Unsupervised speaker adaptation for DNN-based speech synthesis using input codes

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Background（1/2）

Statistical parametric speech synthesis

– Remarkable progress thanks to DNNs

Flexible and controllable speech synthesis

– Speaker, gender, and age codes: “input codes” [Luong+; 16]
  - Multi-speaker modeling
  - Flexible manipulation
  - Speaker adaptation

– Speaker adaptation using back-propagation [Luong+; 16]
  - Speech and text data of a target speaker are required

Speaker adaptation using only speech data

Background（2/2）

Speaker adaptation using a speaker-similarity vector

- Speaker-similarity vector : new speaker code
  - Text-independent ASV models are used
    - Posterior probabilities are concatenated to form the code
  - The code represents acoustic similarity to speakers
- Inputting the estimated code of a target speaker can generate the target speaker's voice

Speaker adaptation using low-quality speech data

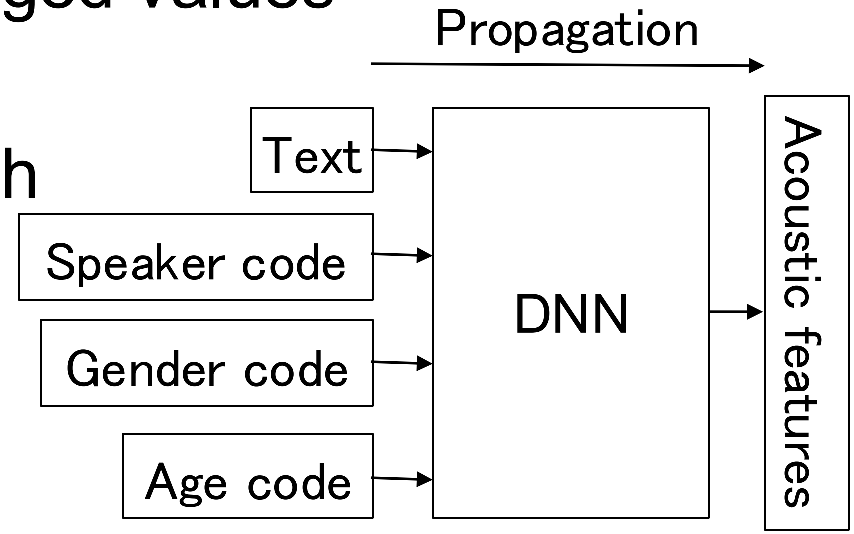
- A robust ASV model is required
- Model training using artificially created low-quality speech
  - Alleviating recording condition mismatch between training and adaptation data

Multi-speaker modelling using input codes

Multi-speaker modelling using input codes

- Input codes: simple additional inputs that differentiate ID, gender and age of speakers
- Generate multiple speakers' voices from a single DNN

- Also good as an initial model for speaker adaptation
  - Input codes that use averaged values
    → Average voice
- Allow us to manipulate speech
  - e.g. flip the gender code
- Morphing
  - Change the code each frame



Adaptation using input code: ‘*phantom code*’

Estimate speaker code using adaptation data

– Estimation based on back-propagation [Bridle et al.; 90]

– Estimate the speaker code only, fix the other codes and other DNN parameters

Update procedures of the speaker codes

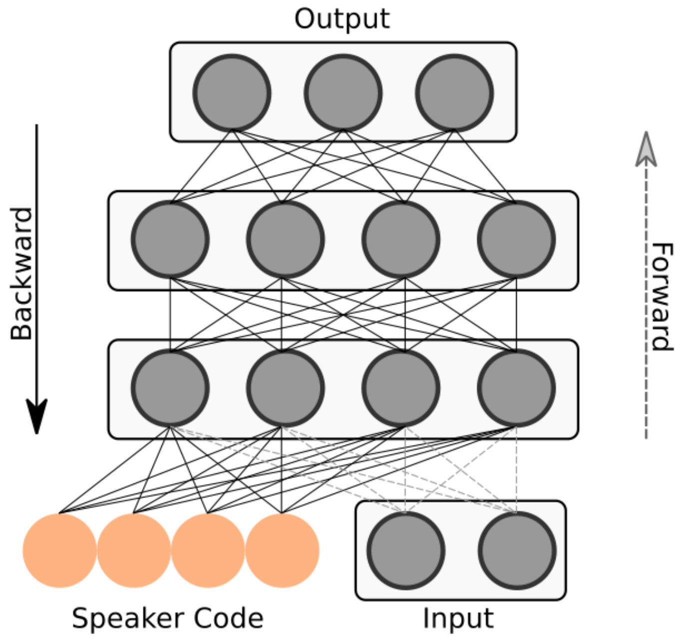
– Initialize the codes with the average

– Update the codes

- Fixed maximum number of epoch:
- Fixed learning rate

– Choose codes that has minimum errors

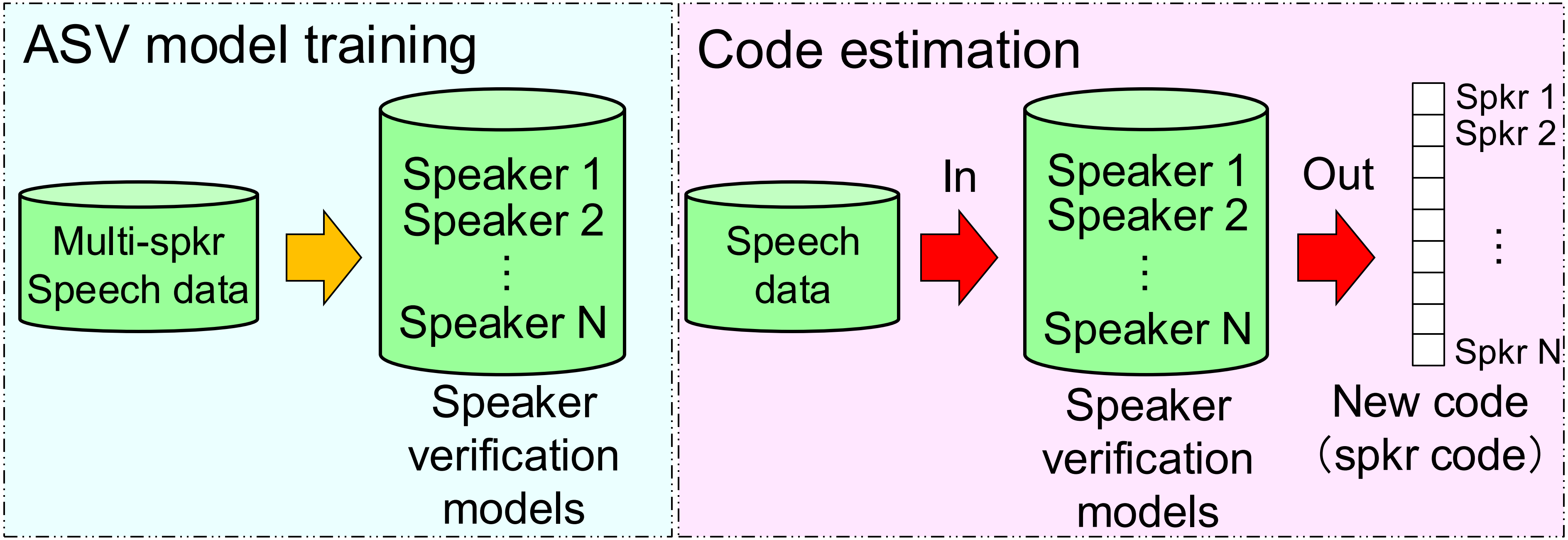
– Simple!!



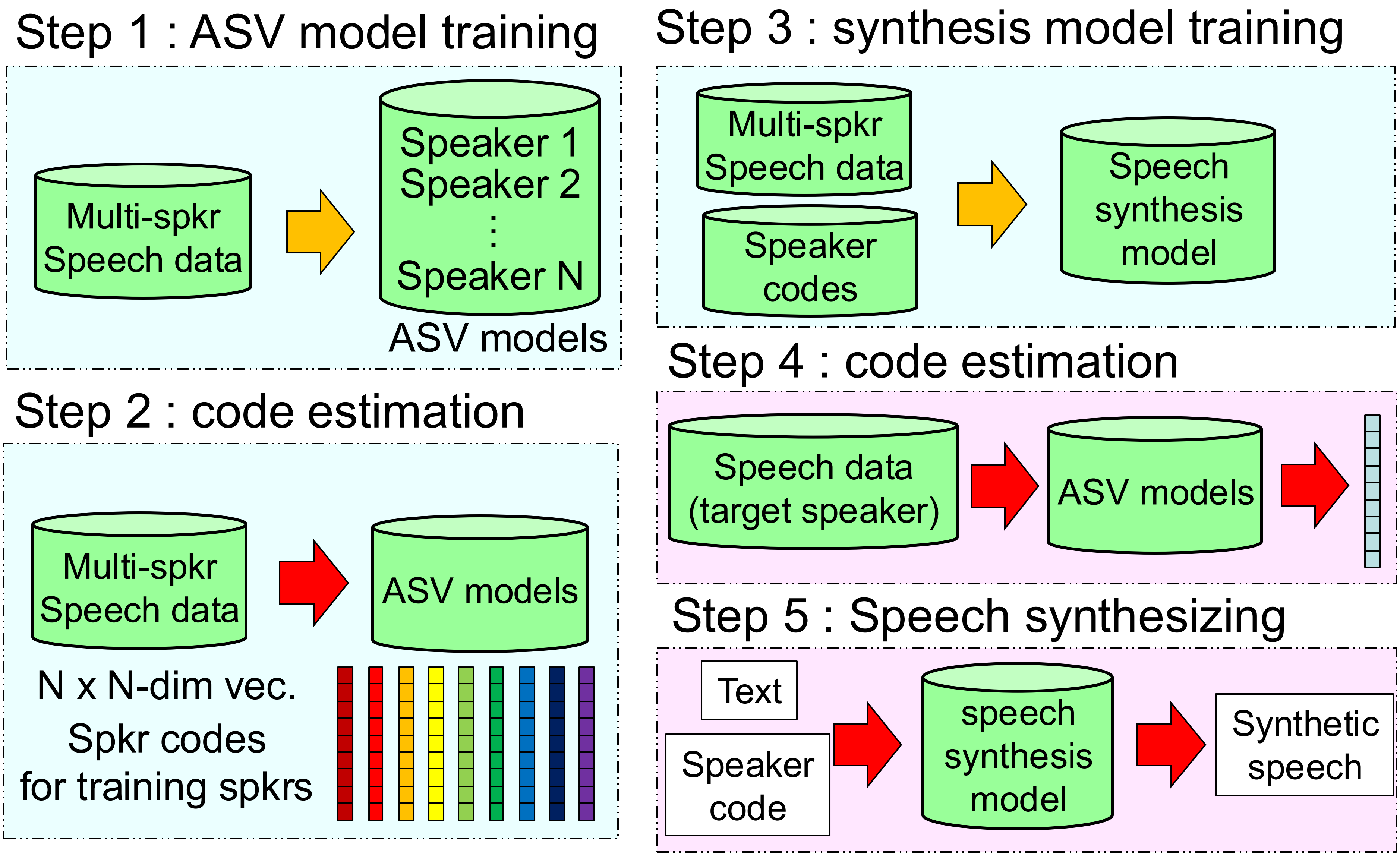
New speaker code : Speaker-similarity vector

Acoustic similarity to each of training speakers

- Replacing 1-hot vectors with speaker-similarity vectors
- Acoustic similarity is represented by posterior probability
  - Using text-independent ASV models
  - GMM-UBM or i-vector/PLDA is used



Flow of the proposed technique



Synthetic speech will vary if the speaker-similarity vector change

Training robust ASV model

Adaptation data is usually low quality

Training ASV models using low-quality data

- Alleviating recording condition mismatch
- Adding noise and reverberation to training speech data
  - An office room and a meeting room
  - Various SNRs
- Demand : Noise database [Thiemann+; 13]
- The Ace Challenge : Reverberation database [Hadad+; 14]
- Adaptation date is also artificially created
  - Office :
  - Meeting :

High SNR ⟷ Low SNR

Experimental conditions（1/3）

Multi-speaker				Adaptation			
Age	Male	Female	Total	Age	Male	Female	Total
10-20	8	8	16	10-20	0	2	2
21-30	8	8	16	21-30	2	2	4
31-40	8	8	16	31-40	2	2	4
41-50	8	8	16	41-50	1	2	3
51-60	8	8	16	51-60	2	2	4
61-70	8	8	16	61-70	2	2	4
71-	8	8	16	71-	0	2	2
Total	56	56	**112**	Total	9	14	**23**

– High-quality Japanese speech database
– Training: 112 speakers, 100 utterances per speaker, total of 11,170 utterances
– Adaptation: 23 speakers, 100 utterances per speaker
– Test: 10 different sentences per speaker

Experimental conditions（2/3）

Acoustic features（speaker verification）

- 19-dim MFCC+$\Delta + \Delta^2$ （MFCC）
- 19-dim WORLD mel-cepstrum+$\Delta + \Delta^2$ （MGC）
- 20-dim F0 features +$\Delta + \Delta^2$ （F0）
  - DCT is applied to F0 of prev., current and next 32 frames

Acoustic features（speech synthesis）

- 59-dim WORLD mel-cepstrum+$\Delta + \Delta^2$
- Voiced/Unvoiced parameter, Log F0+$\Delta + \Delta^2$
- 25-dim band apriodicity +$\Delta + \Delta^2$

Input（speech synthesis）

- 386-dim linguistic features, oracle phone duration
- Speaker, gender and age codes

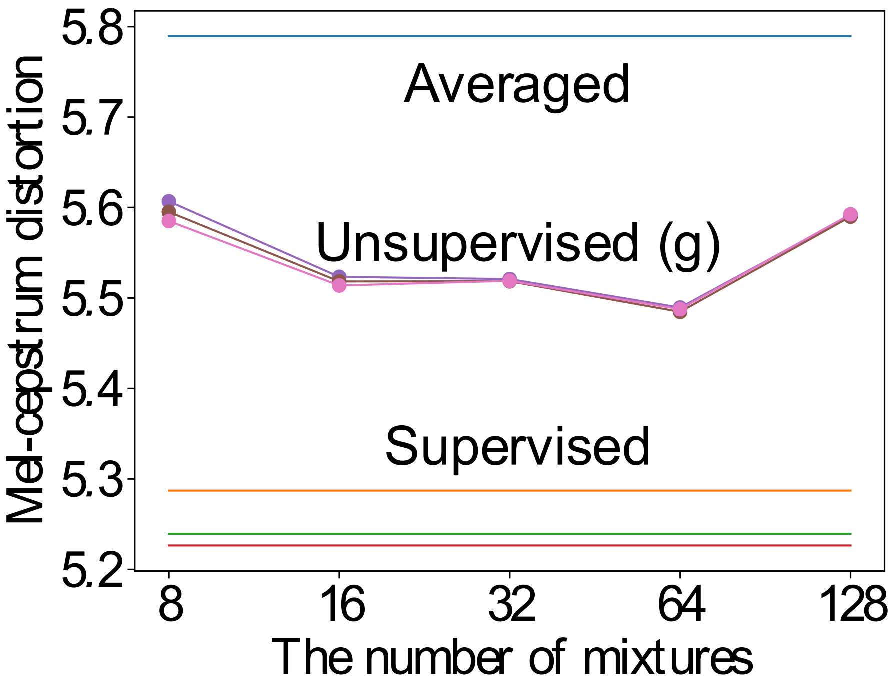
Experimental conditions（3/3）

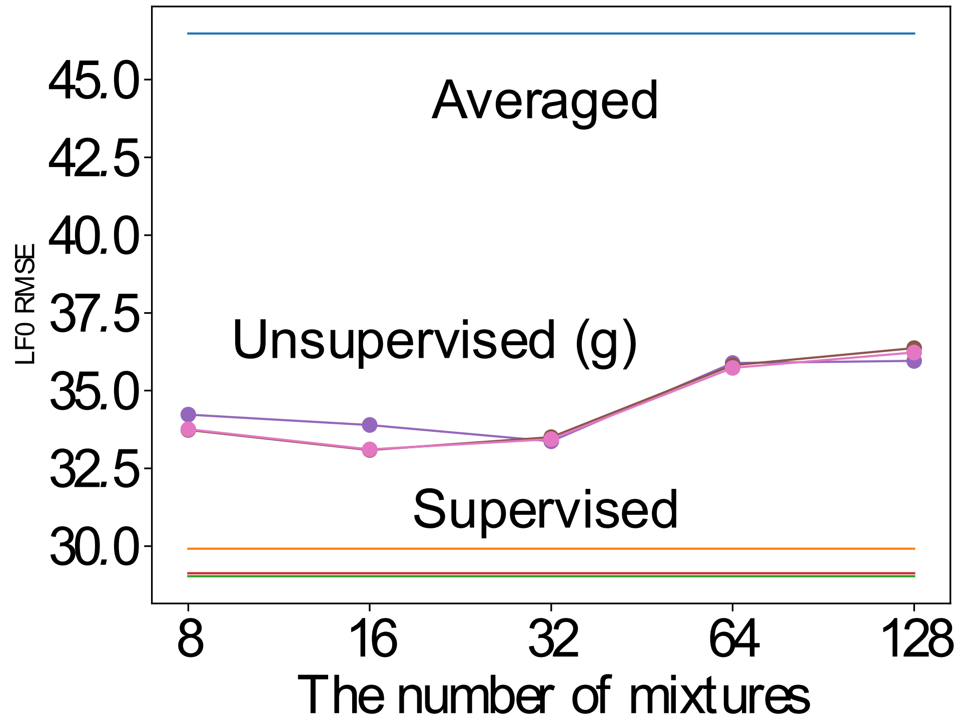
Systems	Multi-speaker model	Adaptation
Averaged	One-hot vector	-
Supervised	One-hot vector	Vector estimated by BP
Unsupervised (g)	Speaker-similarity vec. estimated from GMM-UBM	
Unsupervised (i)	Speaker-similarity vec. estimated form i-vector/PLDA	

– SIDEKIT was used to train ASV models
  - GMM-UBM（#mixtures：8, 16, 32, 64, 128）
  - i-vector/PLDA
    - –The number of mixtures for extracting i-vector：64
    - –i-vector dimension：400

– Speech synthesis model
  - Feed forward DNN（Hidden layers : 5, units : 1024）

Objective result



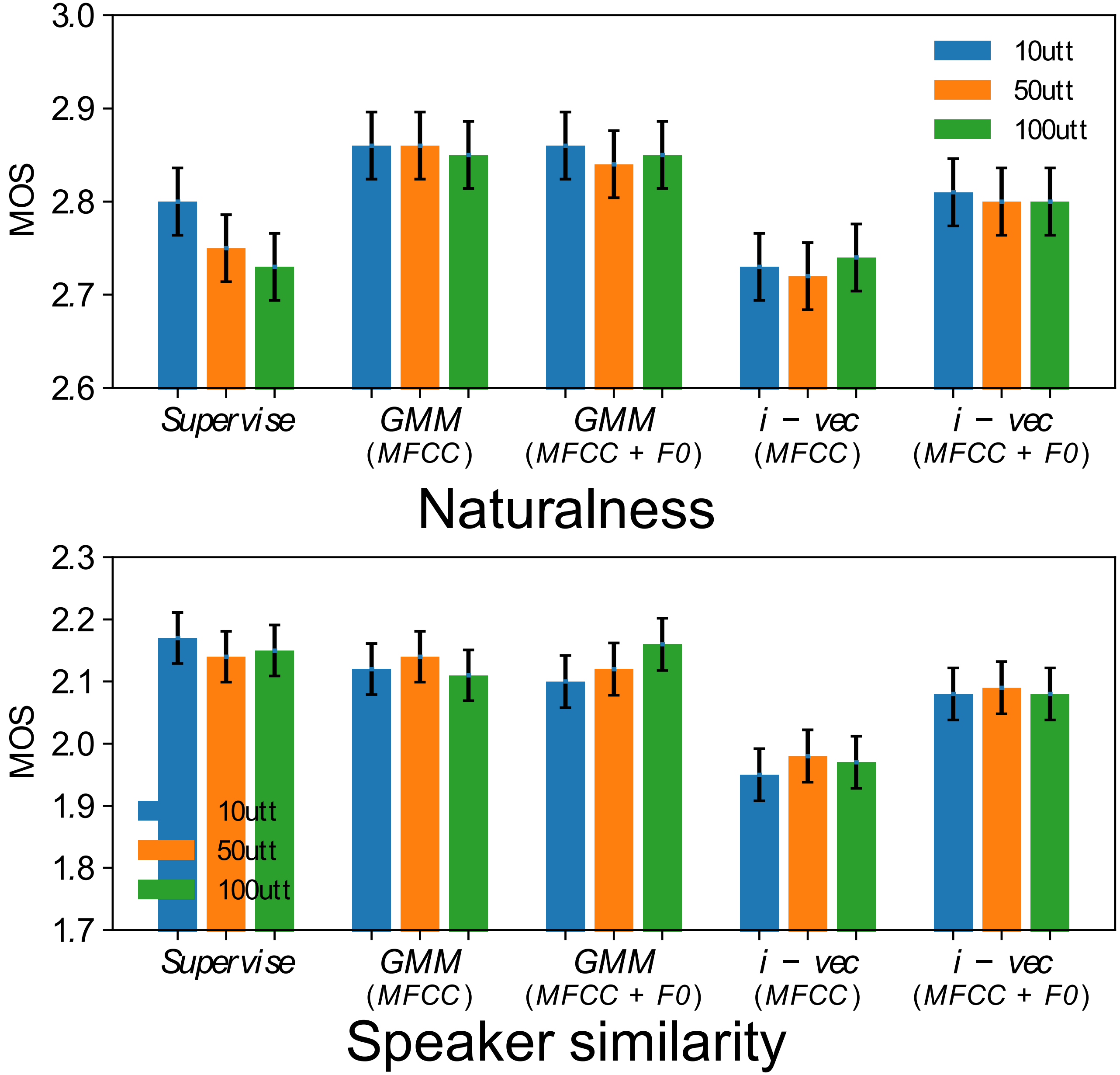


Supervised < Unsupervised (g) < Averaged

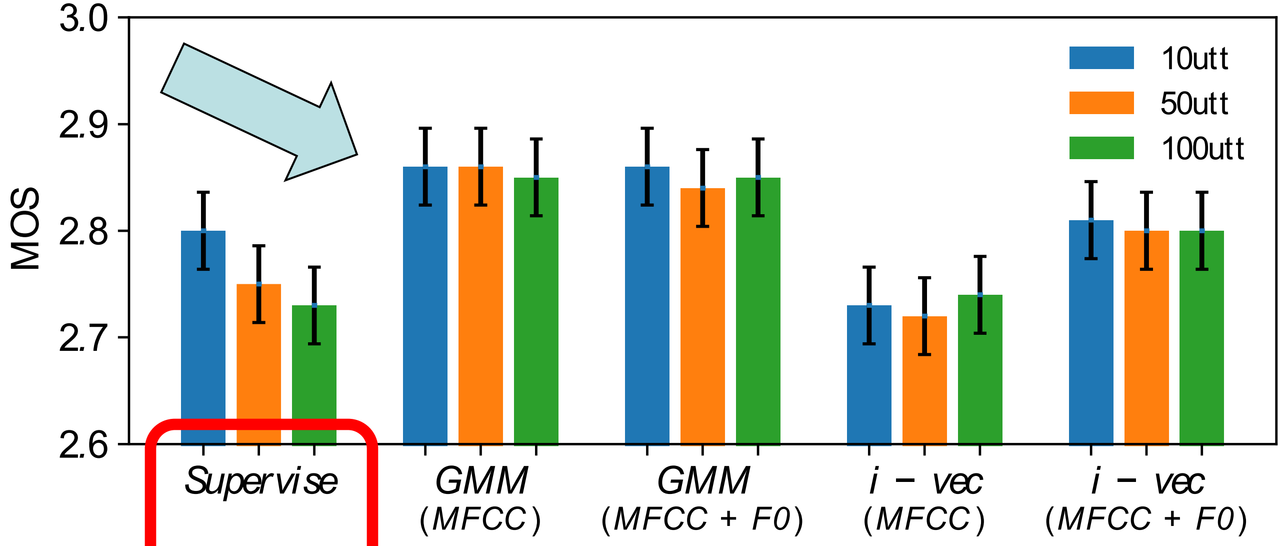
– The proposed technique successfully performed speaker adaptation

– As expected, the results of supervised systems are better

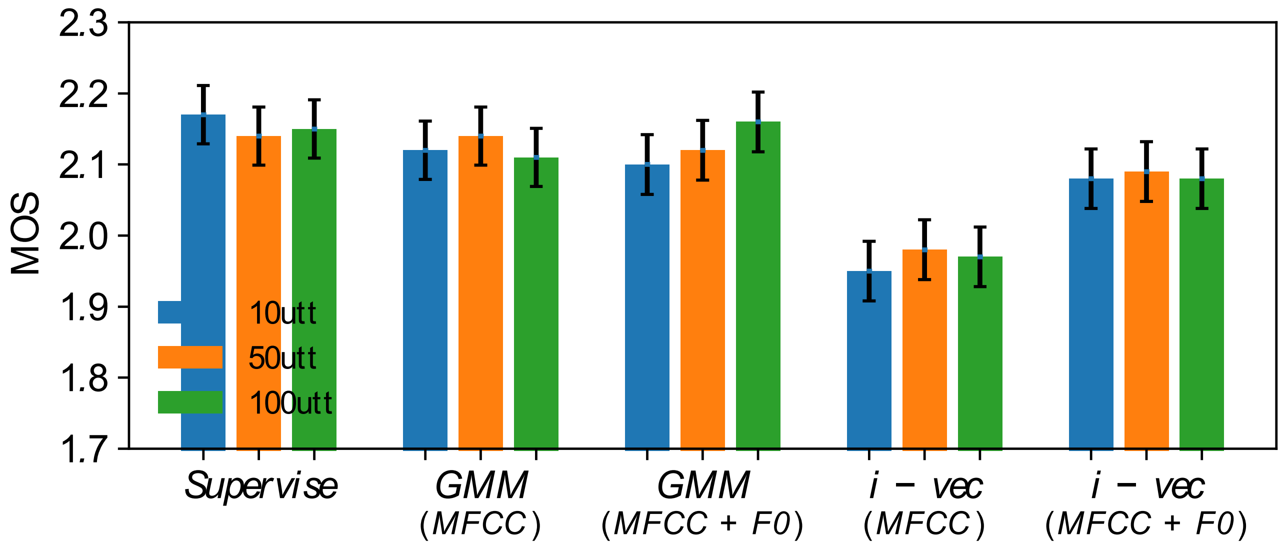
Subjective evaluation results



Subjective evaluation results

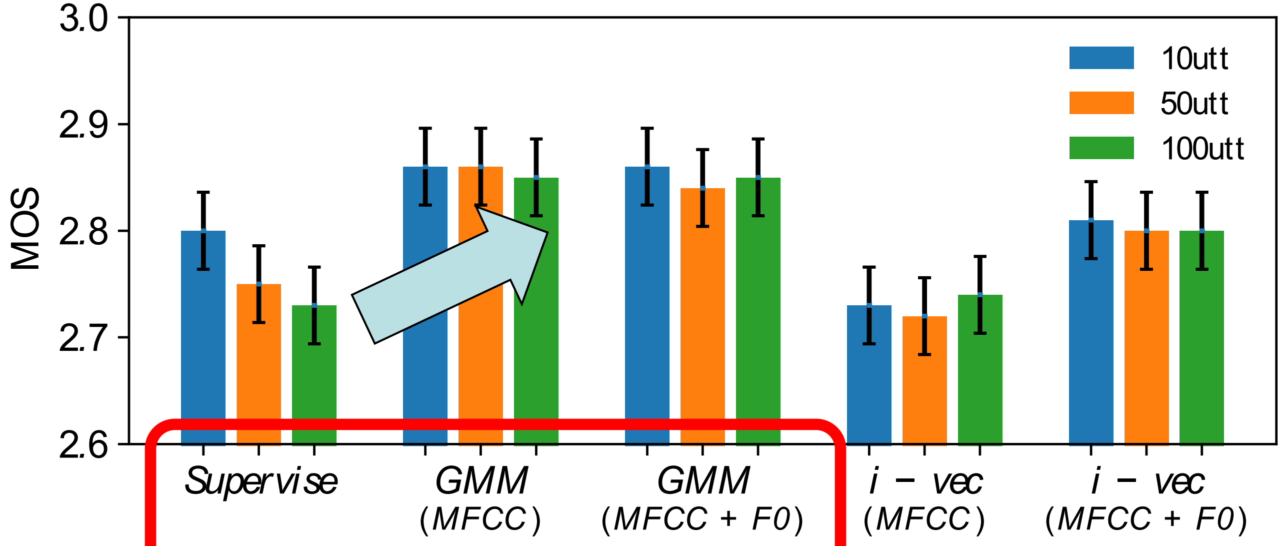


Naturalness



Speaker similarity

Subjective evaluation results



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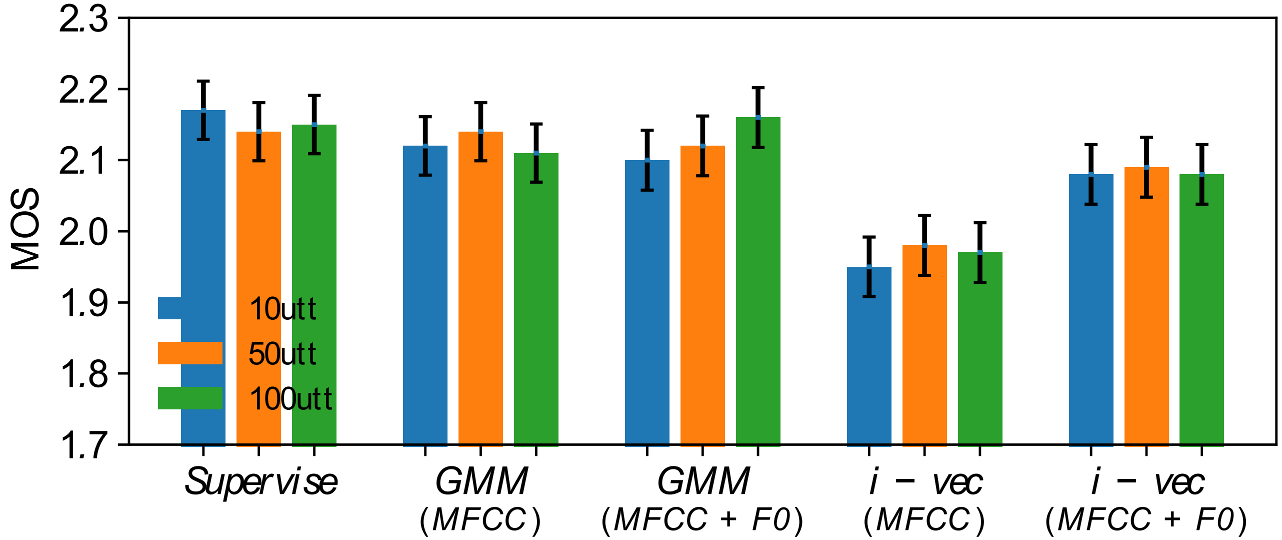

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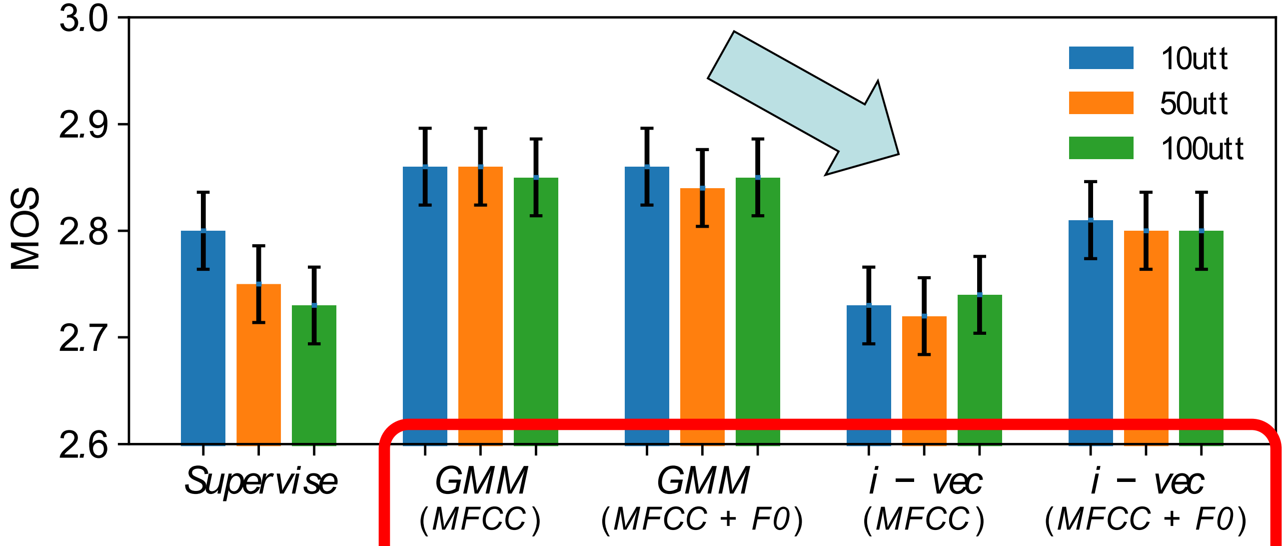

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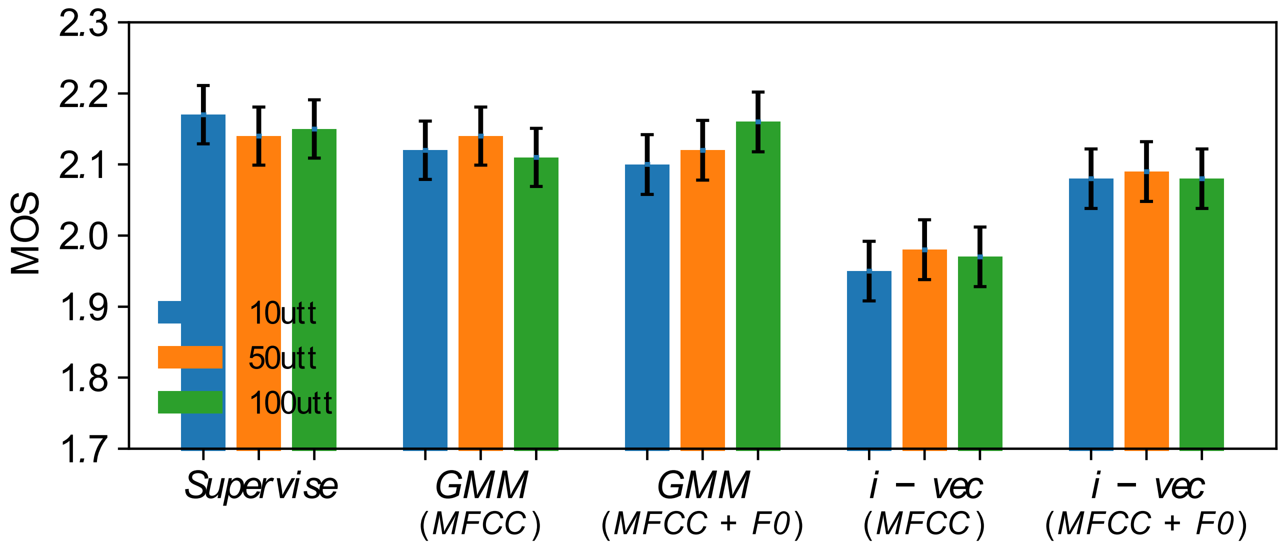

Speaker similarity

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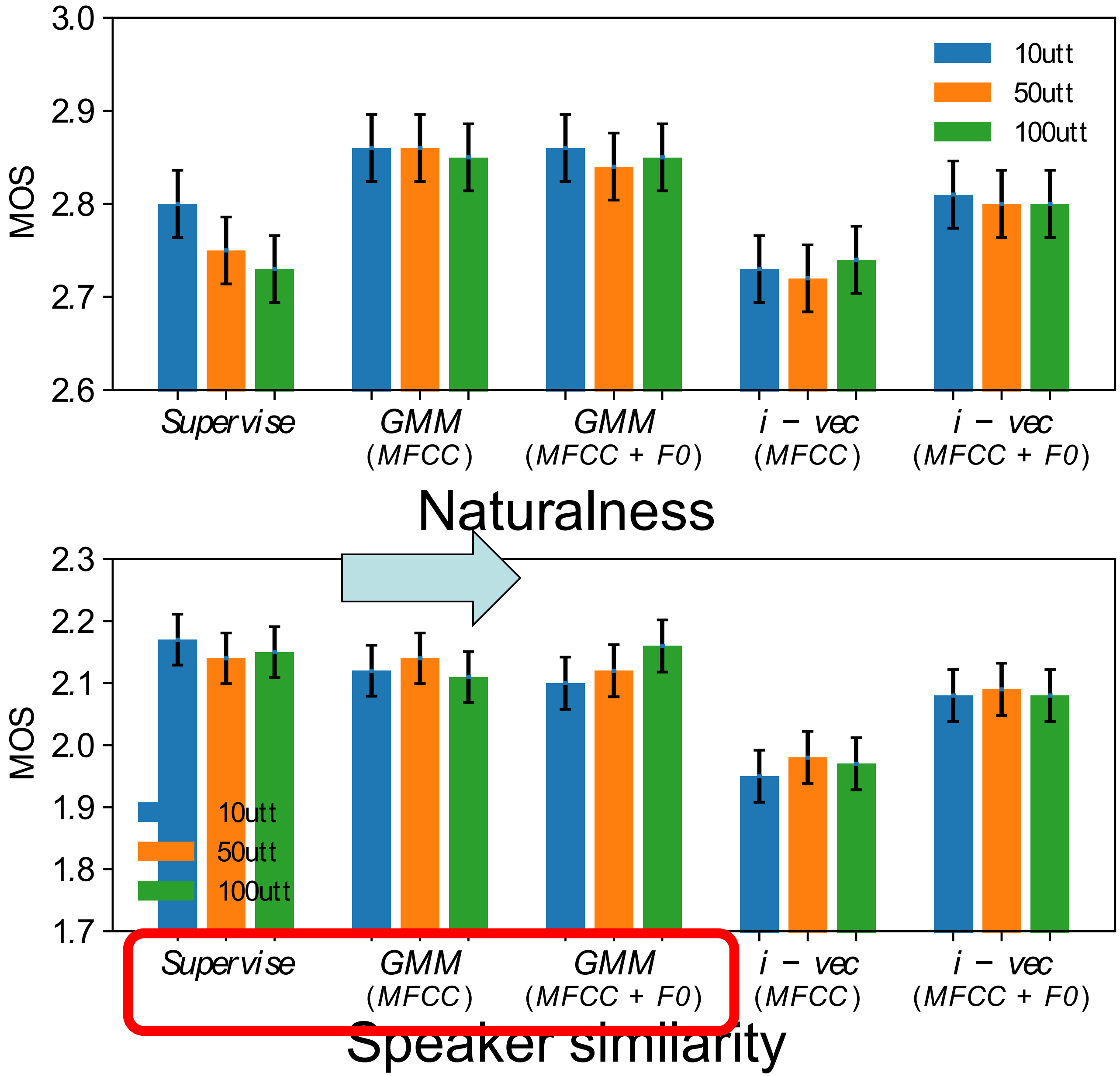


Experiments using low-quality speech data

SNR

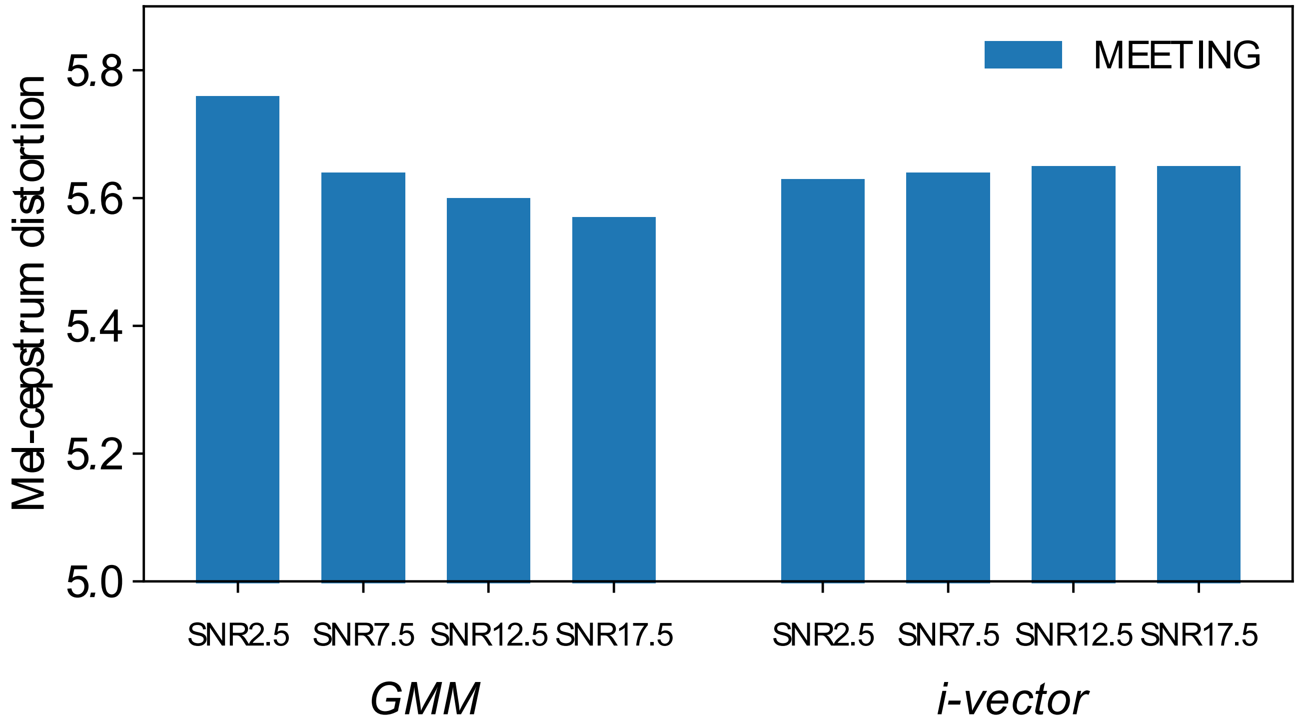
- Training data : 2.5-, 7.5-, 12.5-, or 17.5-dB
- Adaptation data : 0.0-, 5.0-, 10.0-, or 15.0-dB

Quality types of training and adaptation data

Training data	Adaptation data	Quality condition
CLEAN	CLEAN	ideal
CLEAN	OFFICE	mismatched
CLEAN	MEETING	mismatched
OFFICE	OFFICE	matched
MEETING	MEETING	matched

Objective evaluation（1/2）

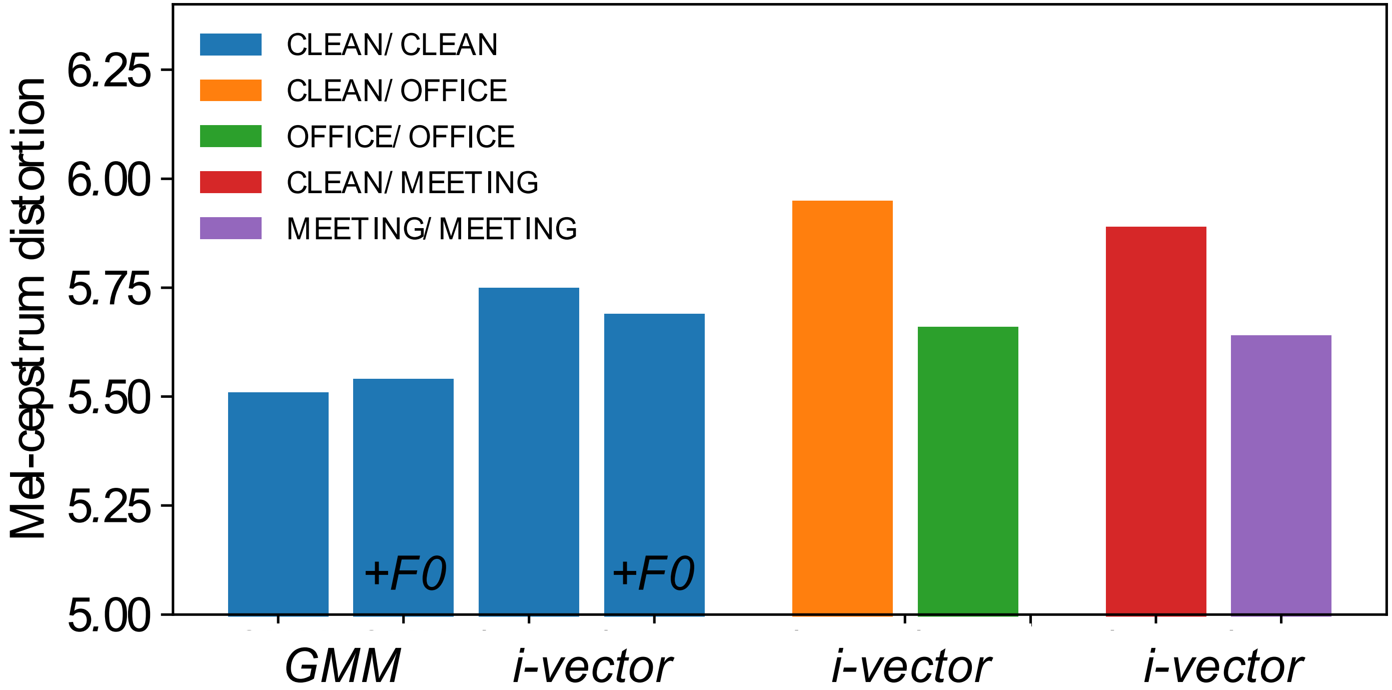
Matched condition



– The performance of i-vector/PLDA was almost the same in all SNR cases
  - i-vector/PLDA was robust for the proposed adaptation

Objective evaluation（2/2）

Labels show conditions（Training/Adaptation）

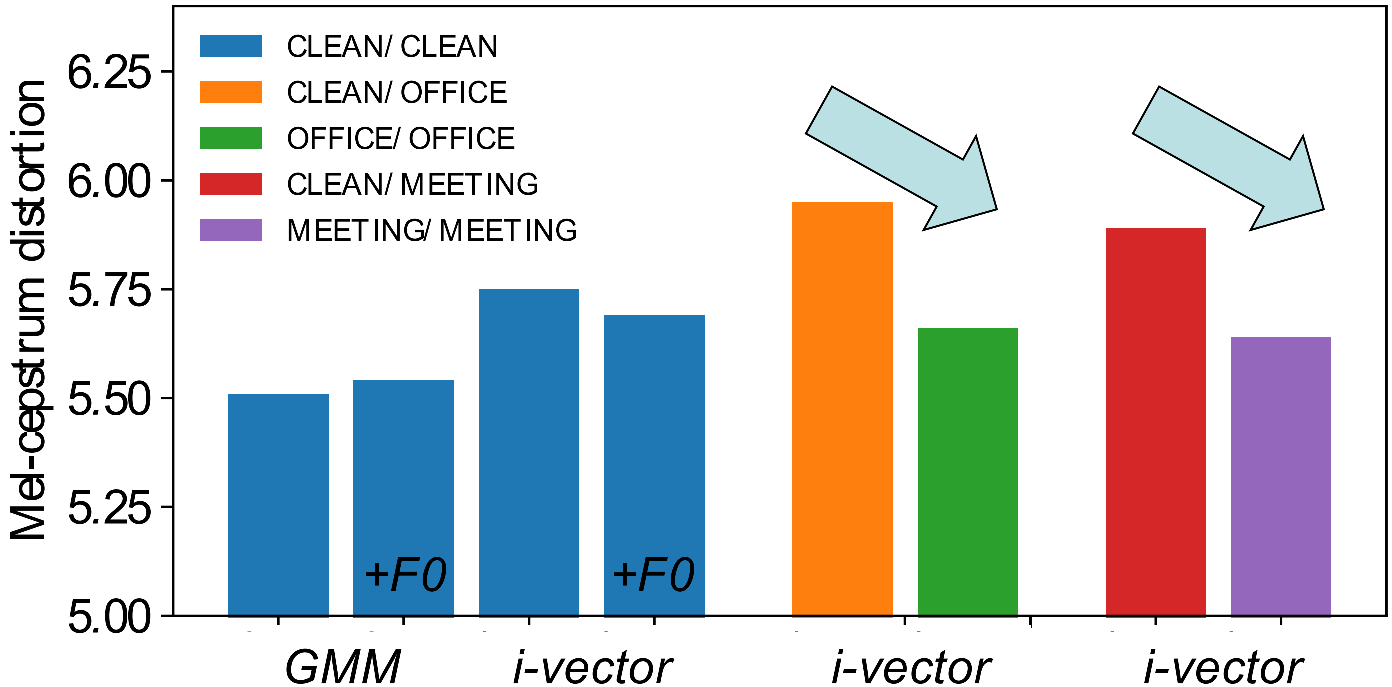


– Ideal < matched < mismatched

- Using speech data whose quality is matched to adaptation data for ASV model training improves the performance

Objective evaluation (2/2)

Labels show conditions (Training/Adaptation)



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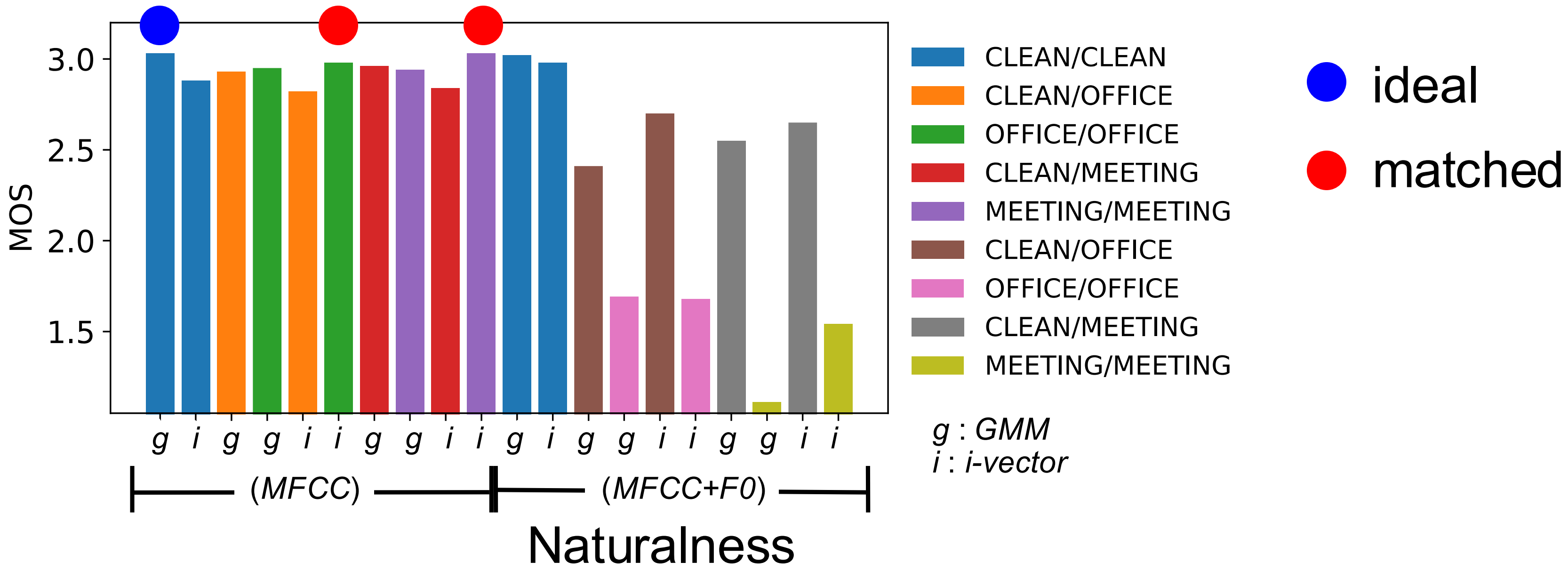
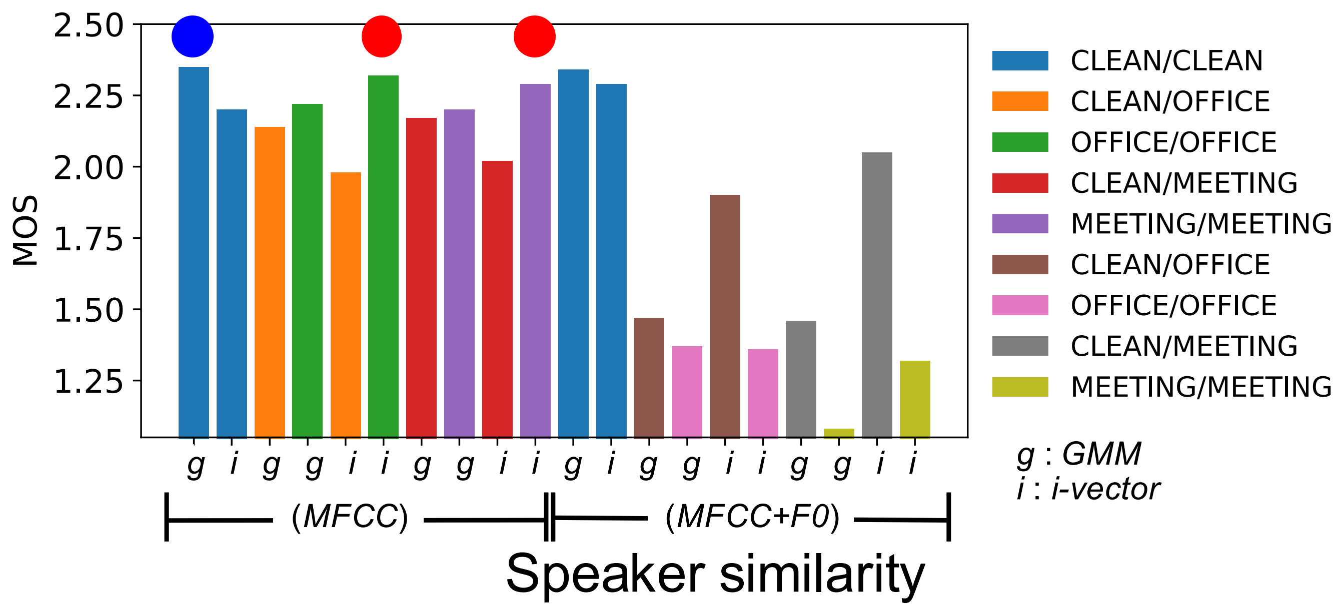
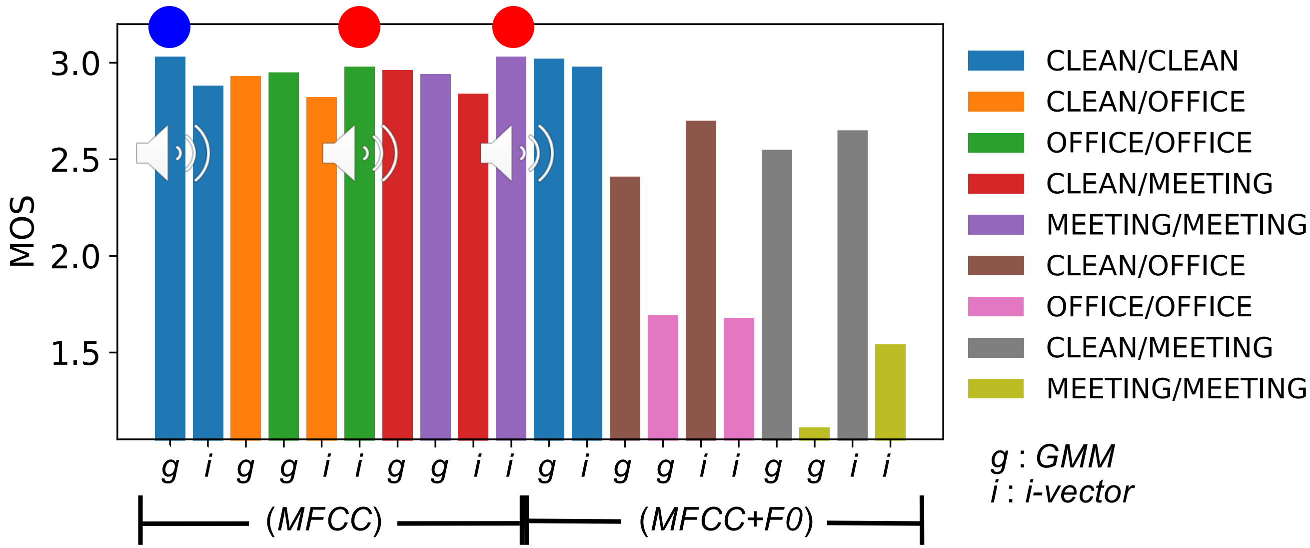
Subjective evaluation

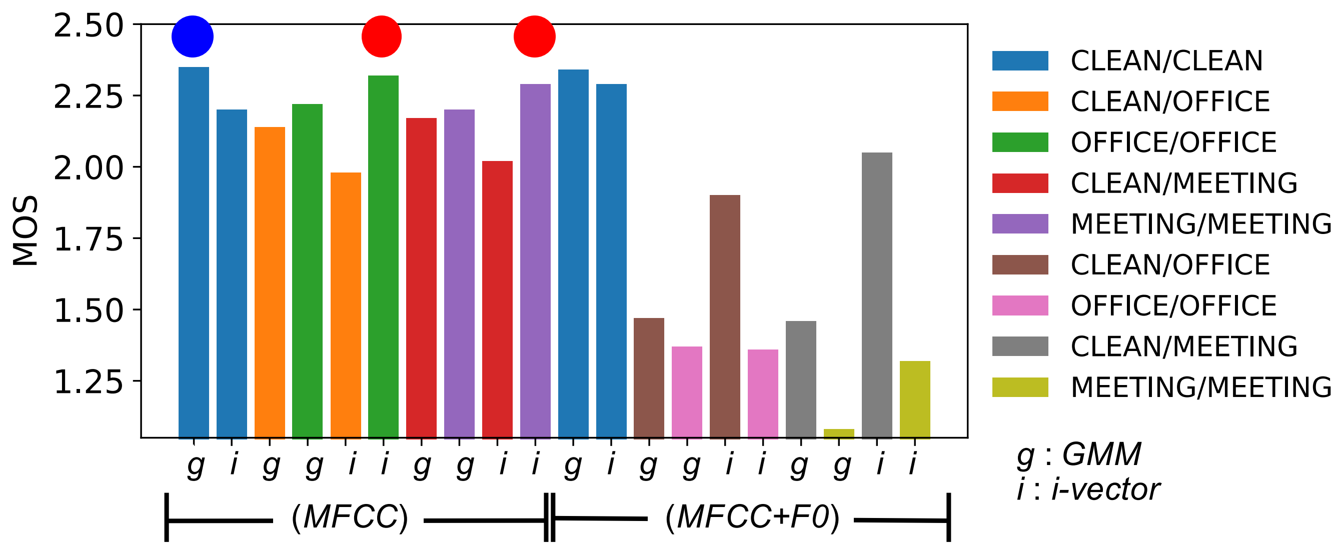




Subjective evaluation



Naturalness



Speaker similarity

ideal

matched

Target

Conclusion

Unsupervised adaptation for speech synthesis

- Adaptation using a speaker-similarity vector
  - The proposed technique change the speaker characteristics
- Robustness against low-quality adaptation data
  - Using speech data whose quality is matched to adaptation data for model training improved the performance

Future work

- MP3 or AMR codec speech
- Speech recorded under real conditions