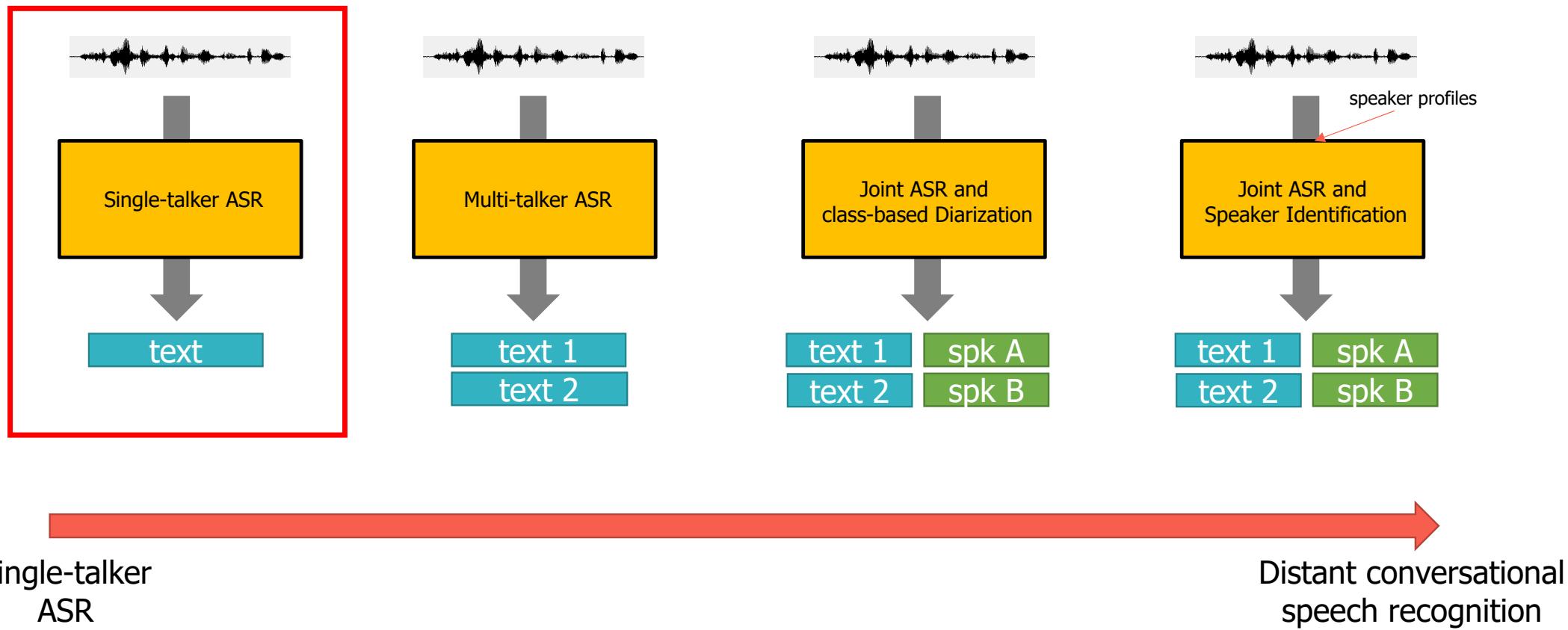


- 😊 Can keep the whole system simpler and smaller (i.e., computationally efficient, less parameters).

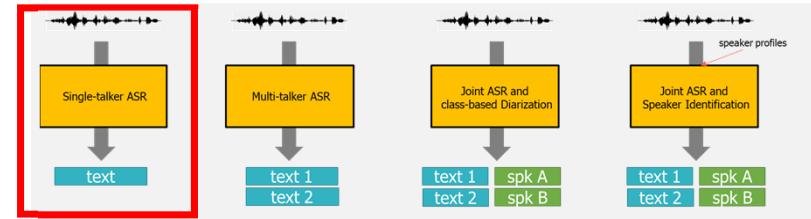
# What has been achieved for “ASR + $x$ ”



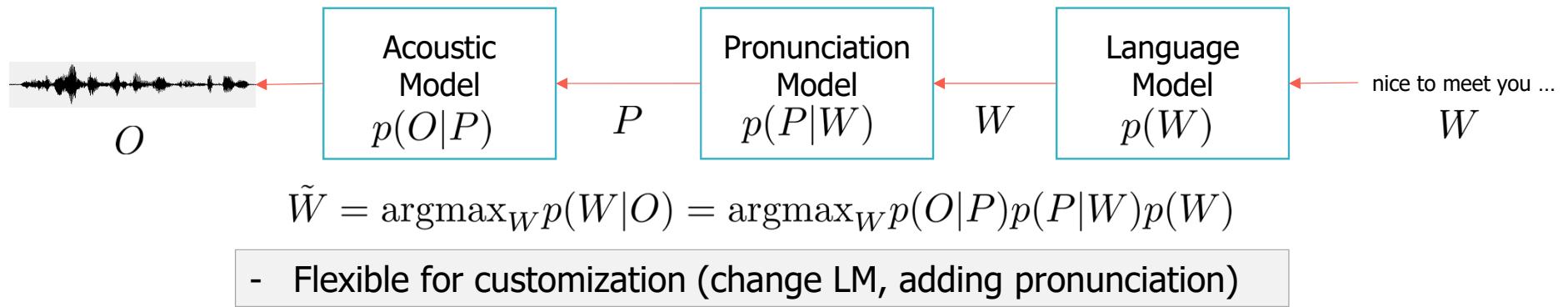
# What has been achieved for “ASR + $x$ ”



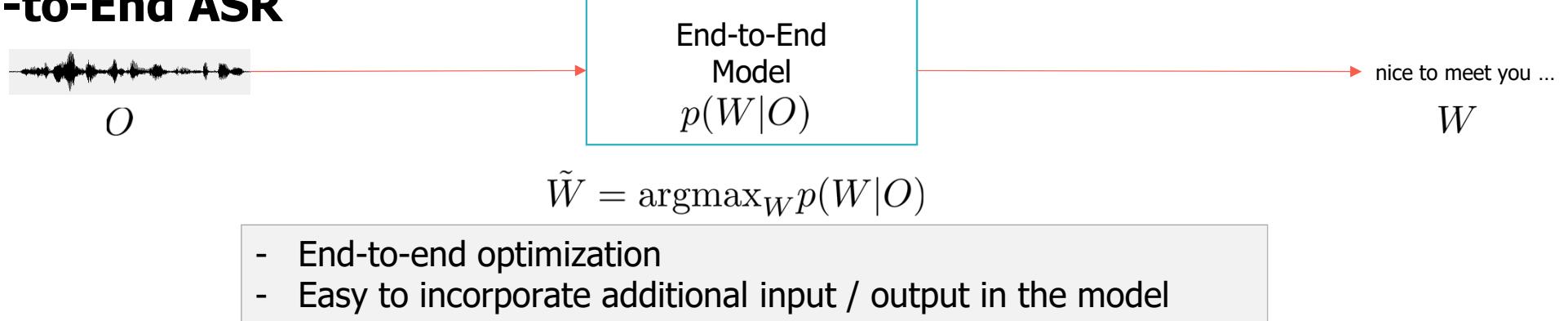
# Hybrid ASR and End-to-End ASR



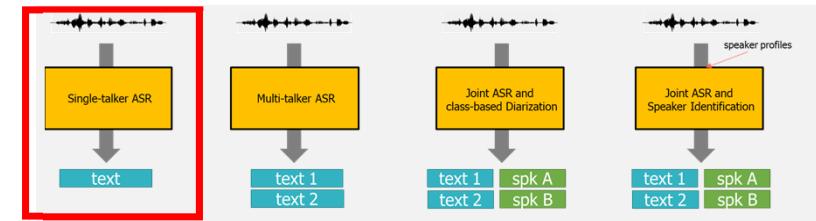
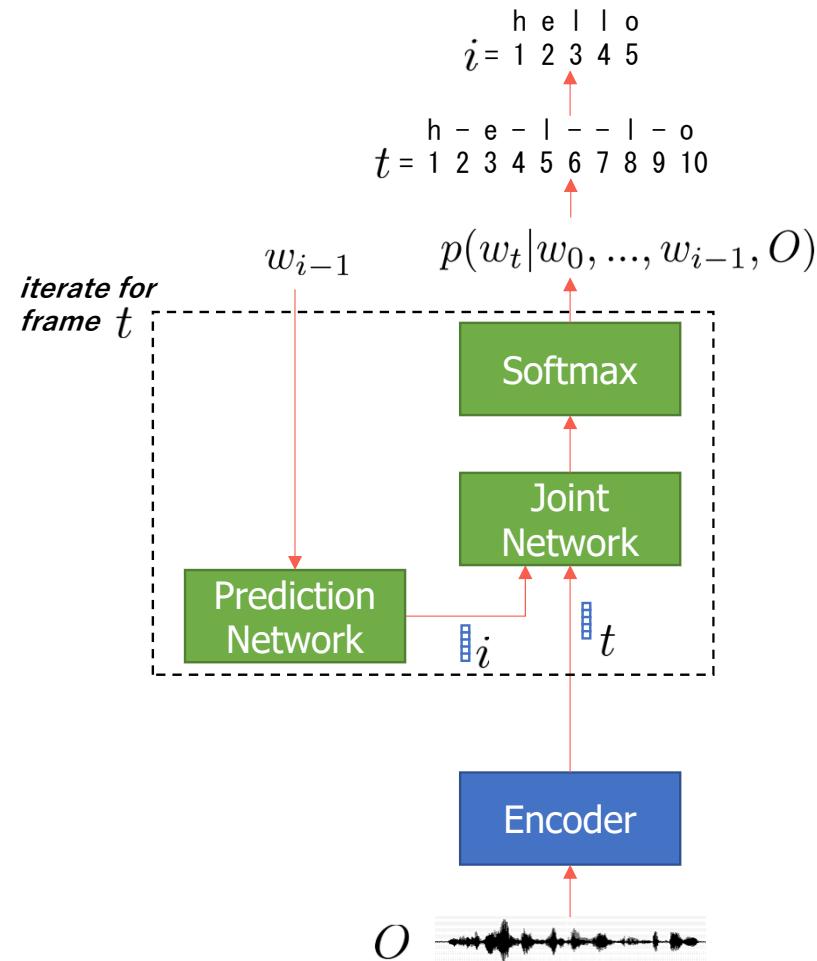
## Hybrid ASR



## End-to-End ASR



# RNN Transducer and Attention Encoder Decoder

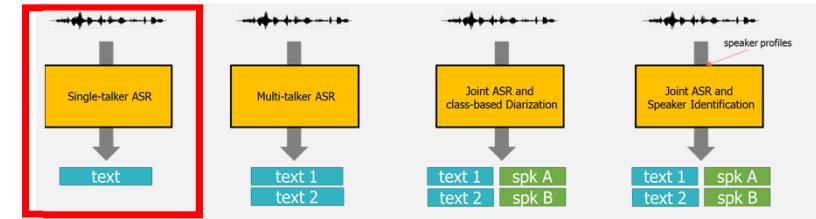
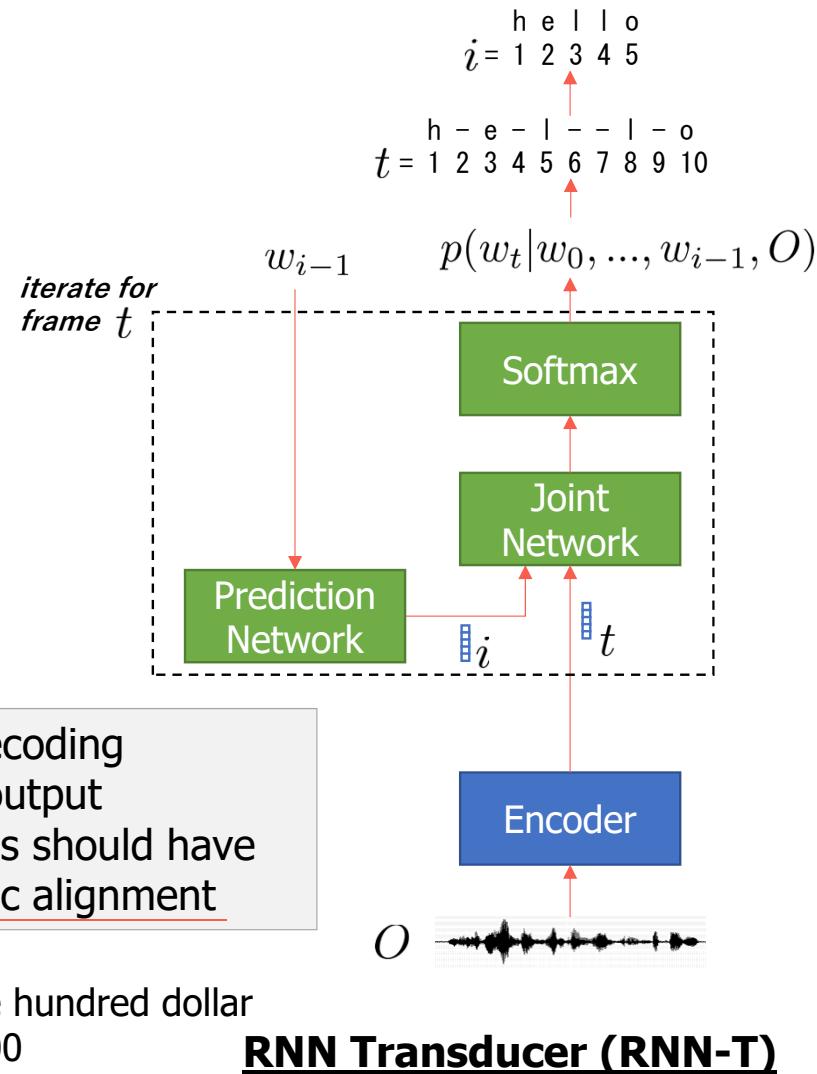


## RNN Transducer (RNN-T)

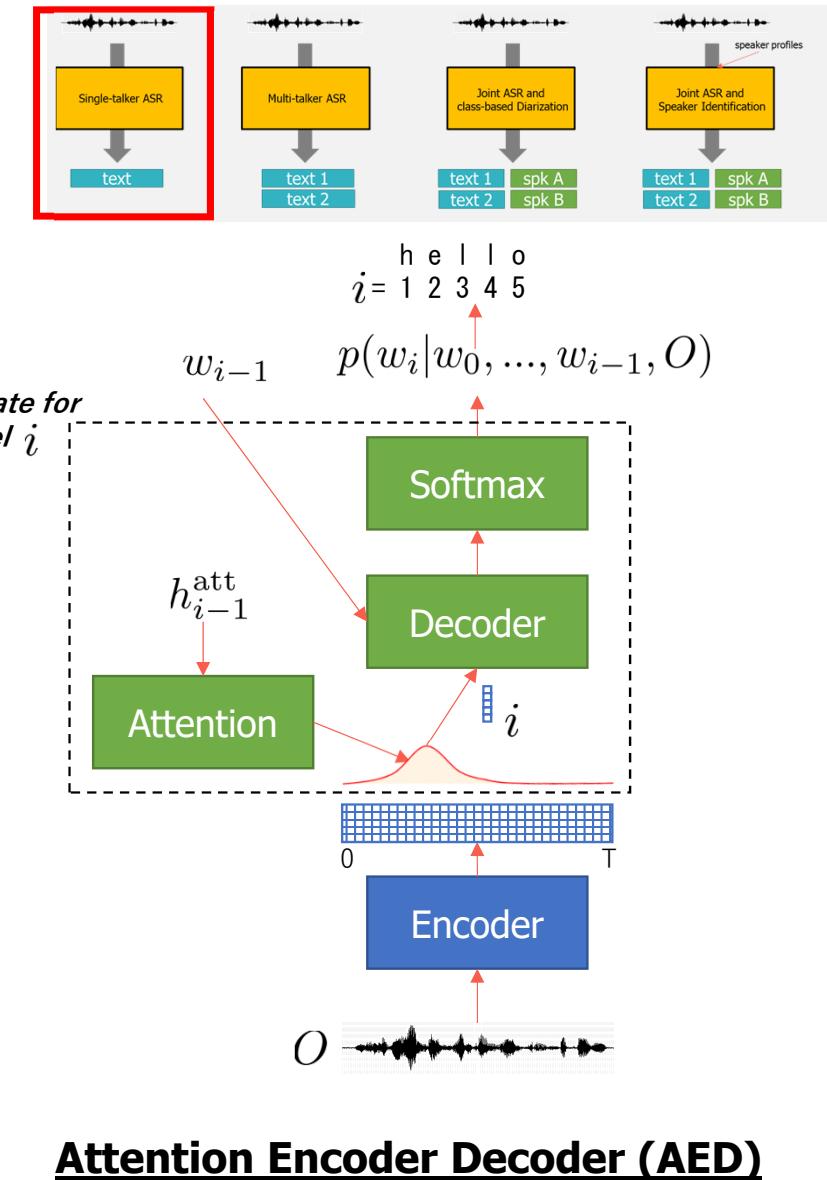
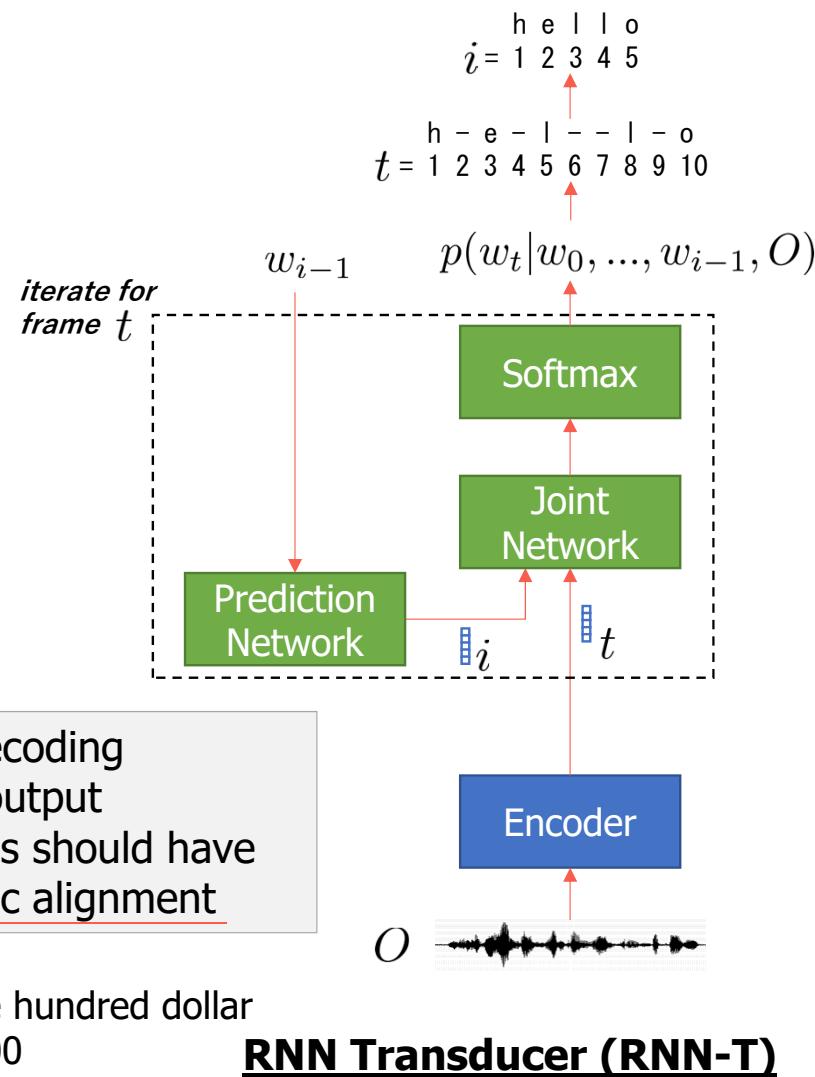
3.3. A new trend toward jointly optimal systems: ASR +  $x$

P. 187

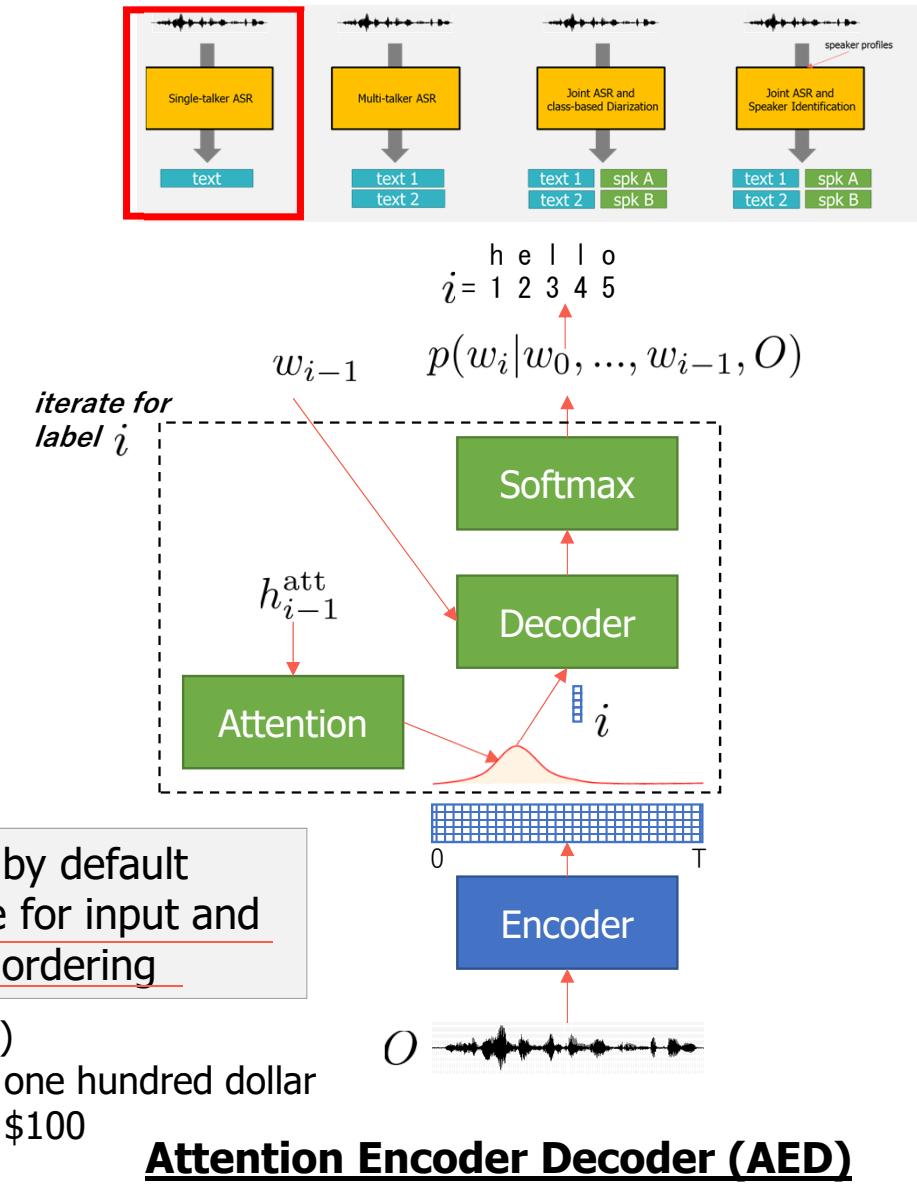
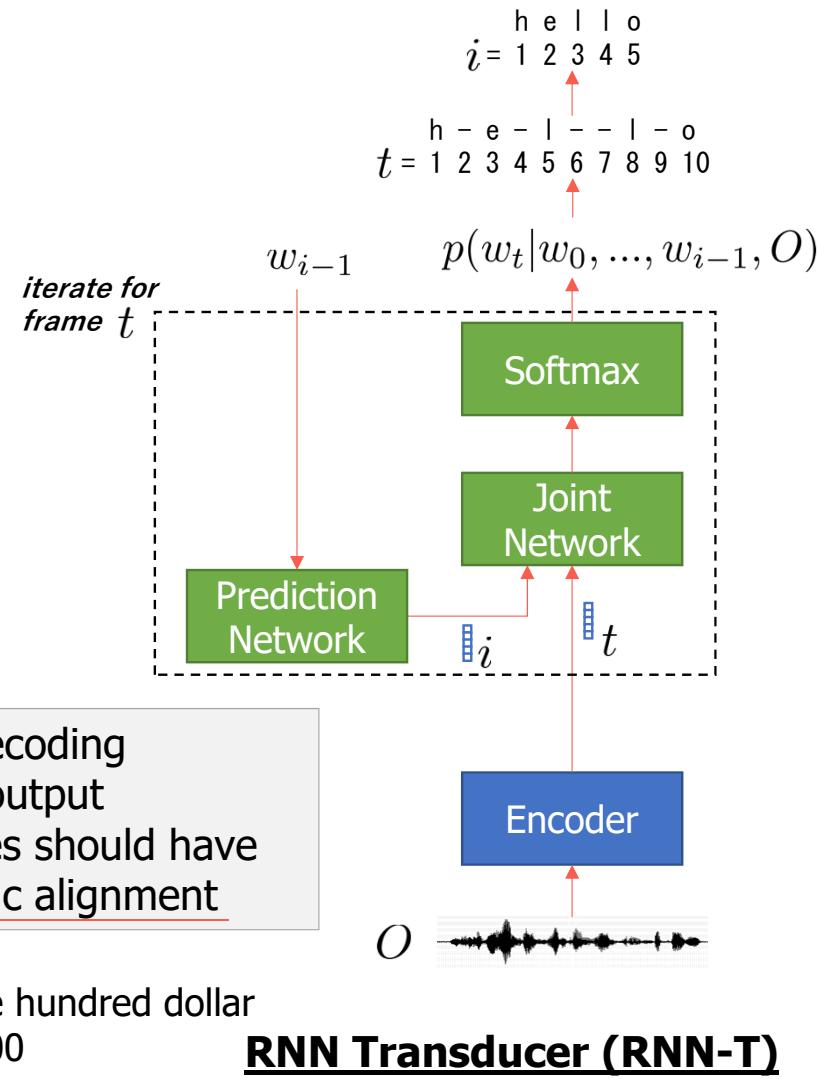
## RNN Transducer and Attention Encoder Decoder



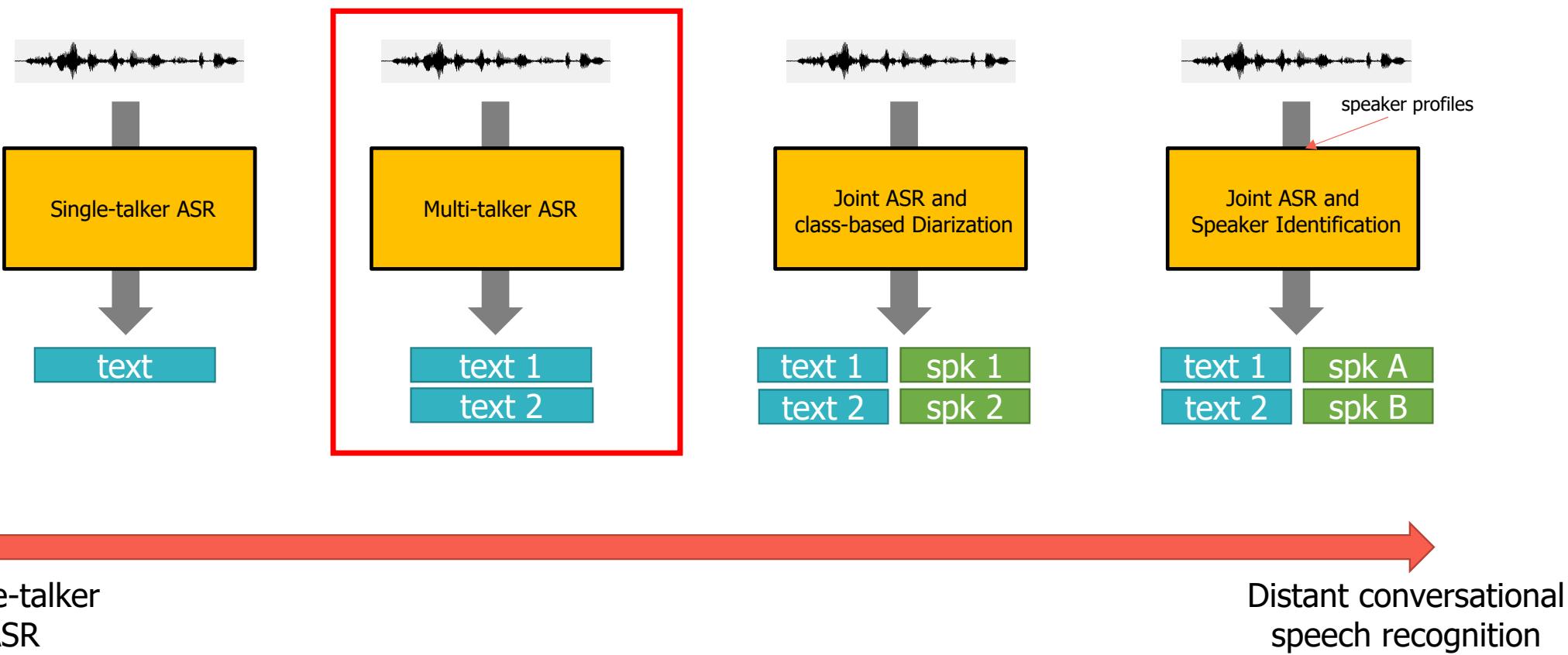
## RNN Transducer and Attention Encoder Decoder



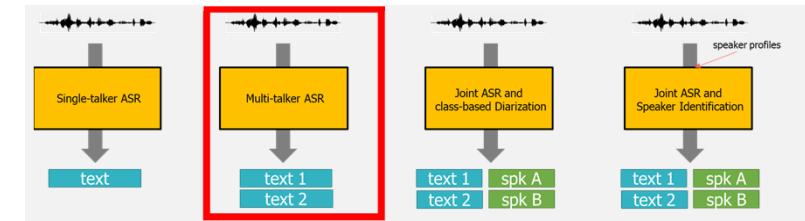
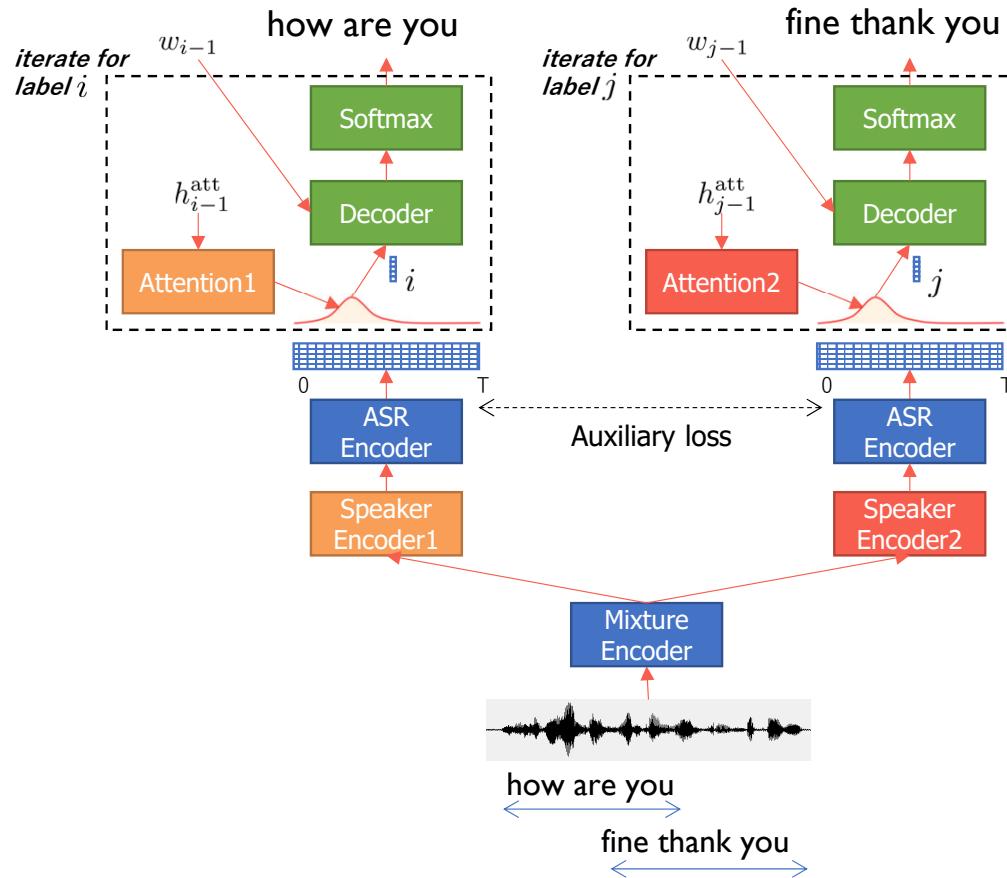
## RNN Transducer and Attention Encoder Decoder



# What has been achieved for “ASR + $x$ ”



# A Purely End-to-end System for Multi-speaker Speech Recognition [Seki+ 2018]



Two output model with *speaker-dependent encoder*

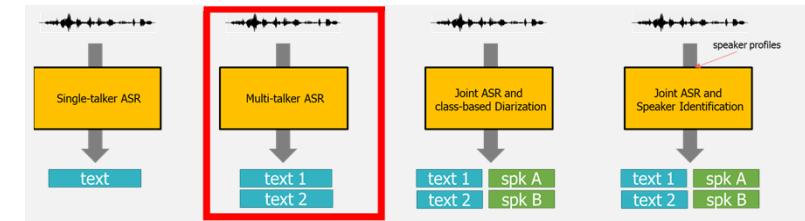
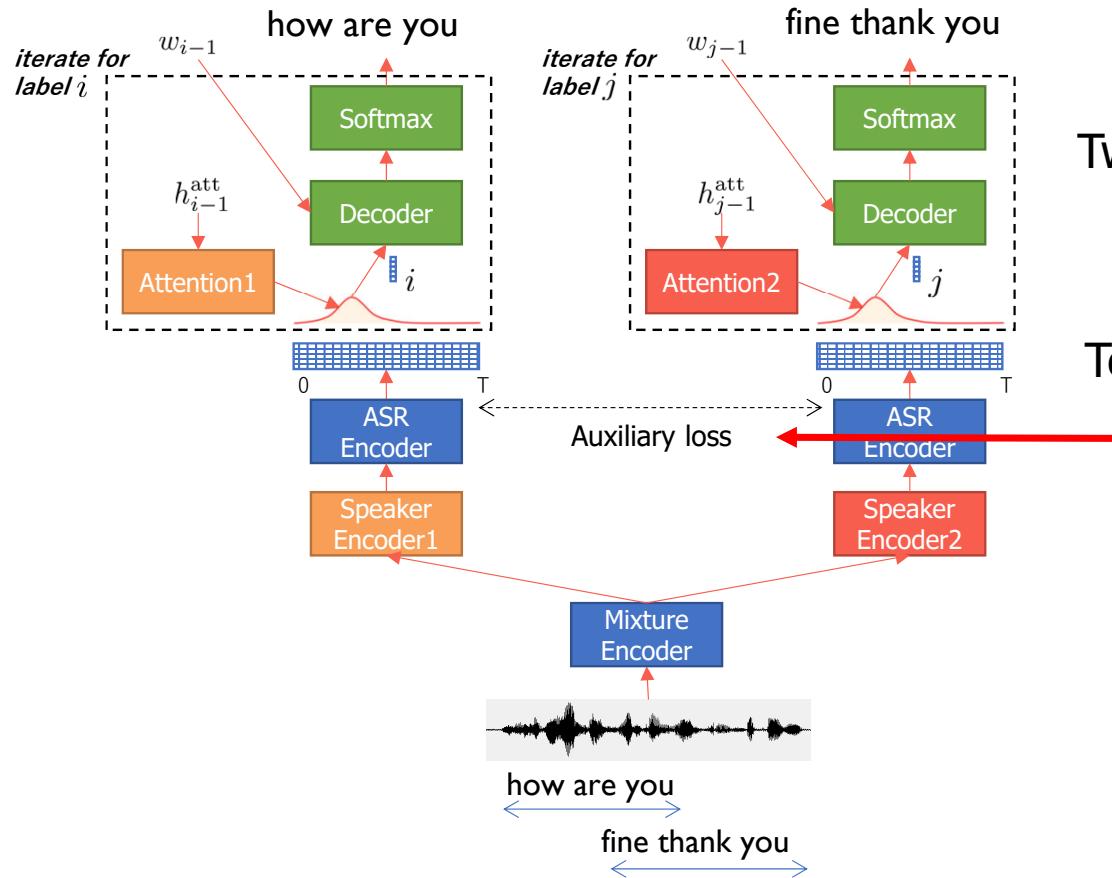


Simpler training.

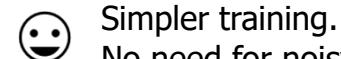
No need for noisy-clean audio pair for training.

- ❑ H. Seki et al., A Purely End-to-end System for Multi-speaker Speech Recognition. *ACL*, 2018.
- ❑ X. Chang et al., End-to-End Monaural Multi-speaker ASR System without pretraining, *ICASSP* 2019.

# A Purely End-to-end System for Multi-speaker Speech Recognition [Seki+ 2018]



Two output model with *speaker-dependent encoder*



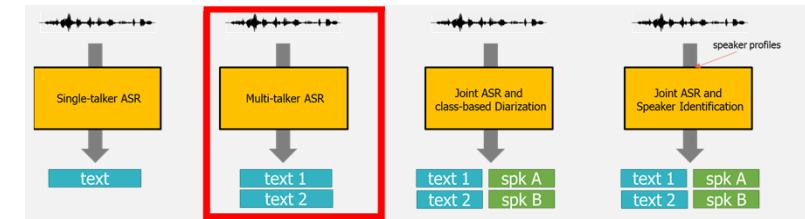
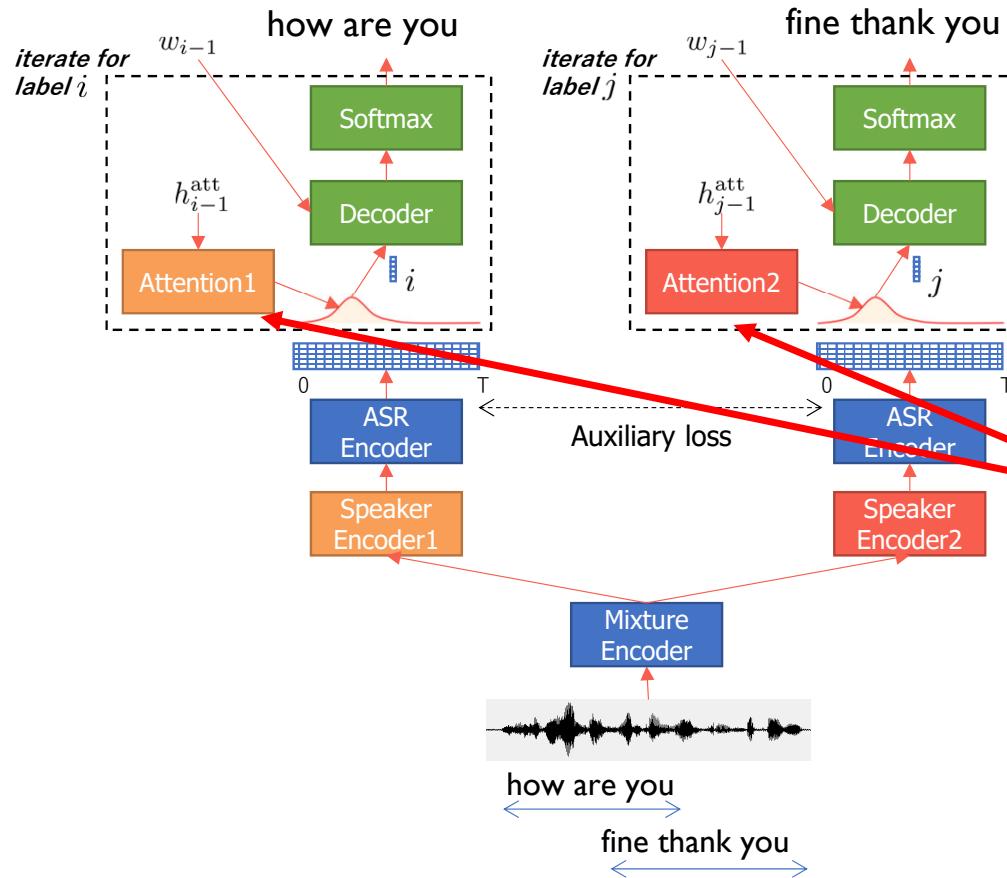
Simpler training.  
No need for noisy-clean audio pair for training.

## Techniques

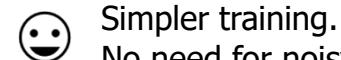
- KL-divergence-based auxiliary loss to make the distance between outputs of ASR encoders different [Seki+ 2018]

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# A Purely End-to-end System for Multi-speaker Speech Recognition [Seki+ 2018]



Two output model with *speaker-dependent encoder*



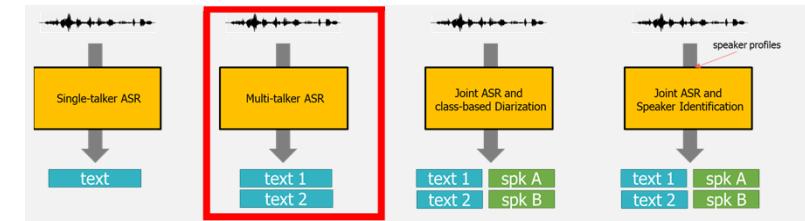
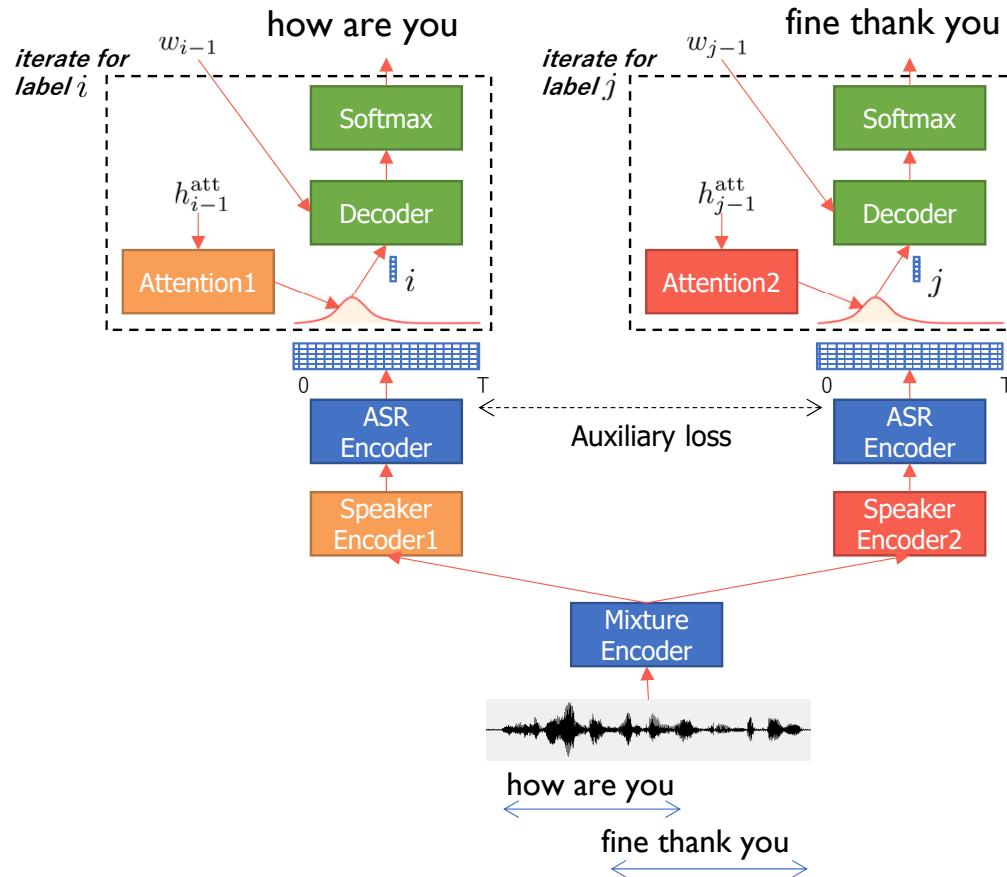
Simpler training.  
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- Speaker-specific attention layers are better [Chang+ 2019]

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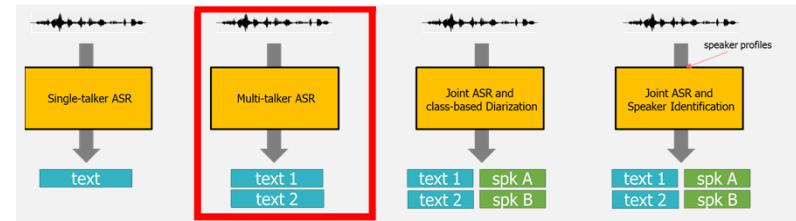
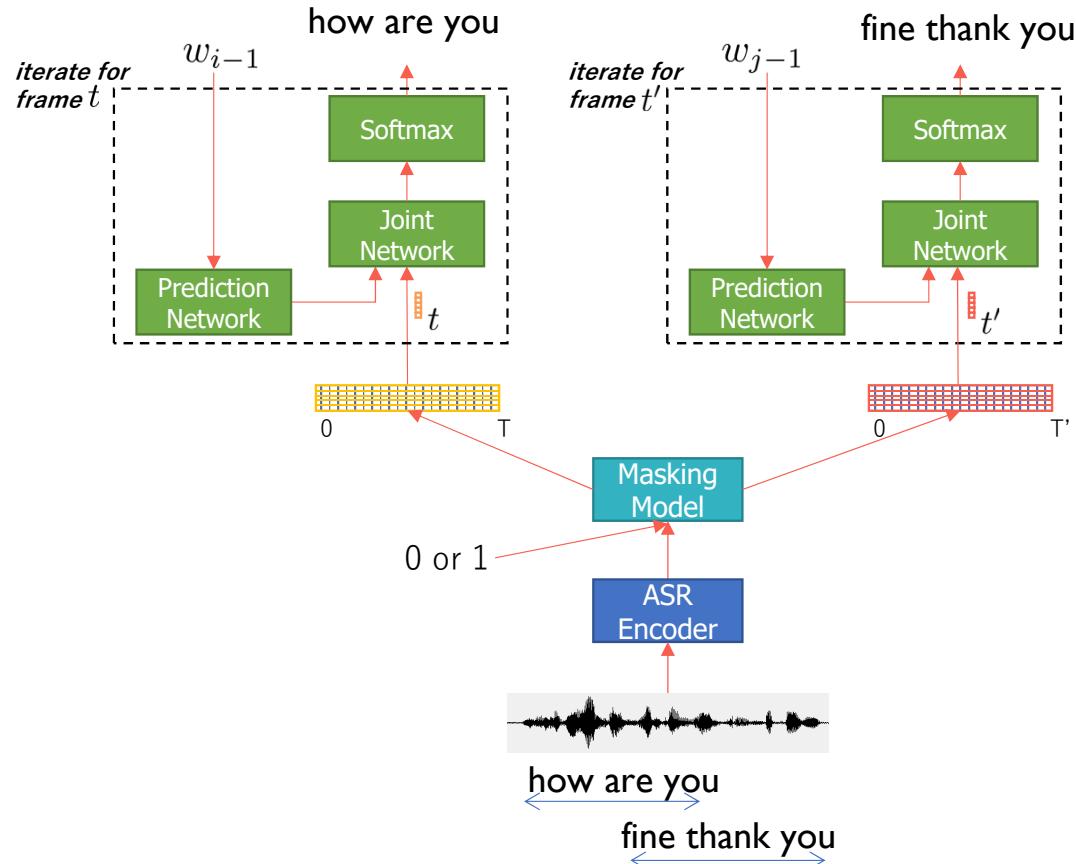
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SWER (%) on 2-speaker mixed WSJ0 corpus (from [Chang+ 2019])

Model	Avg. WER
DPCL+ASR [17]	30.8
PIT-ASR [24]	28.2
End-to-end ASR (Char/Word-LM) [26]	28.2
Proposed End-to-end ASR with SPA (Word LM)	<b>25.4</b>

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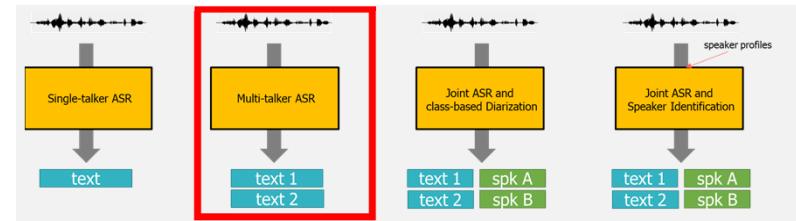
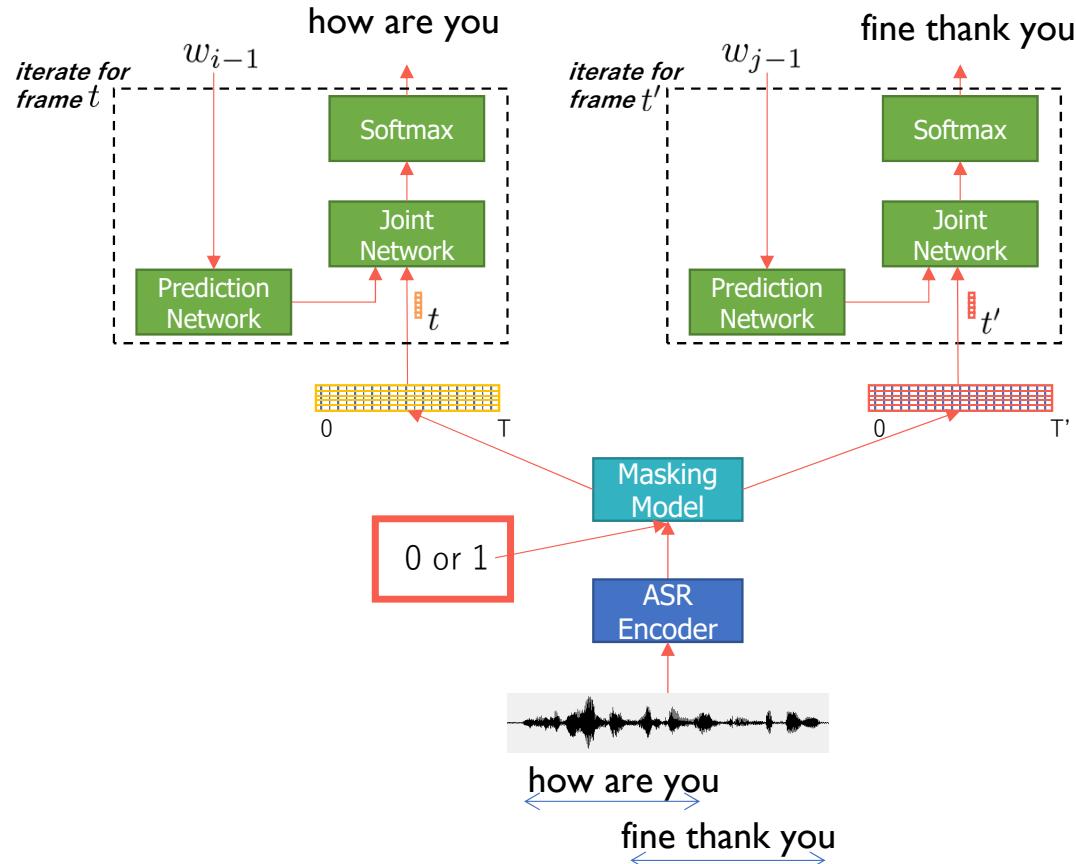
# End-to-end Multi-talker Overlapping Speech Recognition [Tripathi+ 2020]



- Extension for RNN-T (yet offline model)
- Masking model to recognize two speakers

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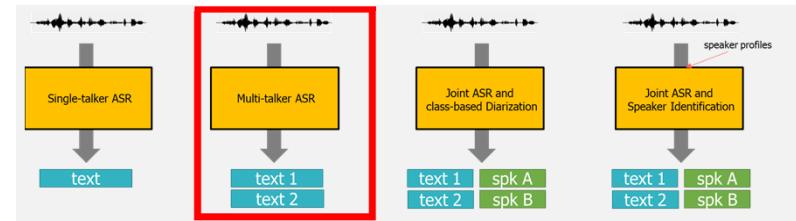
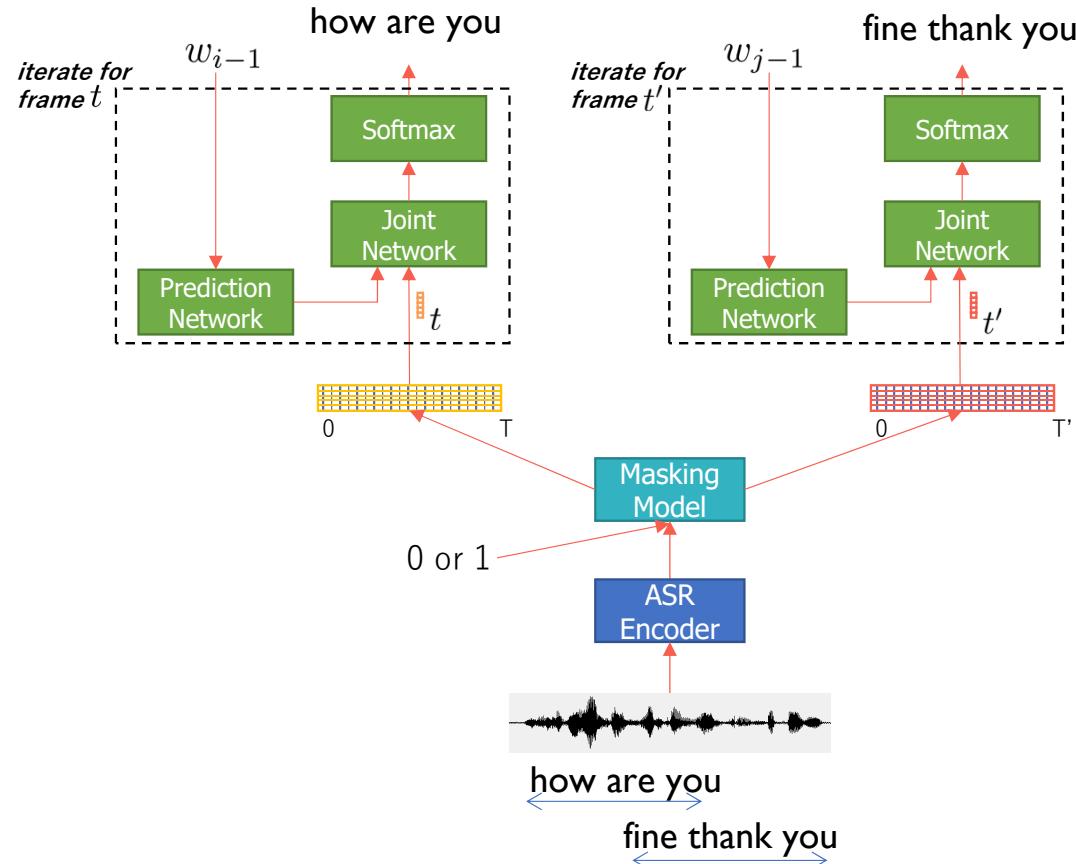
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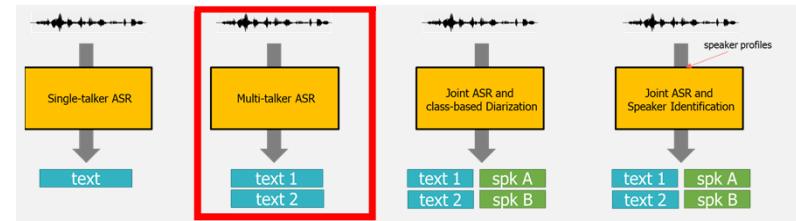
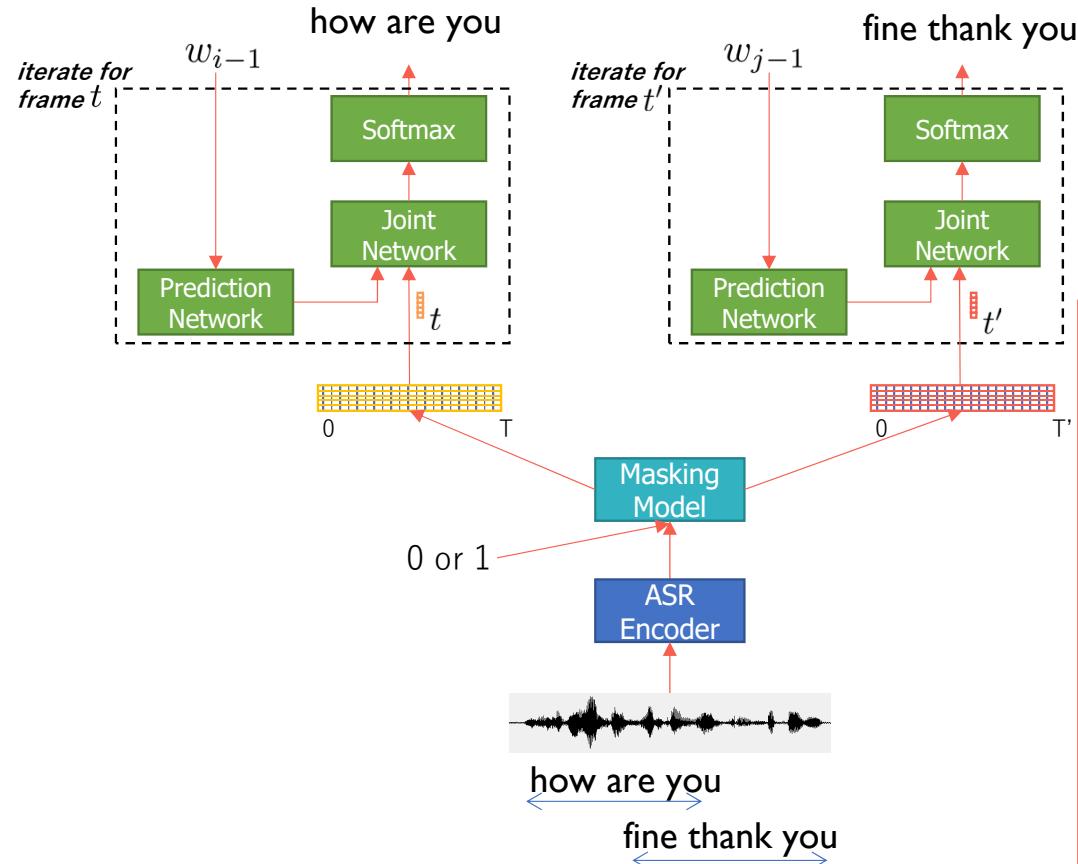


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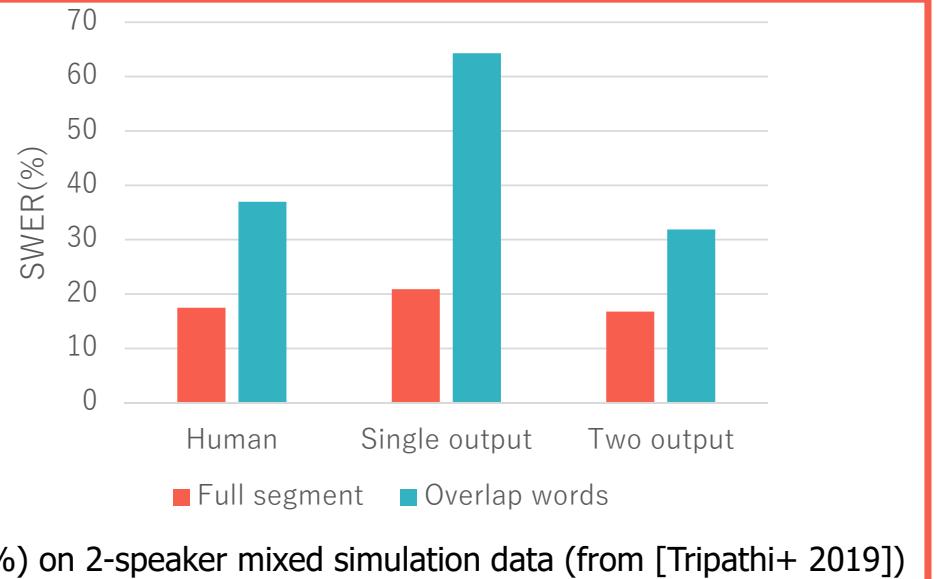
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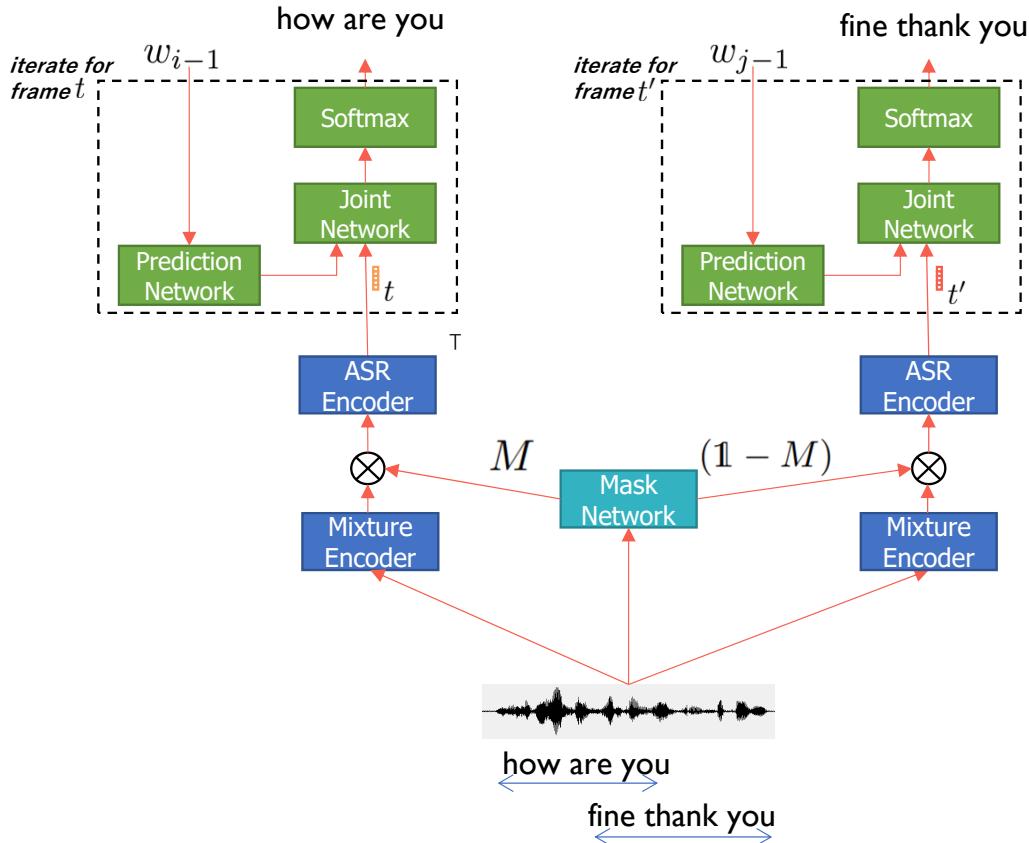


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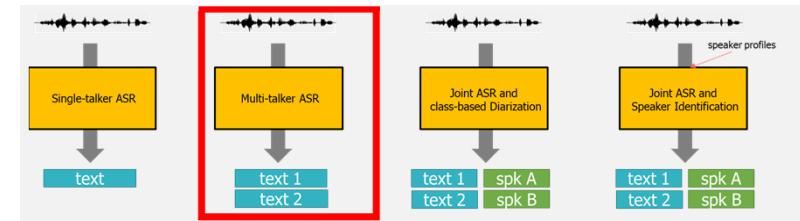
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# Streaming end-to-end multi-talker speech recognition [Lu+ 2021][Sklyar+ 2021]



## Streaming model based on RNN-T

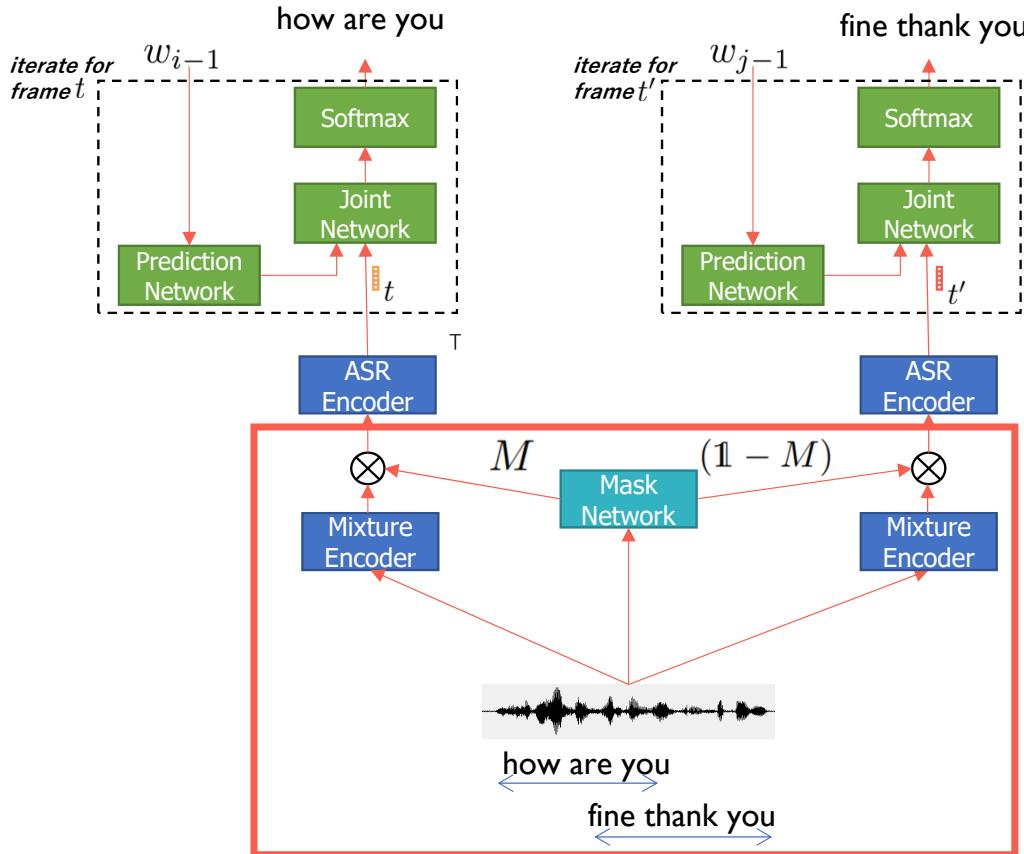
- [Lu+ 2021] Mask-based model or SD-encoder
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## Streaming Unmixing and Recognition Transducer (SURT) with mask-based unmixing model [Lu+ 2021]

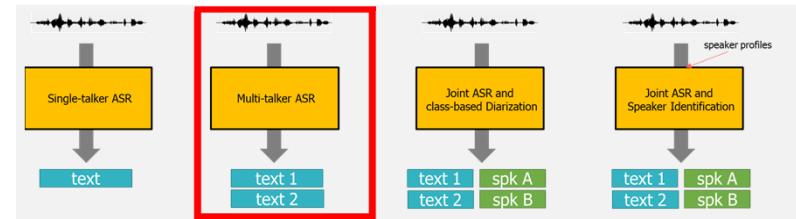
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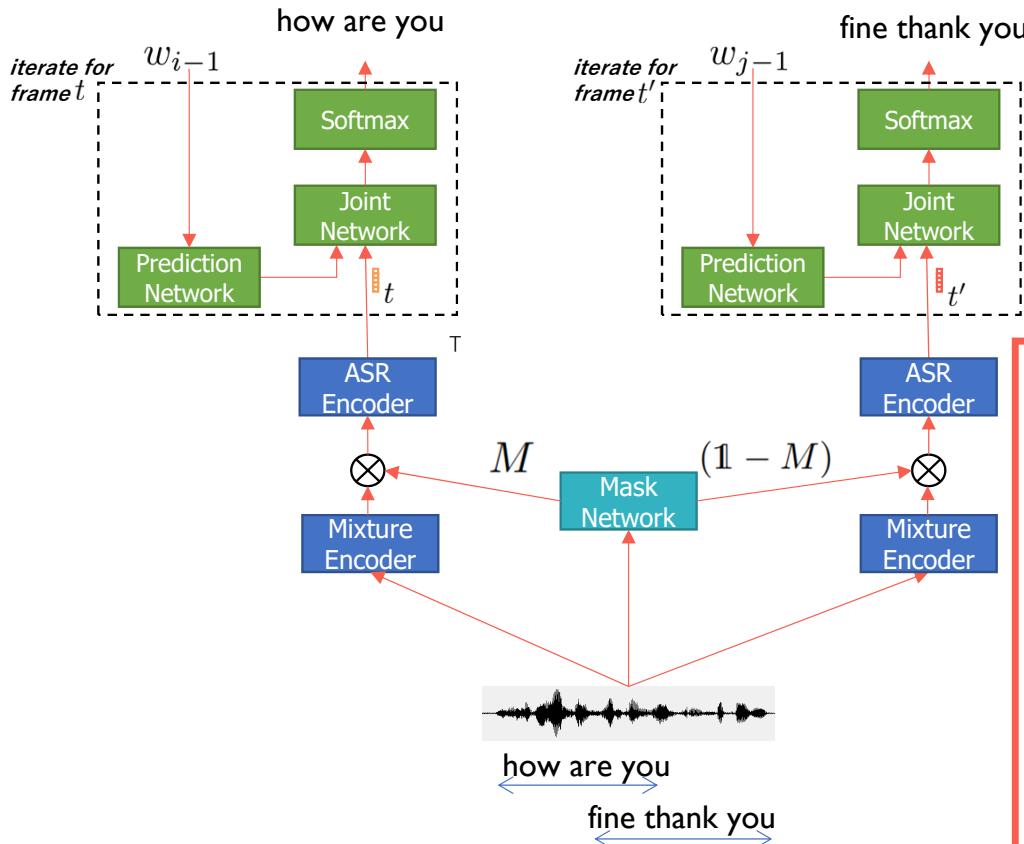
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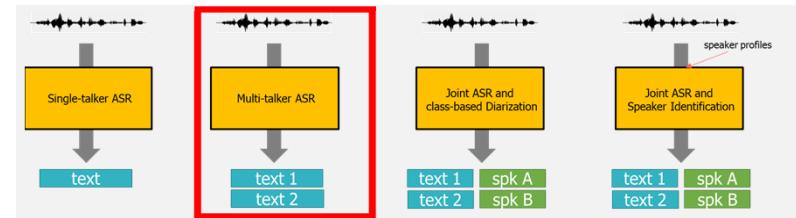
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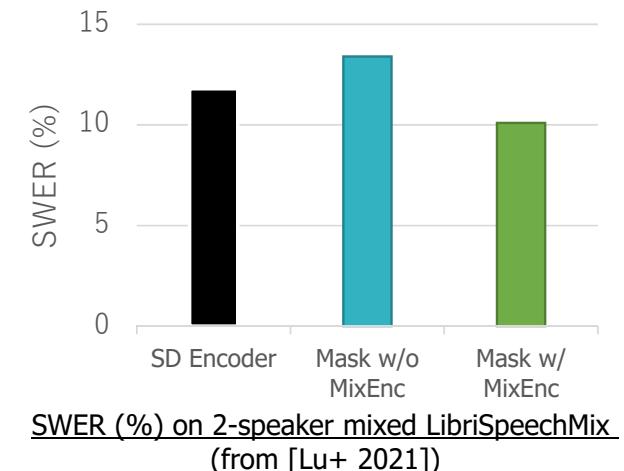
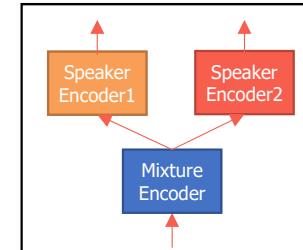
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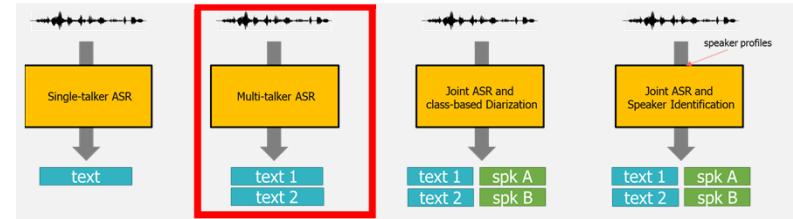
### Techniques

- Mask-based model (left figure) was better than SD-encoder-based model [Lu+ 2021]

Speaker-dependent (SD)-encoder



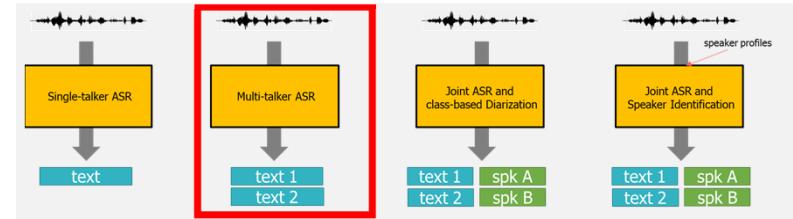
# Reducing the computational cost of Permutation Invariant Training (PIT)



- A multi-output branch model usually uses PIT, which requires  $O(S^3)$  computation w.r.t. the number of outputs  $S$

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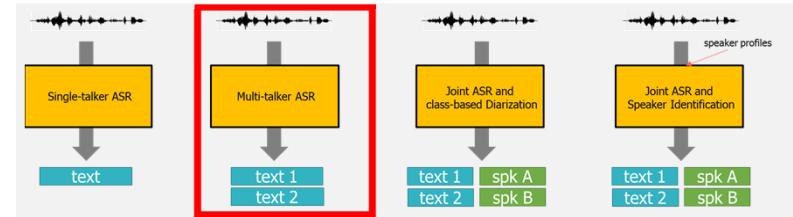
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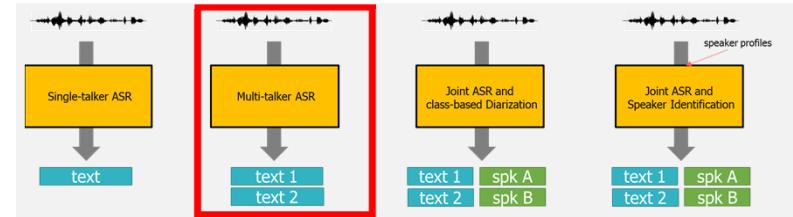
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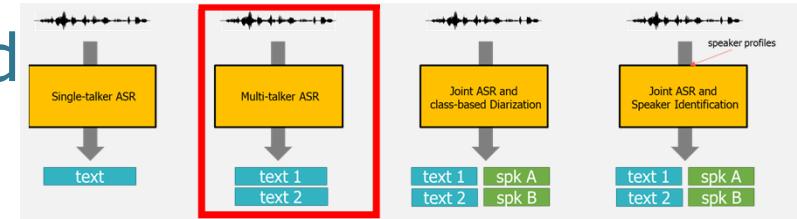
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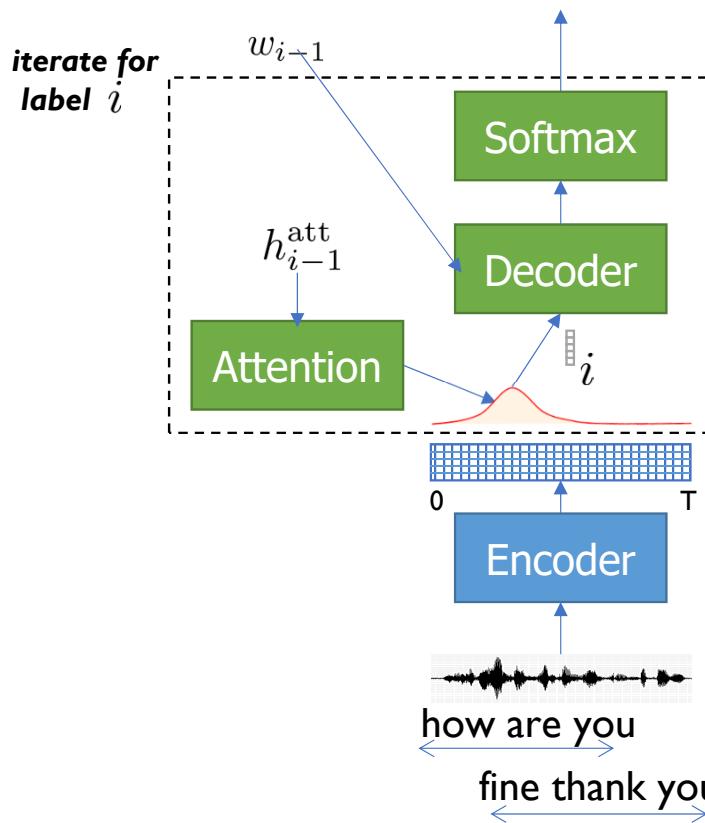
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- Use start time of each utterance to determine the order →  $O(S)$ 
  - Used in [Tripathi+ 2020] without comparison.
  - Later, [Lu+ 2021] and [Sklyar+ 2021] concurrently showed this heuristic error assignment is **as good as PIT**
    - Heuristic Error Assignment Training (HEAT) [Lu+ 2021]; Reported slightly better result than PIT
    - Deterministic Assignment Training (DAT) [Sklyar+ 2021]; Reported slightly worse result than PIT

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# Serialized Output Training (SOT) for End-to-End Overlapped Speech Recognition [Kanda+ 2020]



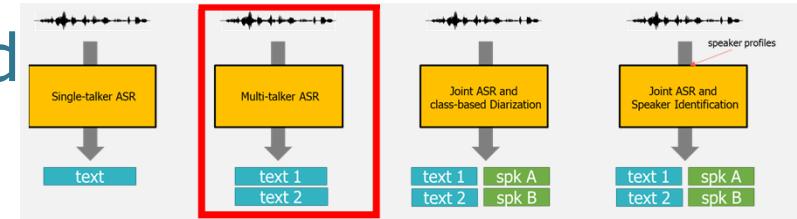
how are you <sc> fine thank you <eos>



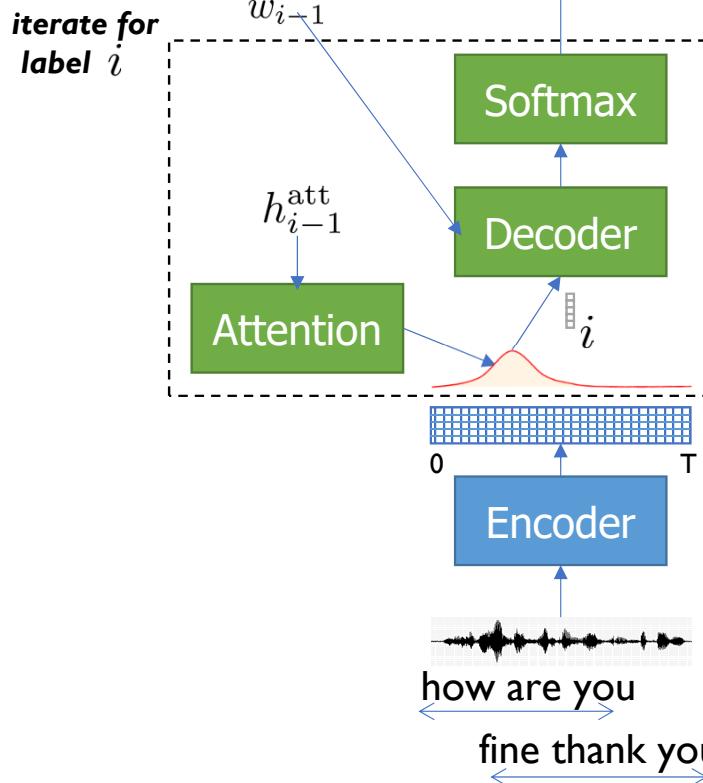
N. Kanda et al., Serialized Output Training for End-to-End Overlapped Speech Recognition. Interspeech, 2020.

## 3.3. A new trend toward jointly optimal systems: ASR + $x$

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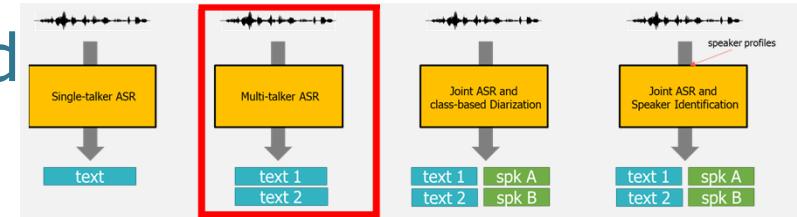
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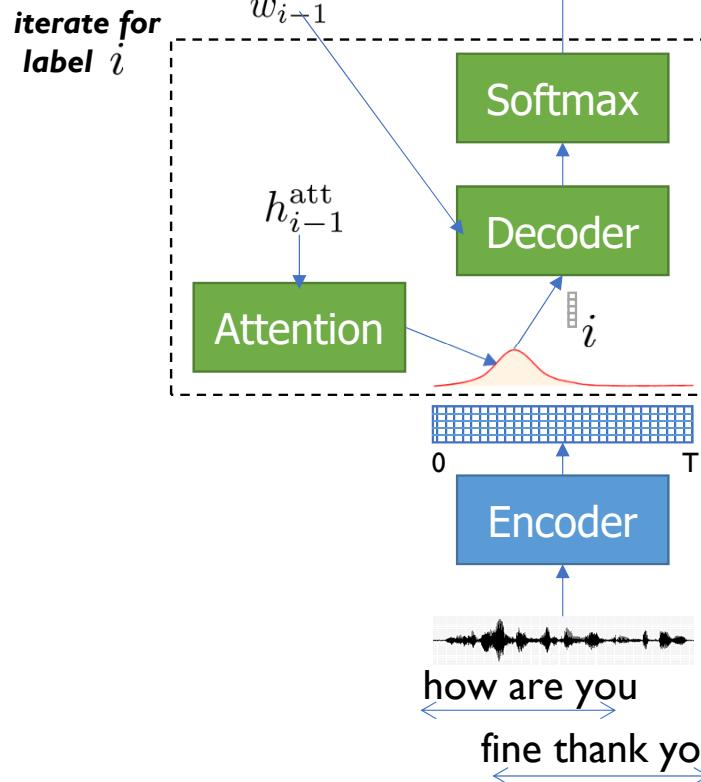
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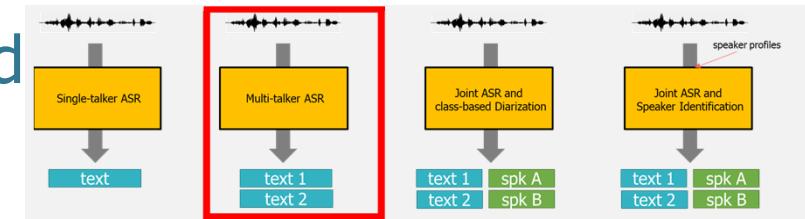
Can recognize **any number of speakers**  
Can **count the number of speakers**

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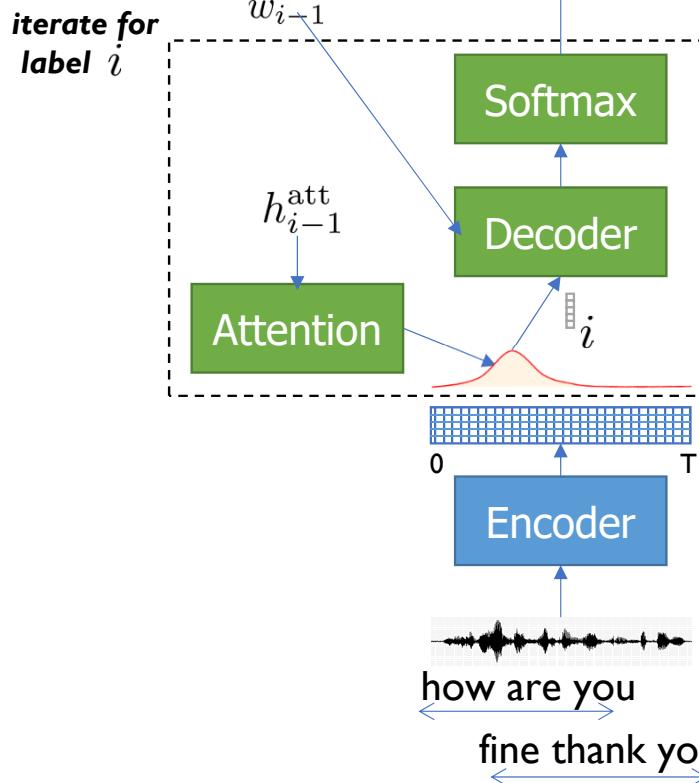
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**First-In, First-Out (FIFO)**



how are you <sc> fine thank you <eos>



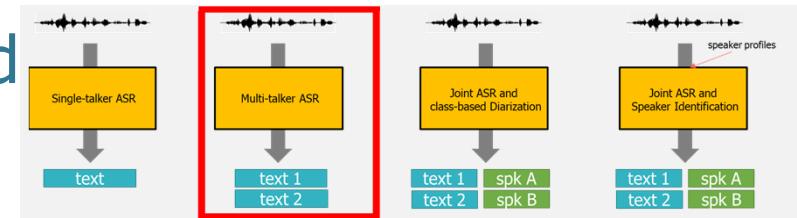
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- ☺ FIFO training for O(S) training (Proposed concurrently with [Tripathi+ 2020] )

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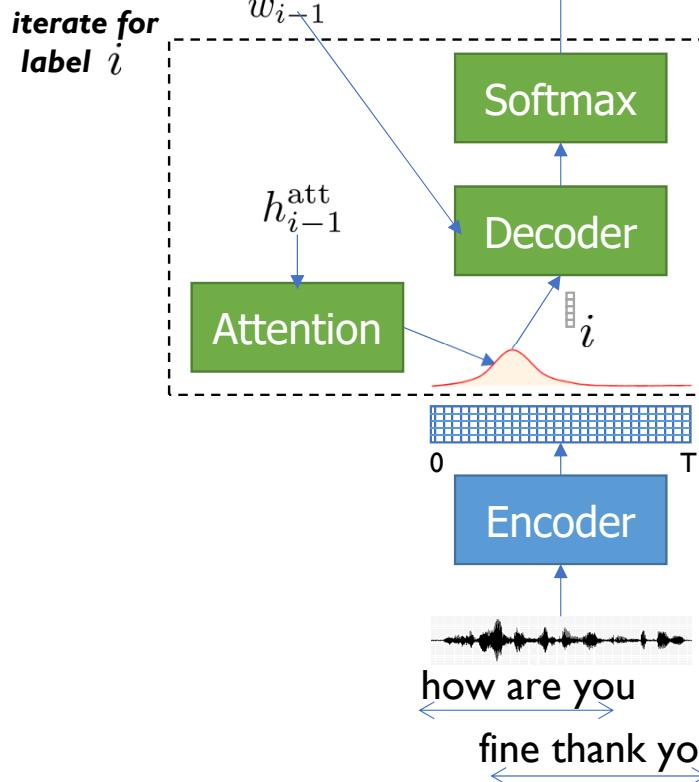
P. 210

# Serialized Output Training (SOT) for End-to-End Overlapped Speech Recognition [Kanda+ 2020]



**First-In, First-Out (FIFO)**

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SWER (%) on LibriSpeechMix (from [Kanda+ 2020])

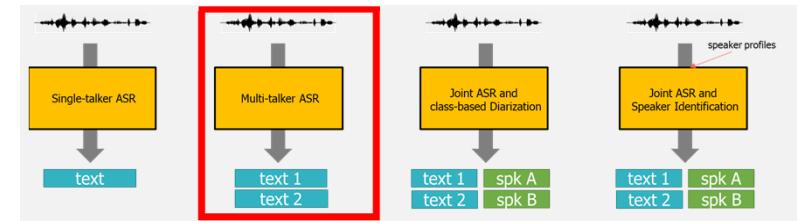
Model (1024-dim)	# of Speakers in Training Data	# of Params.	# of Speakers in Test Data		
			1	2	3
2-output PIT	2	160.7M	(80.6)	11.1	(52.1)
2-output PIT	1,2	160.7M	6.7	11.9	(52.3)
SOT	1,2,3	135.6M	4.6	11.2	24.0

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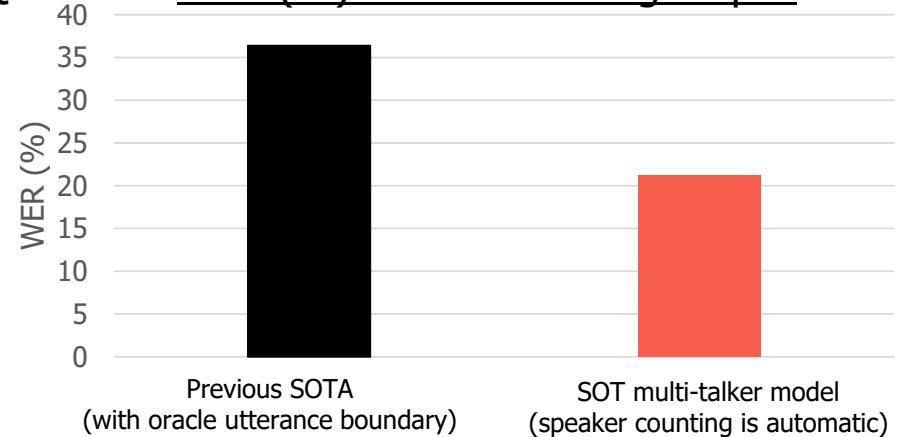
## Large-Scale Pre-Training of End-to-End Multi-Talker ASR for Meeting Transcription with Single Distant Microphone [Kanda+ 2021]

Does multi-talker model really work for real conversation?

→ Yes, SOT model achieved SOTA on AMI single distant microphone test set



WER (%) on AMI meeting corpus

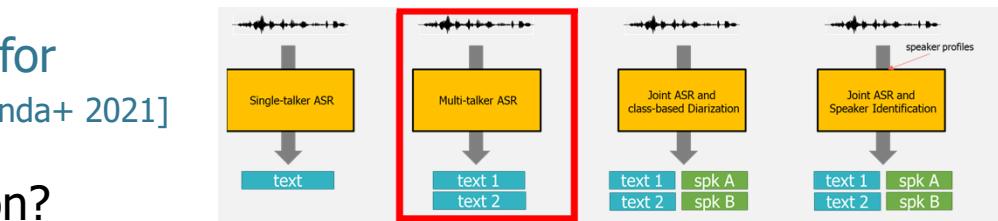
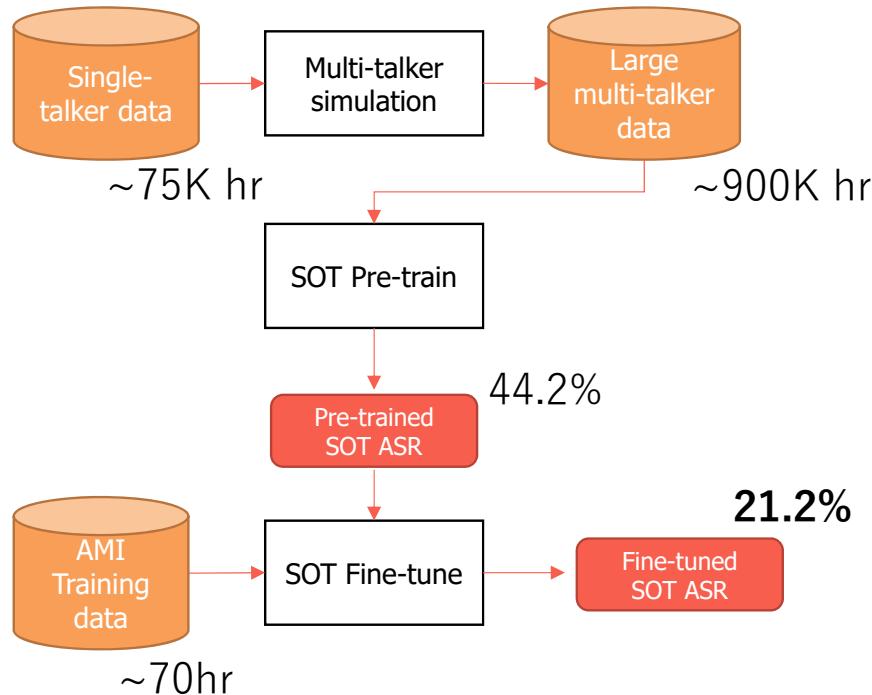


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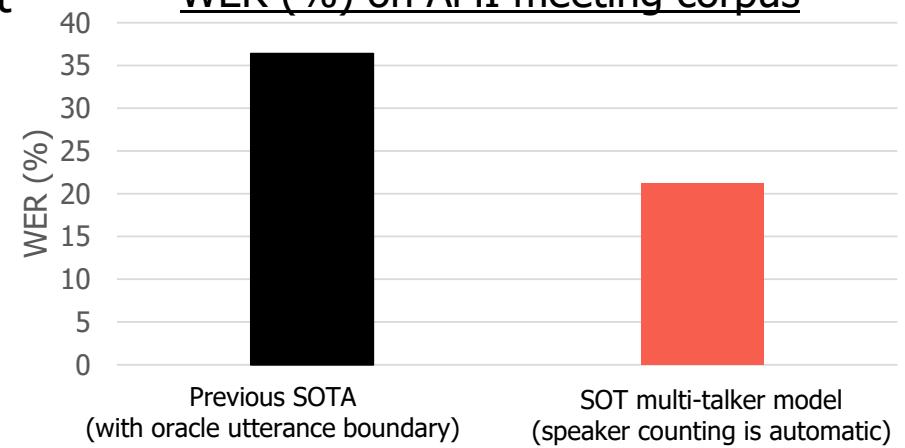
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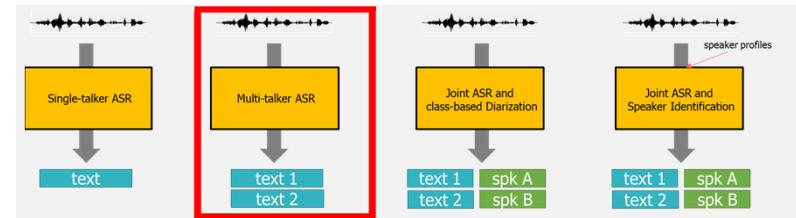
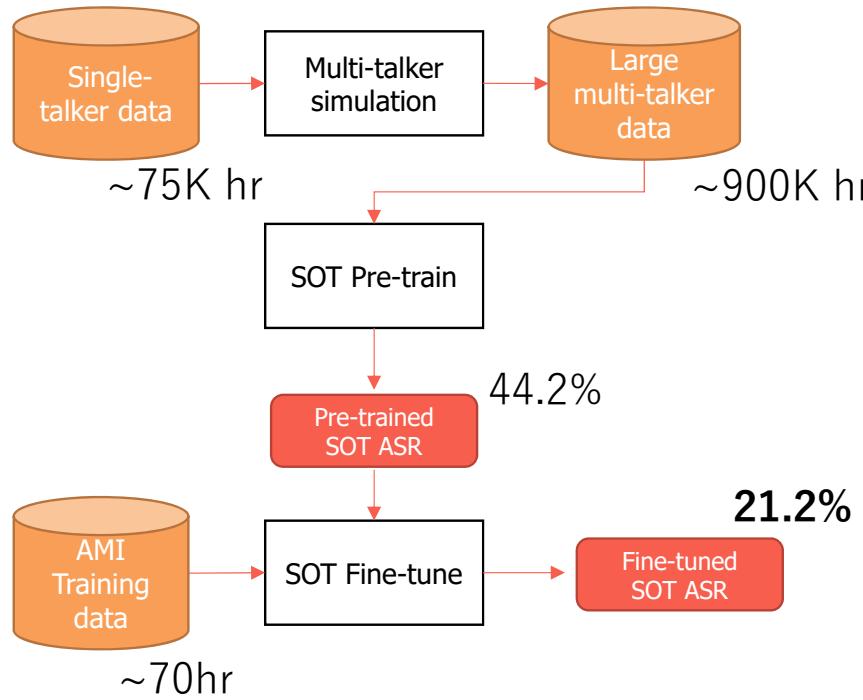


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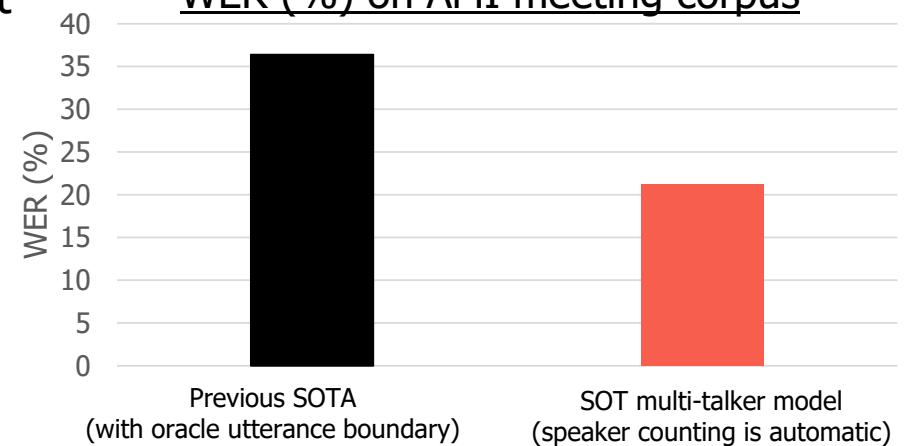
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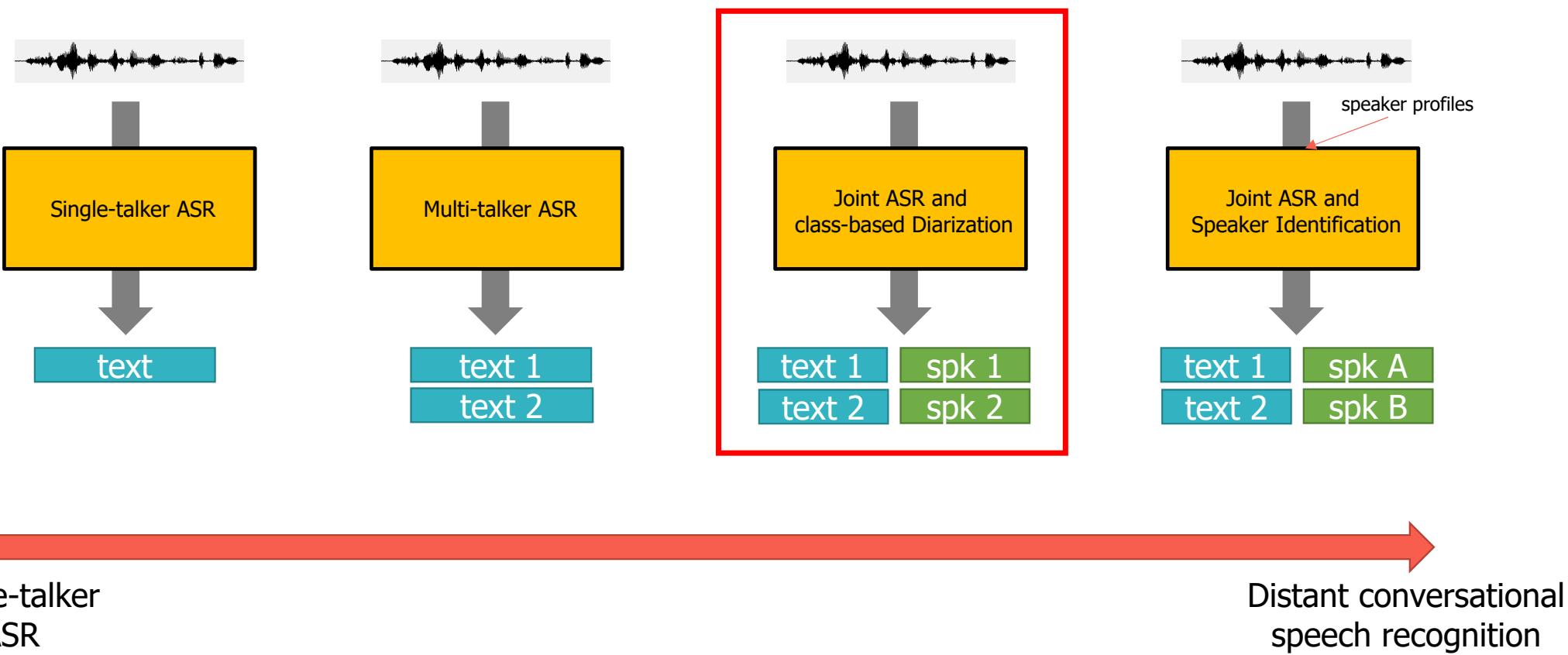
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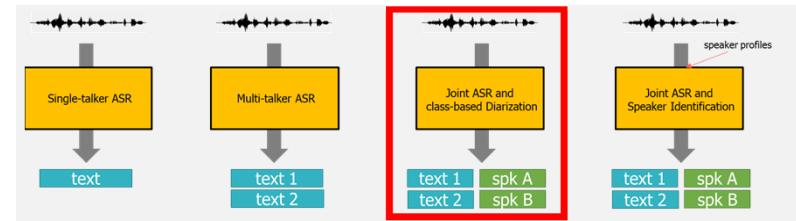
		Speaker Counting Accuracy					
		Estimated # of speakers					
		0	1	2	3	4	5
Actual # of speakers	1	0.2	<b>97.2</b>	2.5	0.1	0.0	0.0
	2	0.0	13.7	<b>80.5</b>	5.9	0.0	0.0
	3	0.0	2.4	32.6	<b>60.2</b>	4.8	0.0
	4	0.0	0.0	9.9	51.2	<b>38.9</b>	0.0

N. Kanda et al., Large-Scale Pre-Training of End-to-End Multi-Talker ASR for Meeting Transcription with Single Distant Microphone, Interspeech 2021 (to appear).

# What has been achieved for “ASR + $x$ ”



# Joint ASR and *Class-based* Diarization



how are you going <class1> i have a headache <class2>



Insert a ***special symbol that represents speaker*** at the end of each utterance.

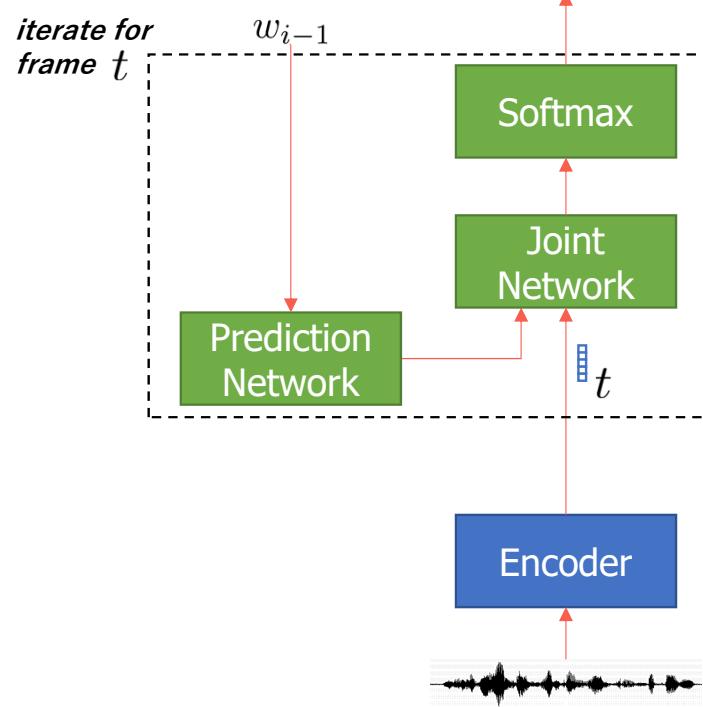
😊 Simple. No need to change ASR code.

😢 Cannot cope with a class that does not appear in training data.

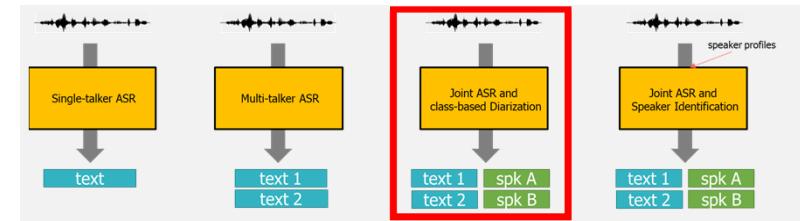
# Joint Speech Recognition and Speaker Diarization via Sequence Transduction [Shafey+ 2019]

**speaker role tag**

how are you going <doctor> i have a headache <patient>



Insert a ***speaker-role*** as one of output tokens.

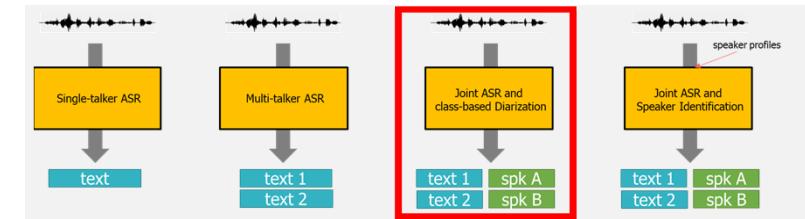
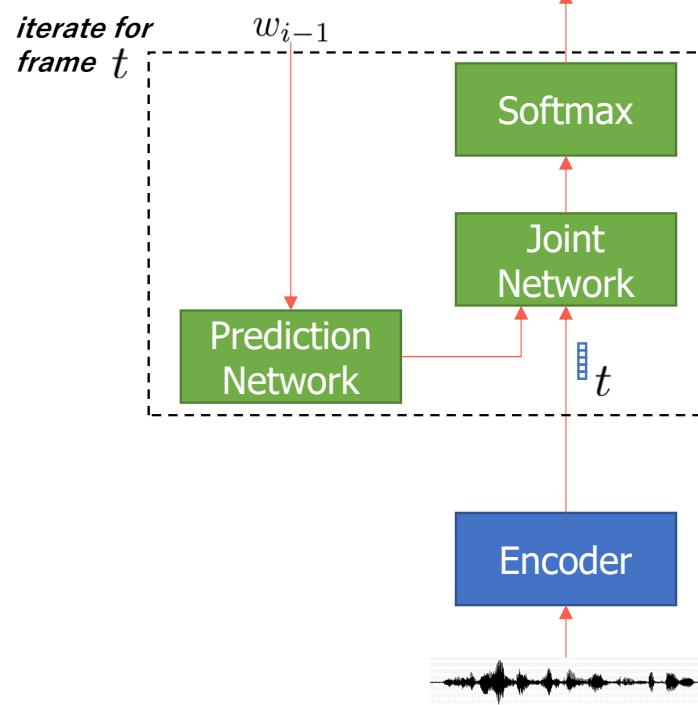


L. El Shafey et al., Joint Speech Recognition and Speaker Diarization via Sequence Transduction. *Proc. Interspeech 2019*, 2019, 396-400.

# Joint Speech Recognition and Speaker Diarization via Sequence Transduction [Shafey+ 2019]

**speaker role tag**

how are you going <doctor> i have a headache <patient>



Insert a **speaker-role** as one of output tokens.

😊 Work well for diarization with marginal degradation of ASR accuracy

Table 1: *Word Diarization Error Rate (WDER), Word Error Rate (WER) and its decomposition in Deletion / Insertion / Substitution errors (D/I/S) on the evaluation set.*

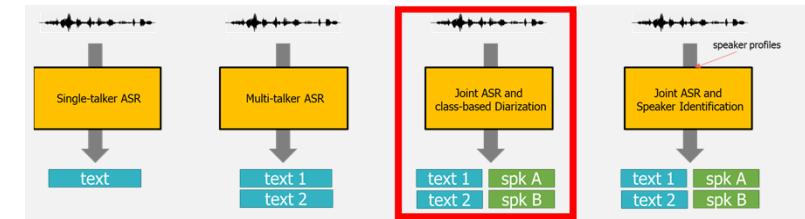
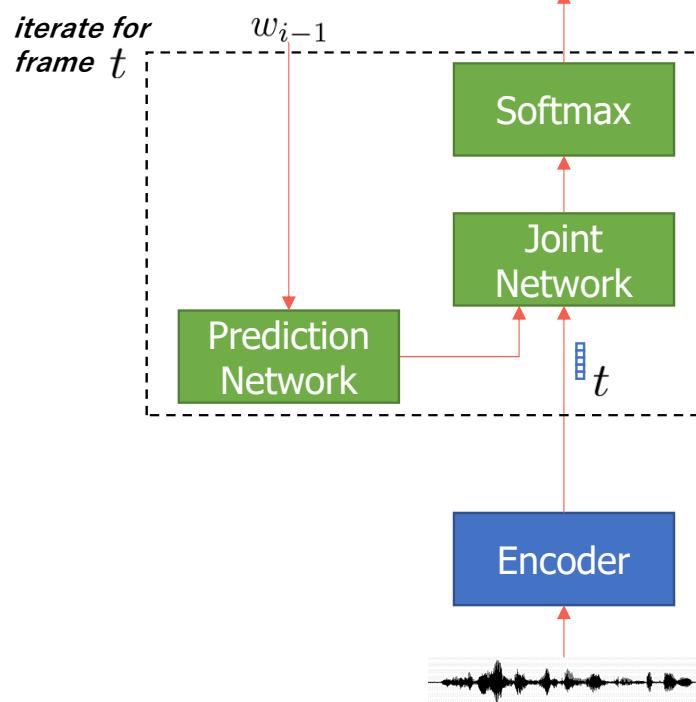
	Baseline	Joint ASR+SD
WDER	15.8%	<b>2.2%</b>
WER	<b>18.7%</b>	19.3%
D/I/S	7.2%/2.1%/9.4%	6.8%/2.8%/9.7%

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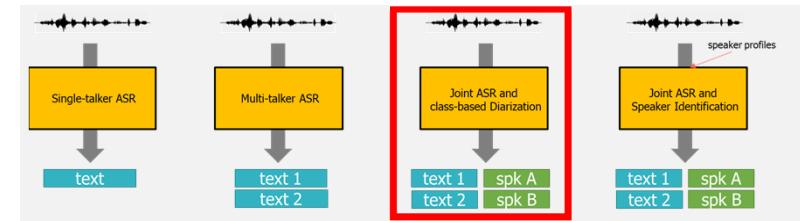
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😦 Work only for trained speaker-roles (such as doctor and patient)

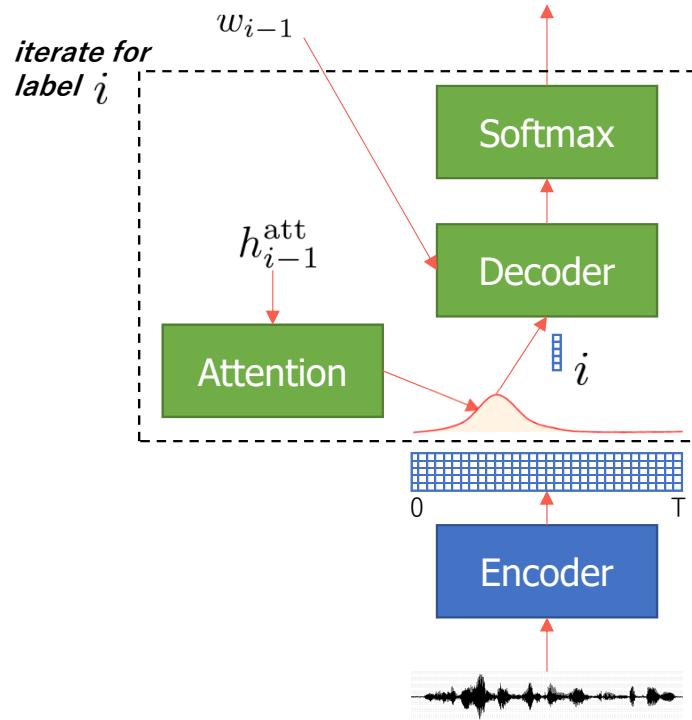
# Speech Recognition and Multi-Speaker Diarization of Long Conversations [Mao+ 2020]



## speaker ID

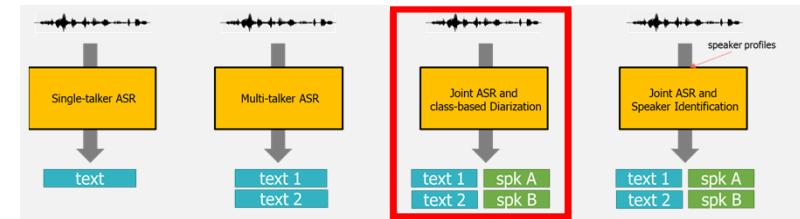
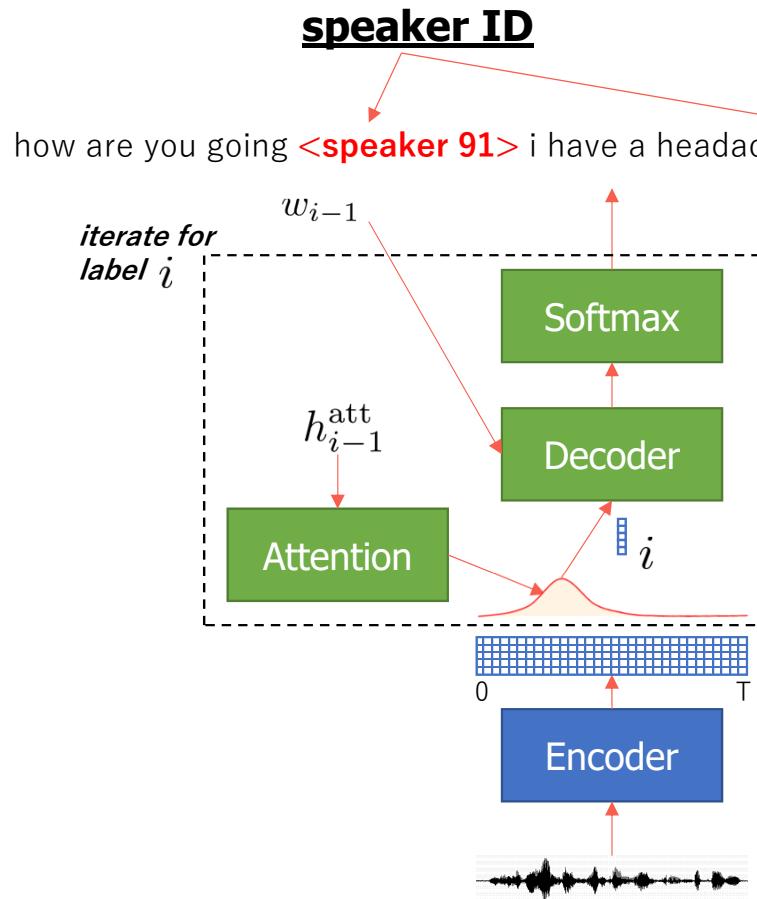
how are you going <speaker 91> i have a headache <speaker 15>

Insert a **speaker-id** as one of output tokens.



H. H. Mao et al., Speech Recognition and Multi-Speaker Diarization of Long Conversations. *Proc. Interspeech 2020*, 2020, 691-695.

# Speech Recognition and Multi-Speaker Diarization of Long Conversations [Mao+ 2020]



Insert a **speaker-id** as one of output tokens.

- 😊 Work well for a difficult case where utterance boundary is not given.
- 😢 Degradation when utterance boundary is given.

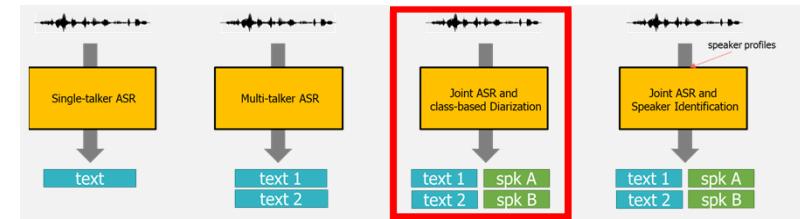
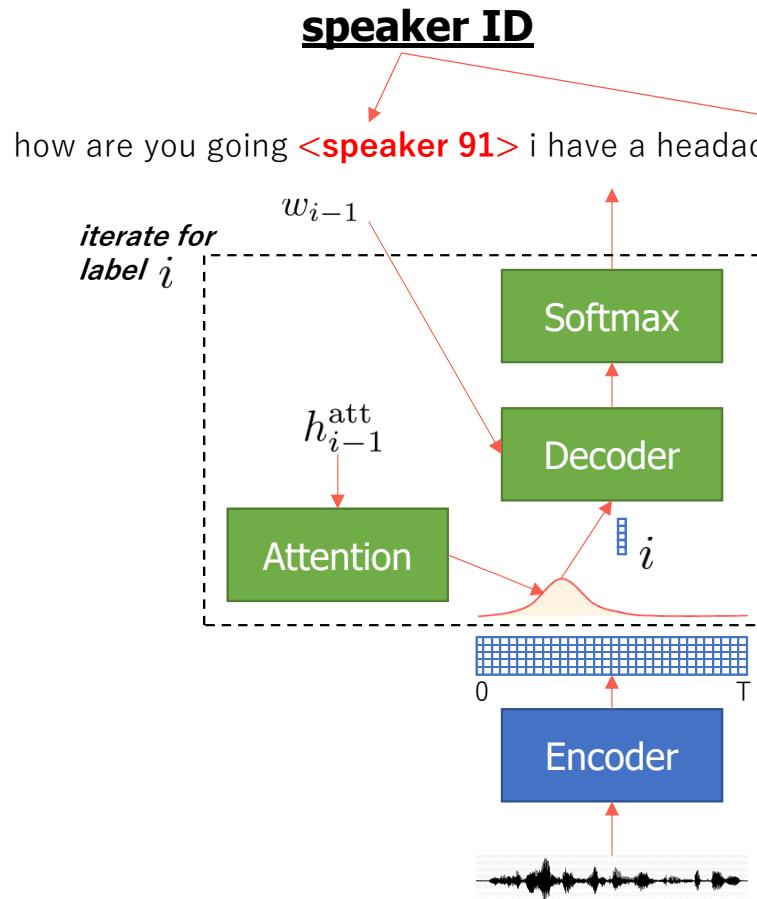
Result for "This American Life" radio program test set (from [Mao+ 2020])

	Utt. boundary is given		Utt. boundary is not given	
	WER	DER	WER	DER
Separate	24.3	15.4	58.3	91.3
Joint(*)	25.4	31.9	58.2	62.2

(\*) Mao et al. also proposed pre-training and data augmentation which showed a better result, but we extract the results without these techniques to highlight the pure difference between the separate model and joint model.

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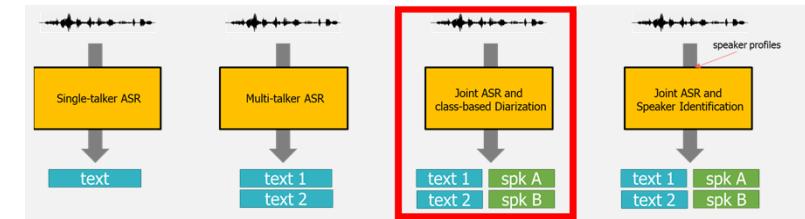
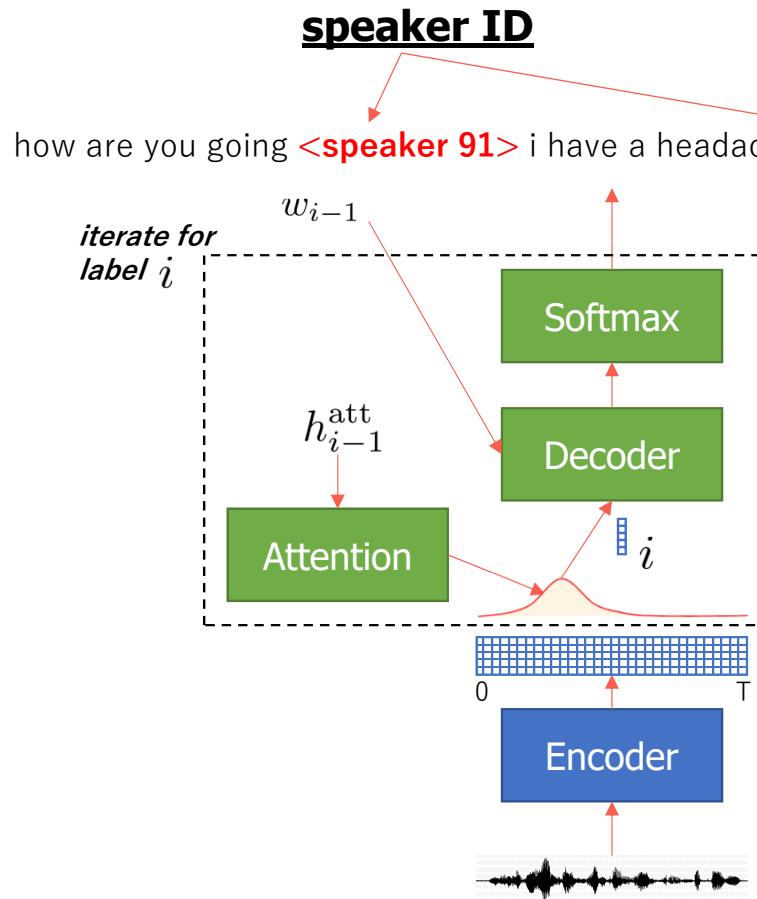
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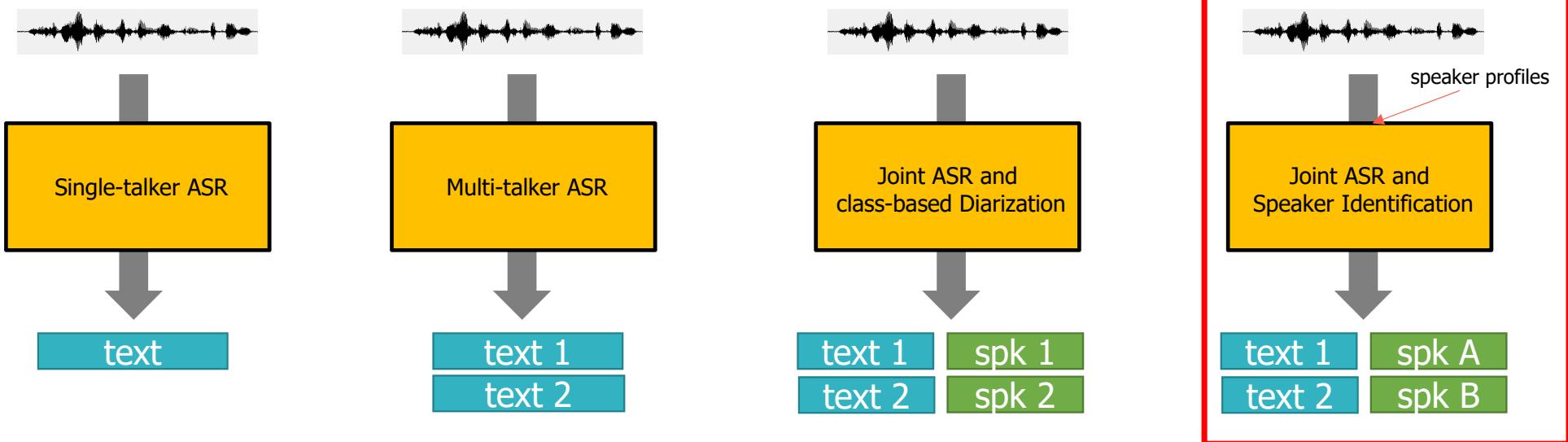
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# What has been achieved for “ASR + $x$ ”

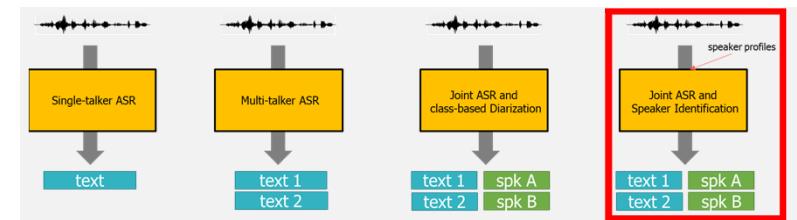


Single-talker  
ASR

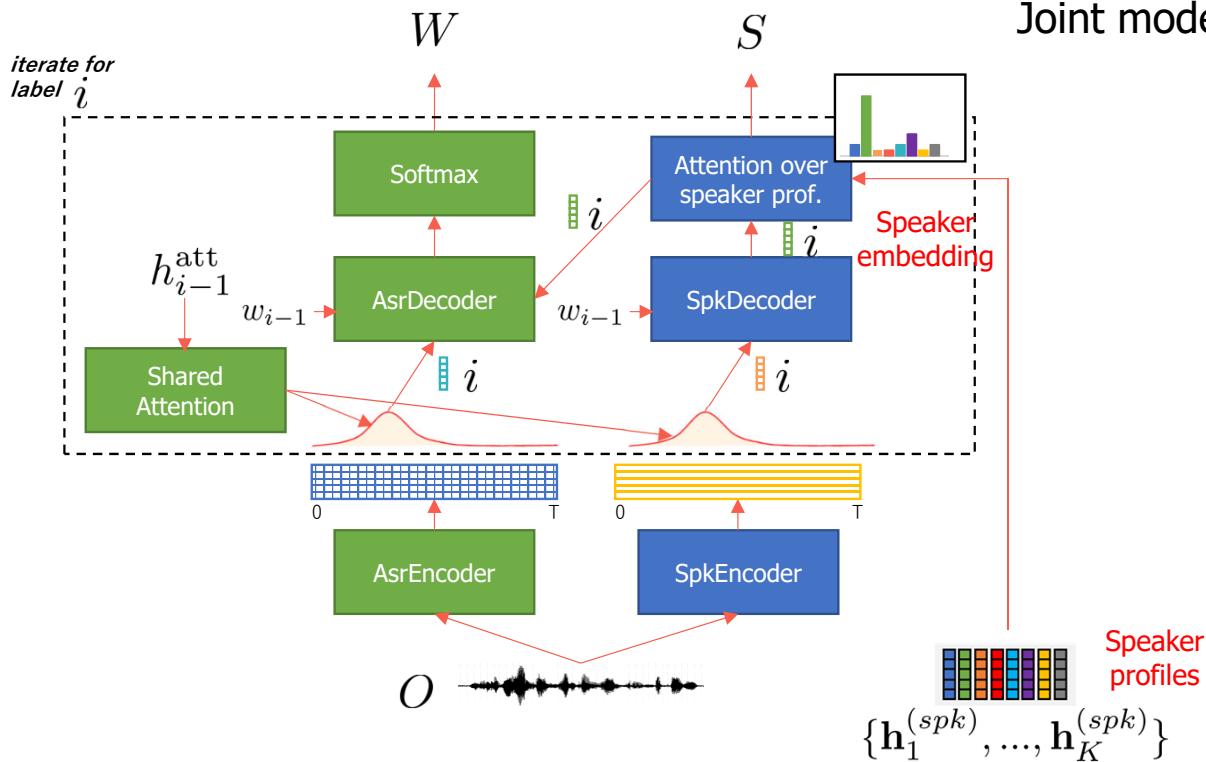
Distant conversational  
speech recognition

## Joint Speaker Counting, Speech Recognition, and Speaker Identification for Overlapped Speech of Any Number of Speakers [Kanda+ 2020]

$W$	how are you <sc>	i am fine thank you <eos>
$S$	2 2 2 2 5 5 5 5 5	



Joint model of ASR and ***speaker identification***



### End-to-End Speaker-Attributed ASR

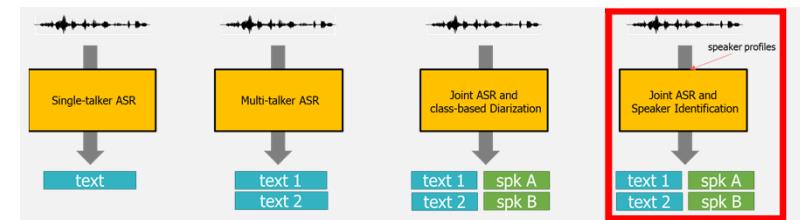
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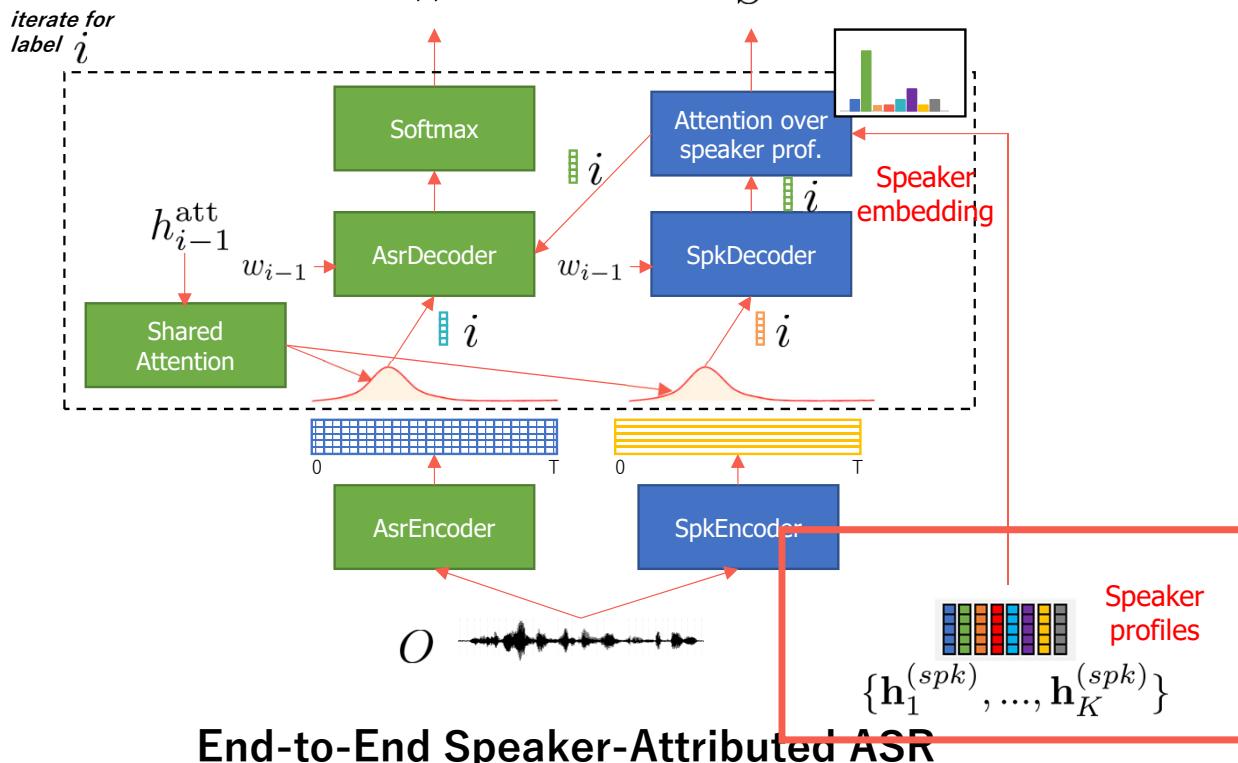
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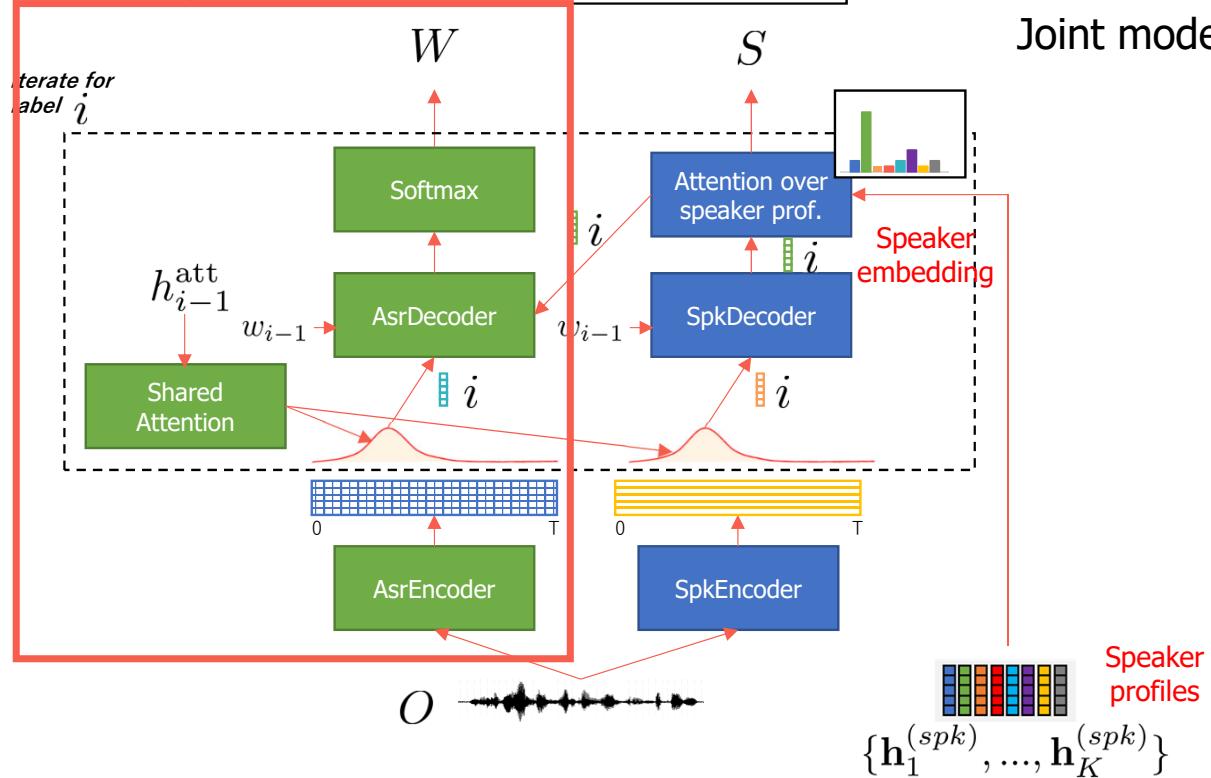
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### 3.3. A new trend toward jointly optimal systems: ASR + x

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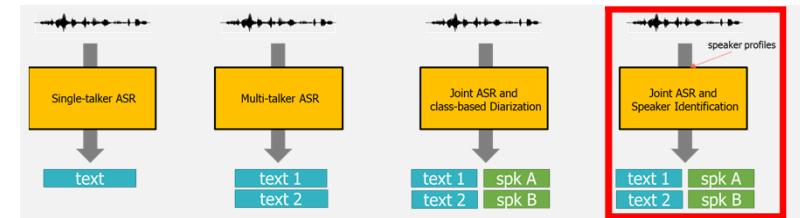


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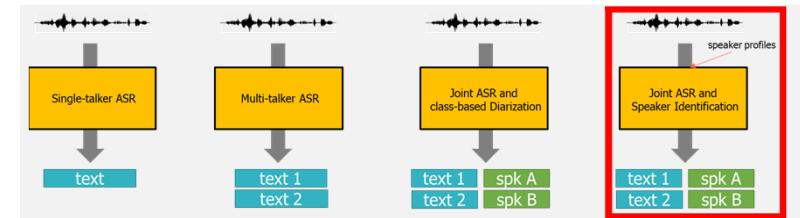
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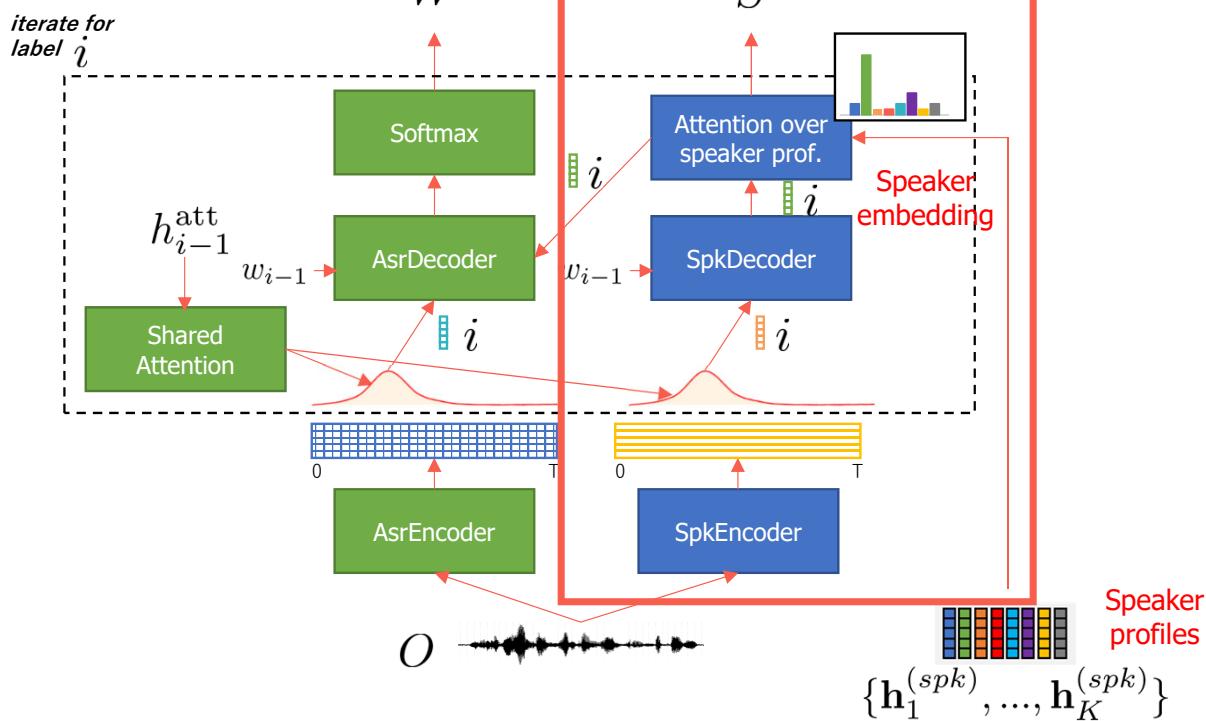
Joint model of ASR and **speaker identification**

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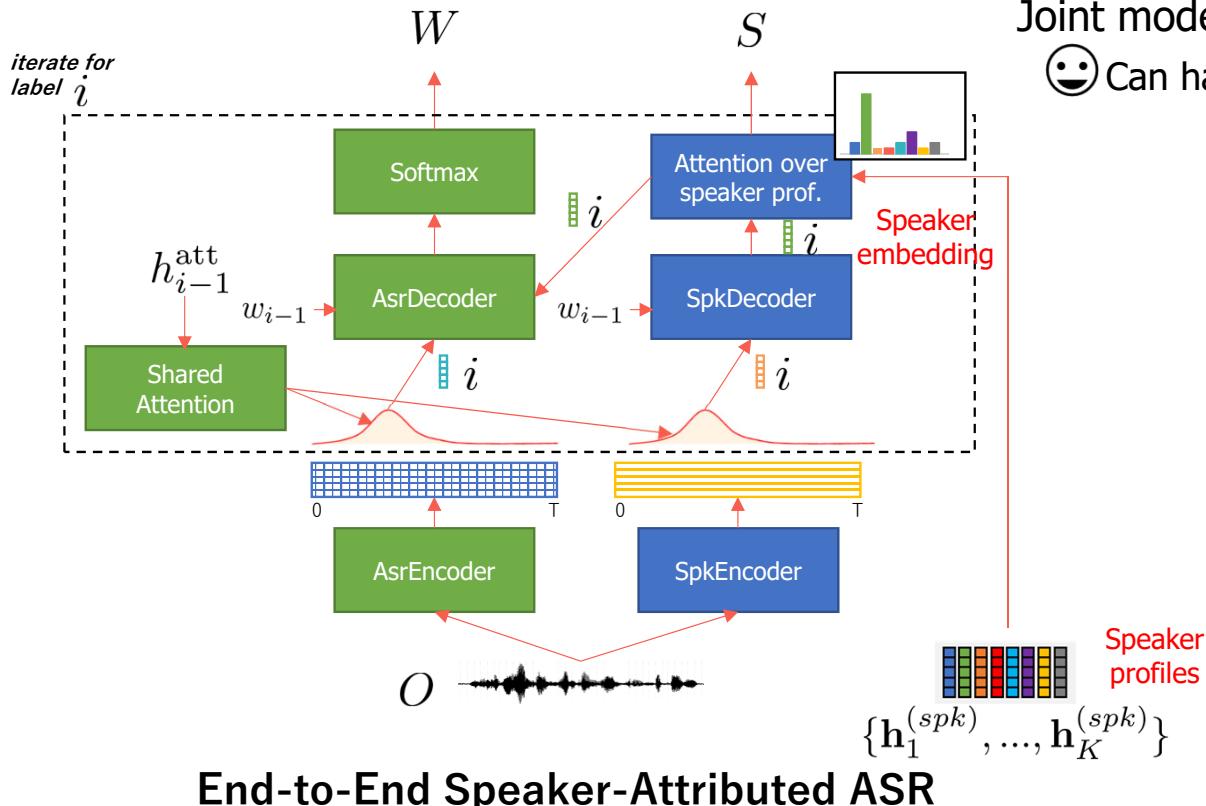
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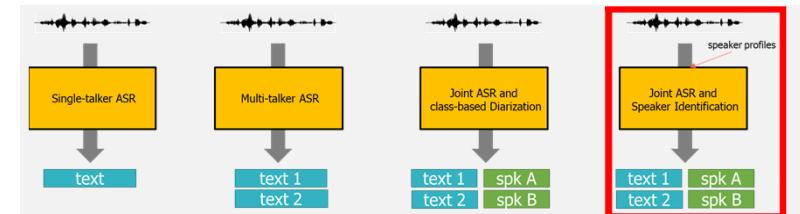
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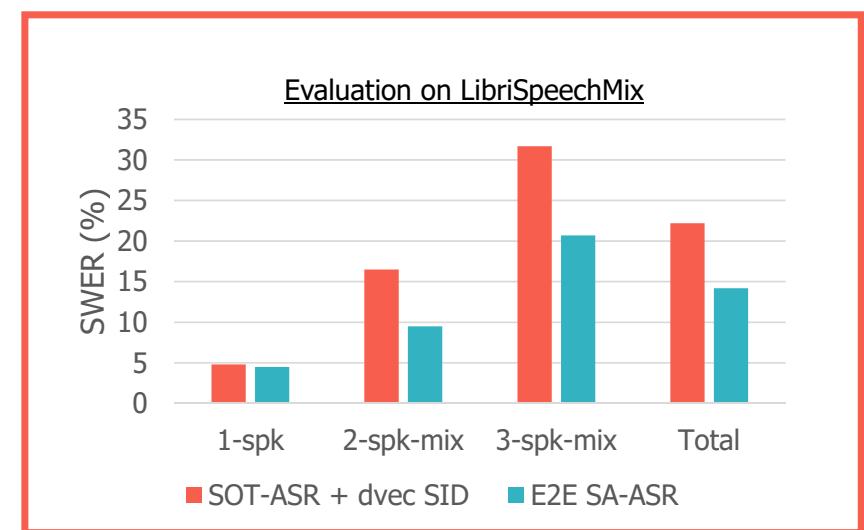


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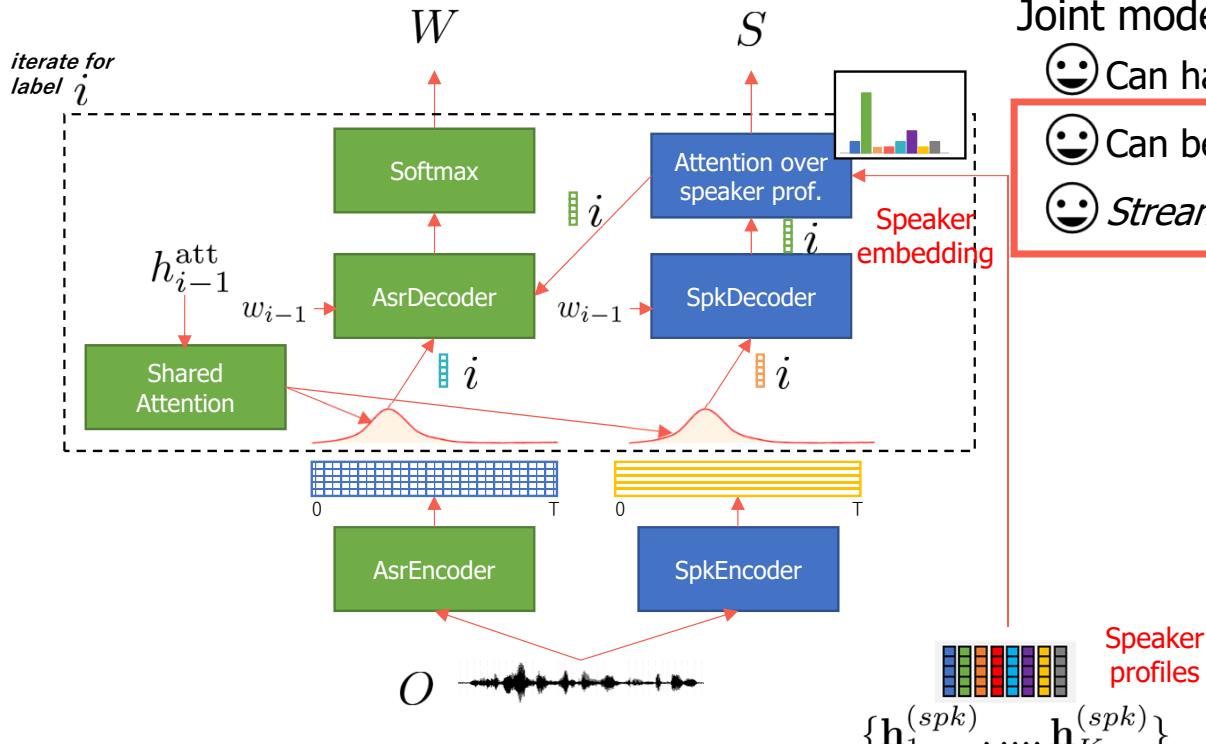
### Joint model of ASR and **speaker identification**

Can handle any number of speakers from overlapped speech



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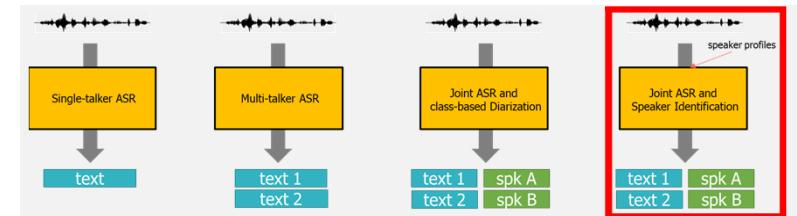


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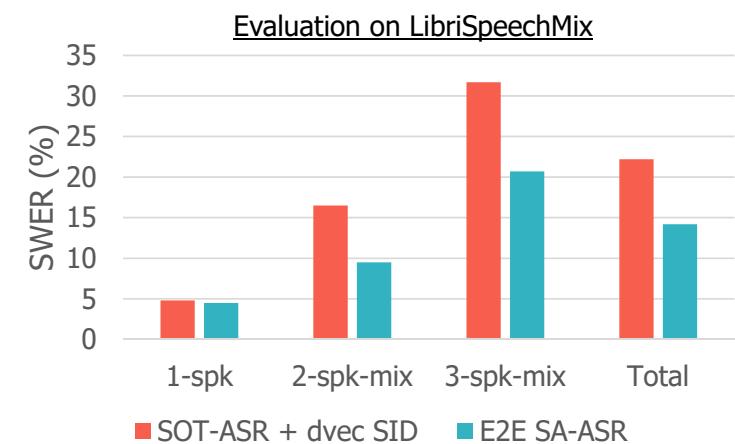
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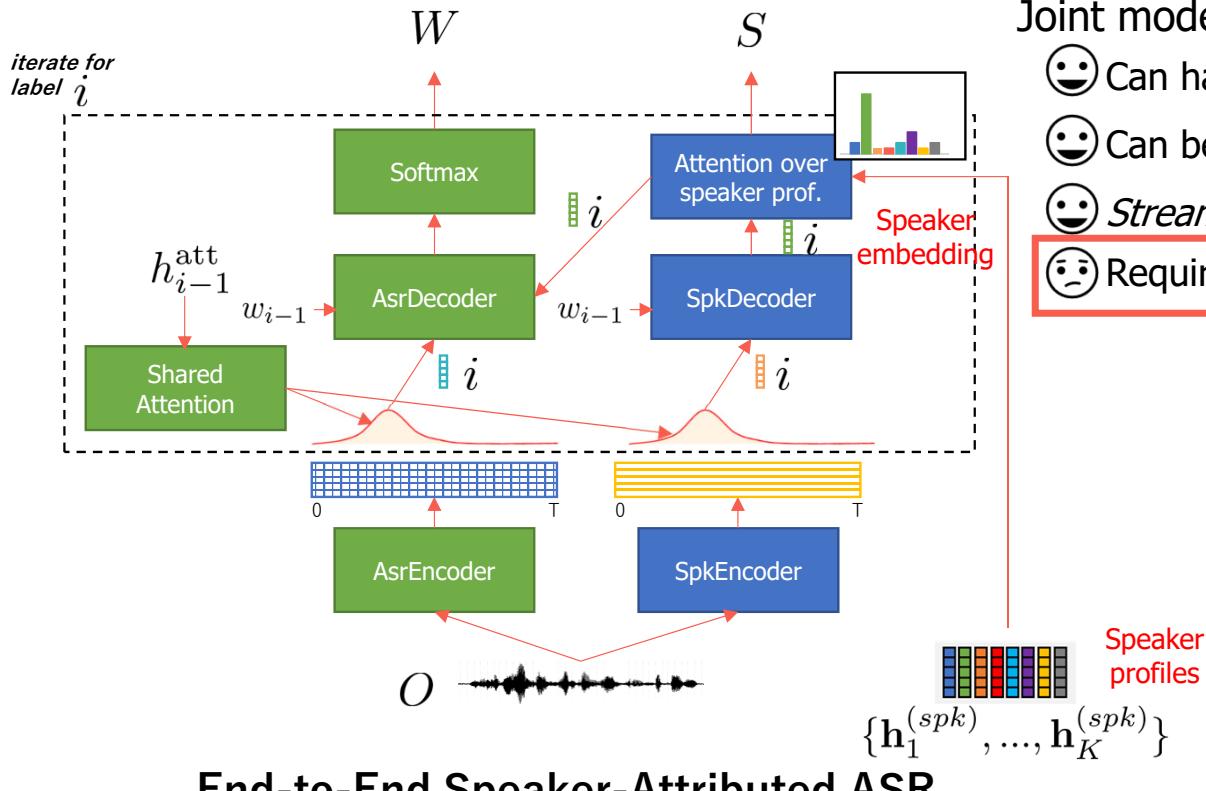
### Joint model of ASR and **speaker identification**

- 😊 Can handle any number of speakers from overlapped speech
- 😊 Can be optimized by minimizing expected SWER [Kanda+ 2021]
- 😊 *Streaming* model is also recently proposed [Lu+ 2021]



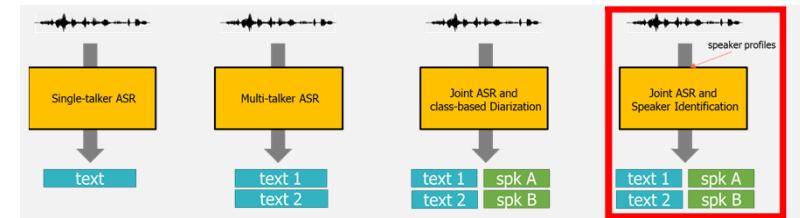
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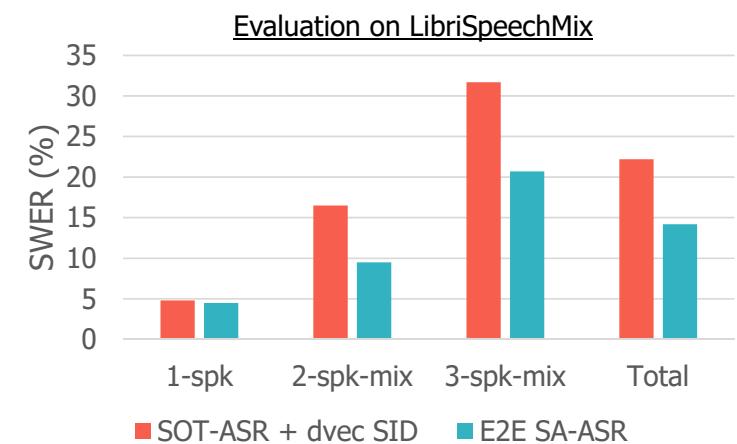
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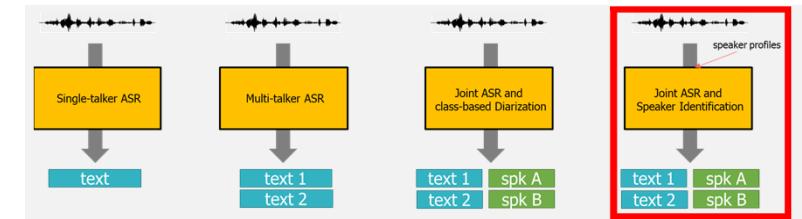


### Joint model of ASR and **speaker identification**

- Can handle any number of speakers from overlapped speech
- Can be optimized by minimizing expected SWER [Kanda+ 2021]
- Streaming* model is also recently proposed [Lu+ 2021]
- Requires speaker profiles as an auxiliary input (not pure diarization)



# E2E SA-ASR with speaker clustering [Kanda+ 2021]

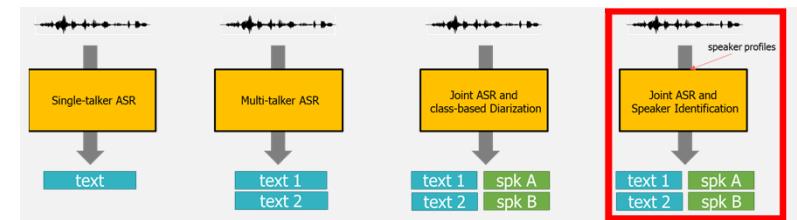


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# E2E SA-ASR with speaker clustering [Kanda+ 2021]



↓ (1) Segmentation with silence detection



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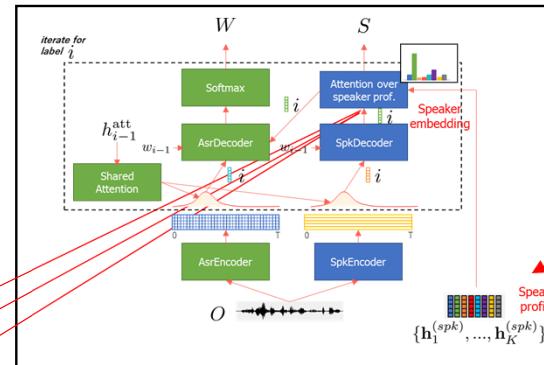
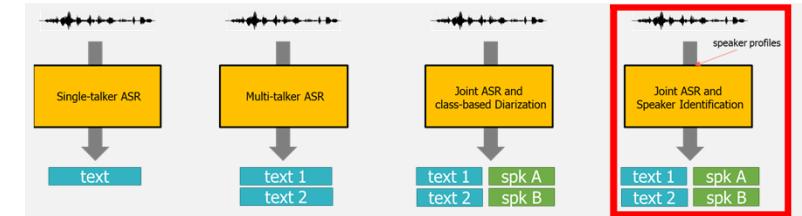
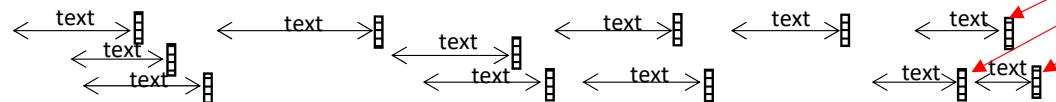
# E2E SA-ASR with speaker clustering [Kanda+ 2021]



↓ (1) Segmentation with silence detection



↓ (2) Apply E2E SA-ASR with *dummy* speaker profiles



Feed ***dummy*** speaker profiles extracted from training data

Remember averaged speaker embeddings for each utterance

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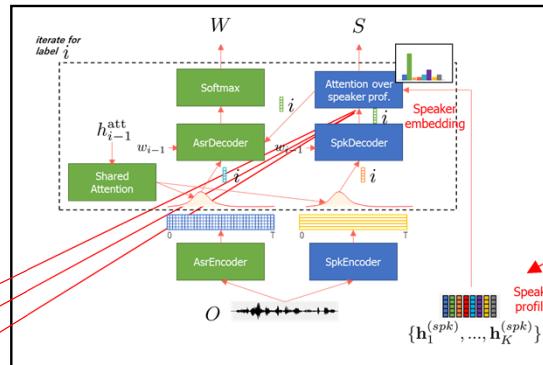
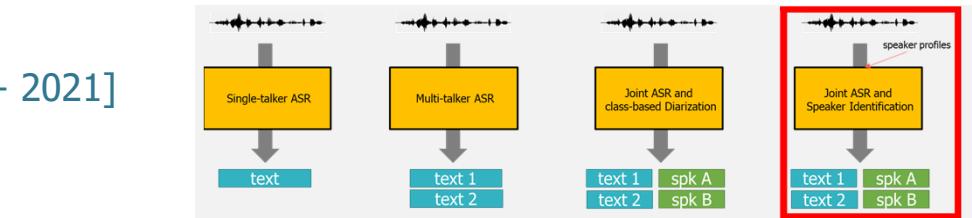
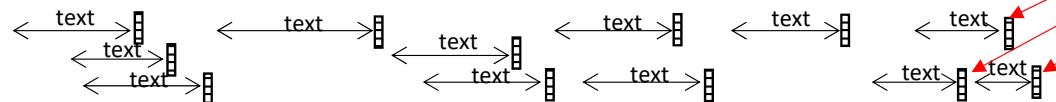
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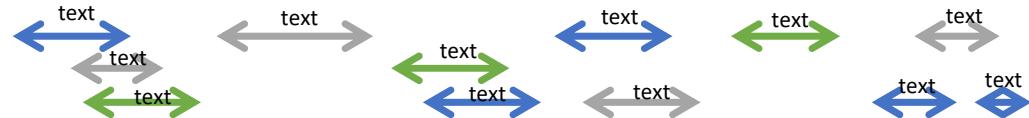
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(3) Speaker counting and clustering based on internal embeddings of E2E SA-ASR



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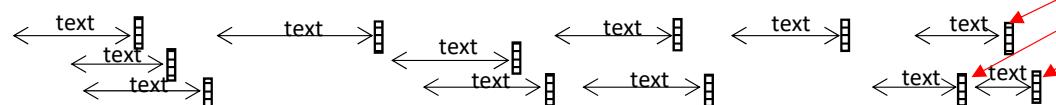
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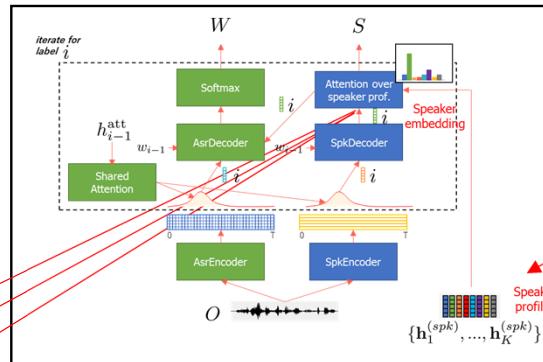
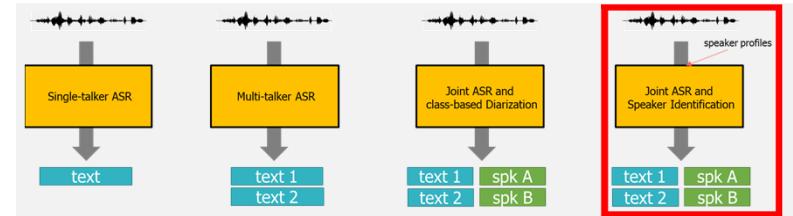
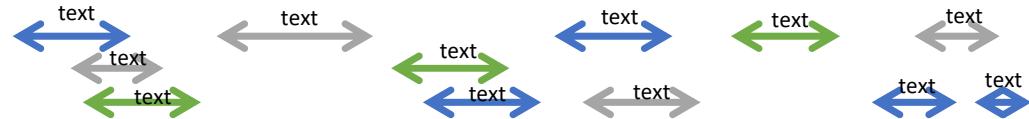
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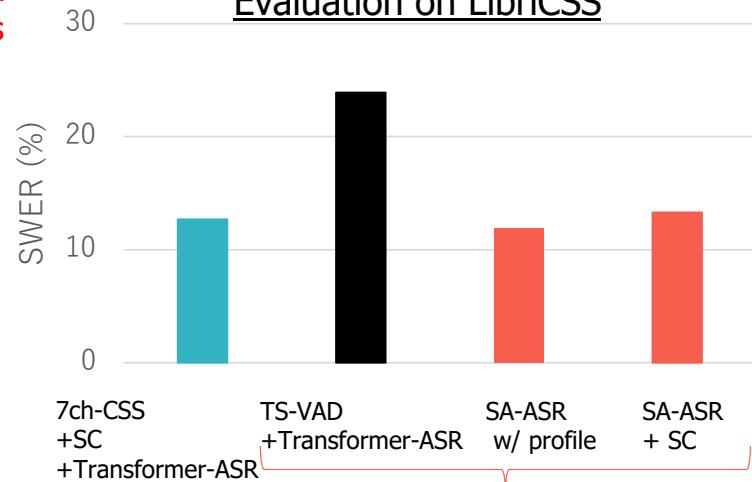
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Evaluation on LibriCSS



**Multi-ch. system**

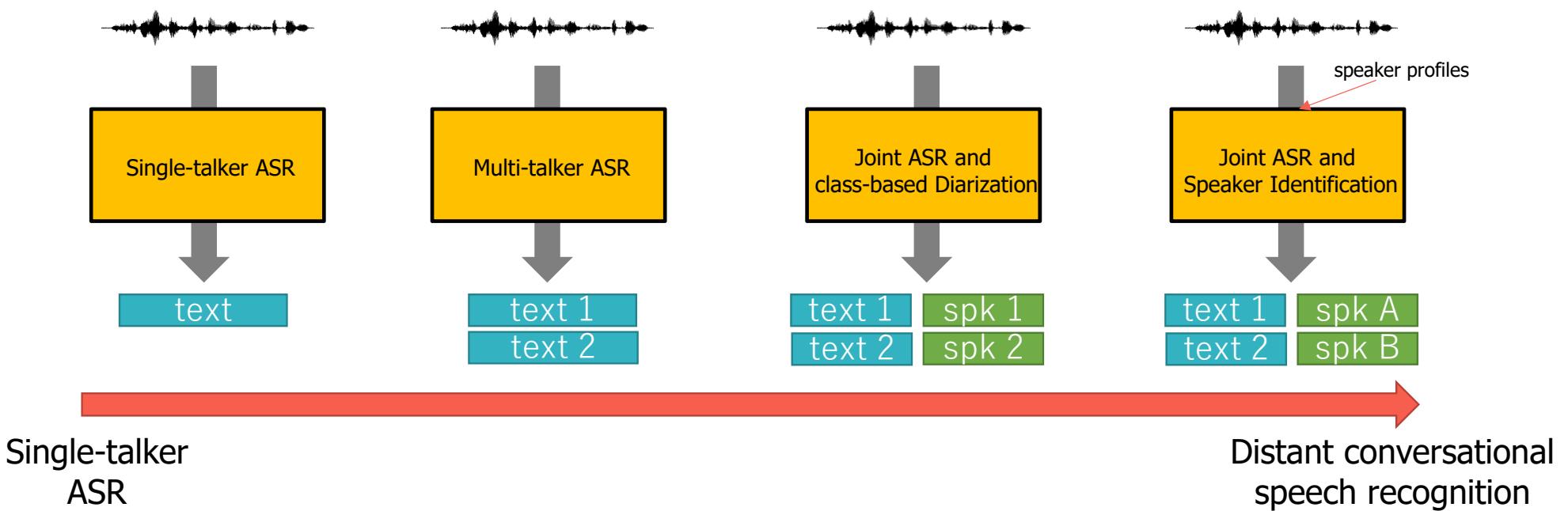
**Single-ch. system**

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# Short summary of ASR + $x$

Review the recent progress from the perspective of the extension of the E2E ASR.

- Multi-talker ASR
  - Two output-branch model
  - Serialized Output Training (SOT) for any number of speakers
- Joint ASR and class-based diarization
- Joint ASR and speaker identification (+ speaker clustering)

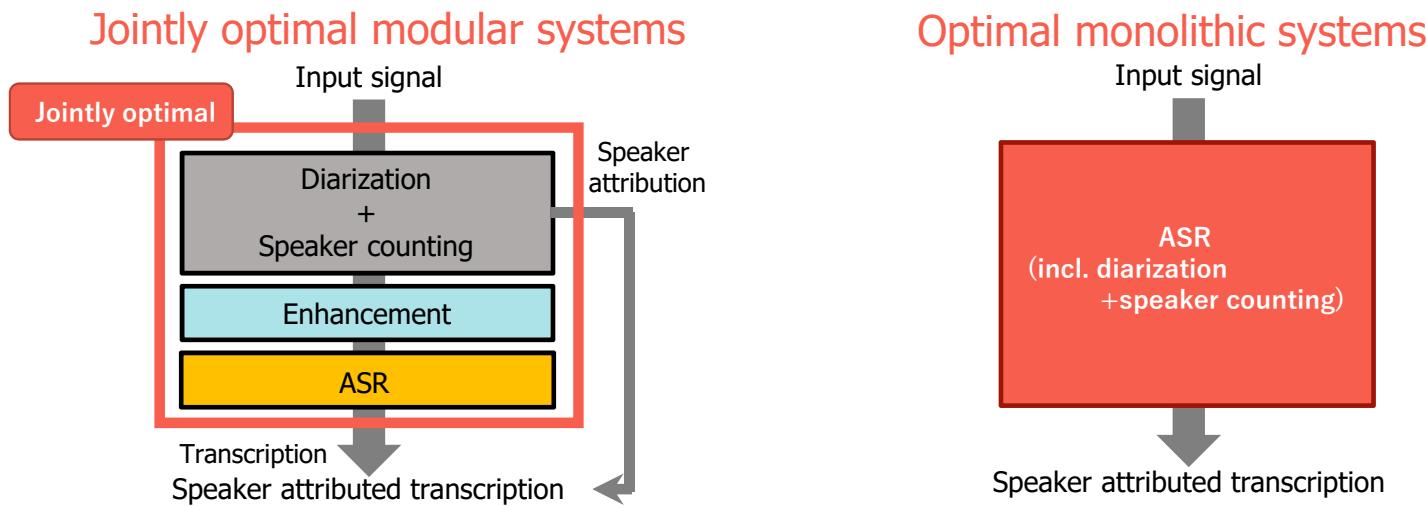


## 4. Discussion and summary

- 4.1. Strengths of each approach
- 4.2. Current challenges and possible future directions
- 4.3. Take home message + Starting points.

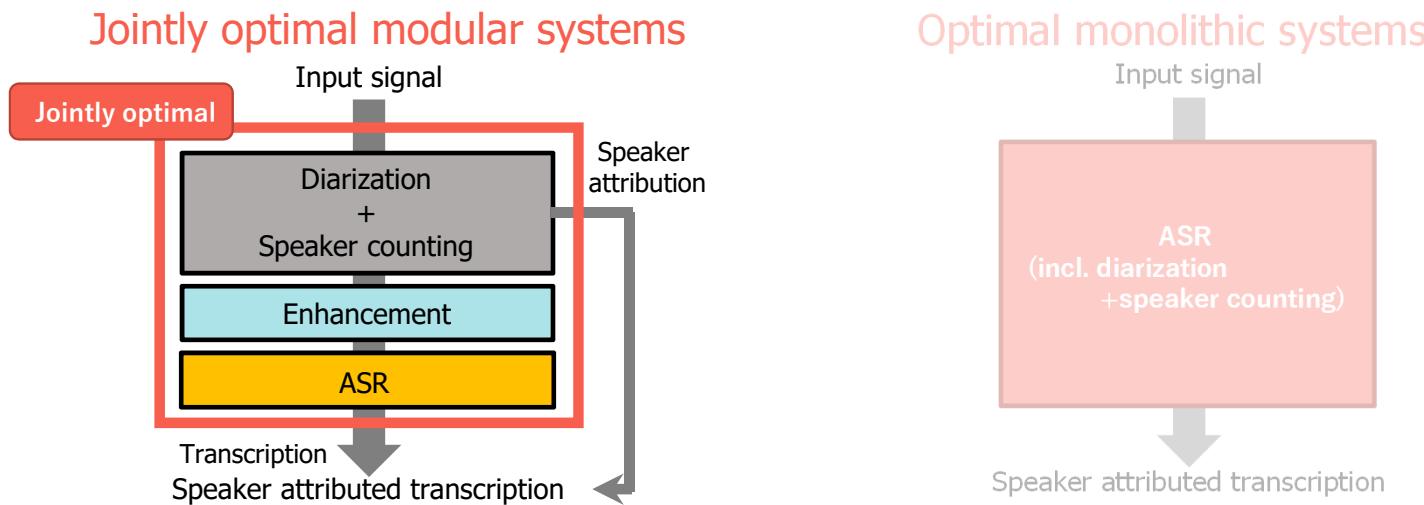
# Strength of each approach

The enhancement- and diarization-originated approaches tend to result in jointly optimal modular systems, while the ASR-originated approach tends to result in optimal monolithic systems.



# Strength of each approach

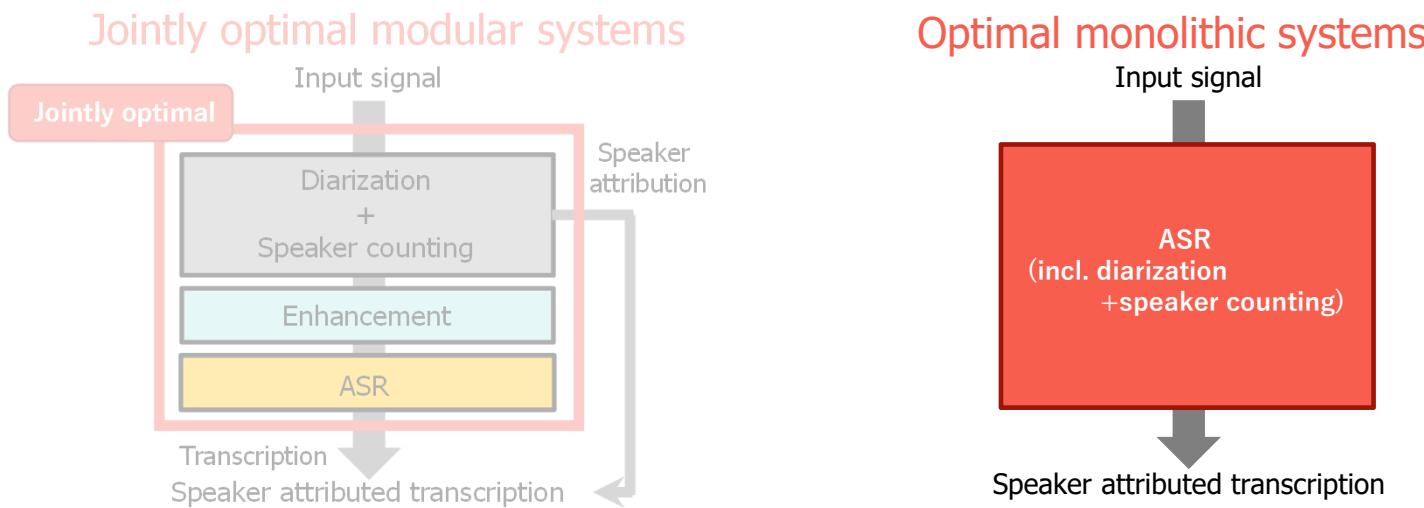
The enhancement- and diarization-originated approaches tend to result in jointly optimal modular systems, while the ASR-originated approach tends to result in optimal monolithic systems.



- 😊 Interpretable. → Easier to debug and to find a source of performance degradation.
- 😊 Can incorporate well-studied/already-developed modules, e.g., single-speaker streaming ASR.
- 😊 Flexible. Can significantly alter the functionality of the total system by modifying a module. E.g., multi-talker ASR to single-talker ASR. Command recognition to utterance recognition.

# Strength of each approach

The enhancement- and diarization-originated approaches tend to result in jointly optimal modular systems, while the ASR-originated approach tends to result in optimal monolithic systems.



- 😊 Can keep the whole system simpler and smaller (i.e., computationally efficient, less parameters).
- 😊 Dedicated system. Can be completely tuned to certain purpose, e.g., distant conversational ASR.
- 😊 Possible to prepare reference labels i.e., transcription, even from real recordings.  
→ Possibility to adapt the system to unseen/real test data.

## **4. Discussion and summary**

**4.1. Strengths of each approach**

**4.2. Current challenges and possible future directions**

**4.3. Take home message + Starting points.**

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# Discussion on challenges and fundamental difficulties

- Currently available solution/systems have not achieved close-talking-microphone ASR performance, i.e., ASR performance obtained without effects of overlaps, noise, reverb, Lombard effect, etc.
- Why? Because we still have many issues such as:
  - **Unsolved subtasks:** Some issues in each subtask are not yet fully solved, e.g.,
    - Diarization (and separation): An arbitrary (potentially large) number of speakers [Horiguchi+, 2020]
    - ASR: Long-form audio [Chang+, ICASSP2021].
    - Enhancement: Noise and reverb still degrade conversational ASR performance a lot [Kanda+, 2021].
  - **Optimality of the total system:** Not yet achieved fully optimal distant conversational ASR system.
  - **Data issue:** Not yet clear what kind of and how much data is necessary
    - E.g., Simply mixing many different corpora can boost conversational ASR performance [Chan+, 2021]
  - **Labels and eval metrics:** Preparation of “correct” transcription is difficult when the input data is spontaneous speech, including fillers, self-edits etc [Saon+, 2017].  
Whether WER is really appropriate or not, can be also debatable.

S. Horiguchi et al. End-to-End Speaker Diarization for an Unknown Number of Speakers with Encoder-Decoder Based Attractors, Interspeech, 2020  
X. Chang et al., Hypothesis Stitcher for End-to-End Speaker-attributed ASR on Long-form Multi-talker Recordings, ICASSP, 2021  
N. Kanda et al. Large-Scale Pre-Training of End-to-End Multi-Talker ASR for Meeting Transcription with Single Distant Microphone, Interspeech, 2021 (to appear)  
W. Chan et al. SpeechStew: Simply Mix All Available Speech Recognition Data to Train One Large Neural Network, arXiv, 2021,  
G. Saon et al., English Conversational Telephone Speech Recognition by Humans and Machines, Interspeech, 2017

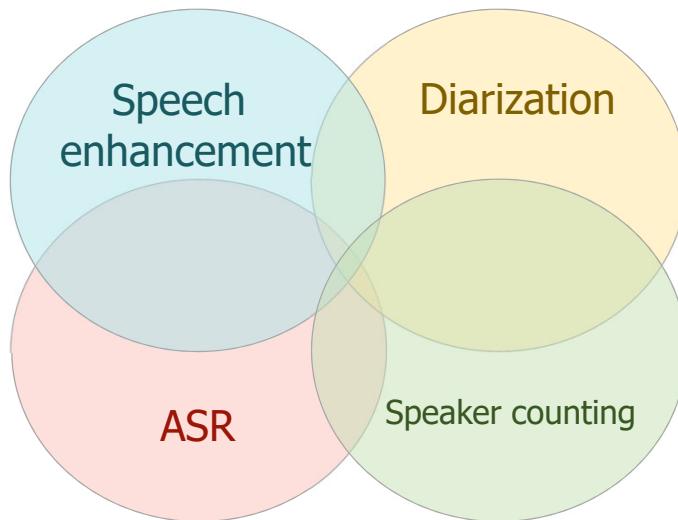
## 4. Discussion and summary

- 4.1. Strengths of each approach
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- 4.3. Take home message + Starting points

# Take-home message

- The distant conversational ASR and analysis is a **core of next-generation speech interfaces**.
- **Many academic institutes and industry** are actively working on this task.
- Lots of **interesting** research problems, increasingly **active** area, many OSS and data.

Let's work together on this topic and solve it!



# Starting points (useful software, database, etc) (1/2)

- Open source software
  - Enhancement
    - Asteroid (Neural source separation) [1]
  - Diarization
    - pyannote.audio (Neural building blocks for speaker diarization) [2]
  - ASR
    - Kaldi (DNN-hybrid ASR) [3]
      - Upgraded CHiME6 baseline system containing diarization, multichannel enh.
      - LibriCSS baseline system containing separation, diarization, multichannel enh.
    - ESPnet (E2E ASR, neural speech enhancement, neural diarization) [4]
    - SpeechBrain (E2E ASR, Speaker diarization/recognition/identification, neural speech enhancement, beamforming) [5]
    - RETURNN (E2E ASR) [6]

[1] <https://github.com/asteroid-team/asteroid> [2] <https://github.com/pyannote/pyannote-audio>  
[3] <https://github.com/kaldi-asr/kaldi> [4] <https://github.com/espnet/espnet>  
[5] <https://github.com/speechbrain/speechbrain> [6] <https://github.com/rwth-i6/returnn>

## Starting points (useful software, database, etc) (2/2)

- Corpora containing real conversational speech data
  - CALLHOME (telephone speech, single-microphone setup)
  - AMI (3 to 5 (mostly, 4) speaker meetings, noise & reverb, multi-microphone setup)
  - CHiME-5 data (4-speaker very spontaneous meetings, noise & reverb, multiple microphone-array setup)
- AISHELL-4 (4 to 8 speaker meetings, noise & reverb, multi-microphone setup)
- CHiME 7 (Coming soon)

## Possible future directions

- For **improving the performance** of the distant conversational ASR and analysis
  - Addressing the issues we described
  - Incorporating other modality features, e.g., language information, visual information, location information
- For **enriching the transcription** to estimate “to whom”, “how”, and “why”
  - Integration with natural language processing technologies, e.g., summarization, translation, spoken language understanding
  - Integration with emotion recognition, language recognition, etc.

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# This slides

The latest version of this tutorial slides is available at



[https://github.com/ICASSP2021-tutorial9/Distant\\_conversational\\_ASR\\_and\\_analysis/blob/main/ICASSP2021\\_tutorial\\_T9\\_slides.pdf](https://github.com/ICASSP2021-tutorial9/Distant_conversational_ASR_and_analysis/blob/main/ICASSP2021_tutorial_T9_slides.pdf)