Employing Median Filtering to Enhance the Complex-valued Acoustic Spectrograms in Modulation Domain for Noise-robust Speech Recognition

Hsin-Ju Hsieh^{1,2}, Berlin Chen² and Jeih-weih Hung¹ Presenter: Jeih-weih Hung



¹Department of Electrical Engineering National Chi Nan University, Taiwan ²Department of Computer Science & Information Engineering National Taiwan Normal University, Taiwan



Outline

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- Median filtering
- Proposed method
- Experiments and results
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Introduction

• The effect of noise in speech signal



- Extending our previous work in the task of spectrogram enhancement
 - leveraging median filtering (MF) to reduce the relatively fastvarying anomaly in the modulation spectra for acoustic spectrograms caused by noise
 - the magnitude and phase components of acoustic spectrograms can be enhanced implicitly

Median filtering

- The idea of median filtering:
 - Median filtering (MF) is a rank-order filtering technique, which is a nonlinear operation
 - often used to remove speckle noise in data while preserving the embedded sharp contrasts
 - The procedure of MF:

 $\tilde{x}_t = \operatorname{med}(x_{t-d}, \dots, x_t, \dots, x_{t+d})$

• $med(\cdot)$ denotes the median operator, which sorts the elements within the length-(2d + 1) sliding window and gives the d^{th} largest element as the output

Proposed method (1/4)

- Motivations:
 - The odB-SNR curve contains larger and sharper fluctuations than the clean noise-free one



The magnitude modulation spectral curves of the imaginary acoustic spectrograms at acoustic frequency 375 Hz under three SNR cases (noise type: airport) for one utterance in the Aurora-2 database

Proposed method (2/4)



• Step 1. Compute the *modulation spectrum* for the complexvalued acoustic spectrogram with respect to *each single acoustic frequency*

> real part: $\chi_r[k,m] = DFT\{X_r[n,k]\}\$ imaginary part: $\chi_i[k,m] = DFT\{X_i[n,k]\}\$

Proposed method (3/4)

• For both the training and testing utterances:



• **Step 2.** Update the magnitude modulation spectrum via median filtering (MF)

$$\tilde{A}_{r}[k,m] = MF\{A_{r}[k,m]\} = med\{A_{r}[k,m-d], \dots, A_{r}[k,m+d]\}$$
$$\tilde{A}_{i}[k,m] = MF\{A_{i}[k,m]\} = med\{A_{i}[k,m-d], \dots, A_{i}[k,m+d]\}$$

Proposed method (4/4)



The diagram of MAS-MF

• **Step 3.** Synthesize the acoustic spectrogram, which is then converted to speech features, MFCC

real part: $X_r[n,k] = IDFT\{\tilde{A}_r[k,m]e^{j\theta_r[k,m]}\}$ imaginary part: $X_i[n,k] = IDFT\{\tilde{A}_i[k,m]e^{j\theta_i[k,m]}\}$

Experiments and results (1/4)

• Speech Corpus

Aurora 2.0					
Speech content	0, 1, 2,, 8, 9, oh				
Training data	8440 clean sentences				
Testing data	Set A: 28028 (subway, babble, car, exhibition)				
	Set B: 28028 (restaurant, street, airport, train station)				
	Set C: 14014 (MIRS subway, MIRS street)				
SNR	clean, 20dB,15dB, 10dB, 5dB, 0dB, -5dB				

• Hidden Markov Model

- Each digit model contains 16 states
- Each state contains 20 Gaussian mixture

Experiments and results (2/4)

Recognition accuracy rates (%) on the **Aurora-2 task**, achieved by baseline MFCC and various robustness methods

	Set A	Set B	Set C	Avg.	RR
MFCC baseline	54.87	48.87	63.95	54.29	-
CMS	66.81	71.79	67.64	68.97	32.12
CMVN	75.93	76.76	76.82	76.44	48.46
HEQ	80.03	82.05	80.10	80.85	58.11
AFE	87.68	87.10	86.27	87.17	71.93
MAS-HEQ	86.49	88.13	84.98	86.84	71.21
MAS-MF (3)	87.71	88.58	86.23	87.76	73.22
MAS-MF (5)	87.52	88.81	86.39	87.81	73.33
MAS-MF (6)	87.65	88.90	86.47	87.91	73.55
MAS-MF (7)	87.53	88.85	86.31	87.81	73.33
MAS-MF(9)	87.57	88.84	86.11	87.79	73.29

RR (%) is the relative error rate reduction over the MFCC baseline

Experiments and results (3/4)

Recognition accuracy rates (%) on the **Aurora-4 task** (viz. Sets 8 to 14), achieved by baseline MFCC and various robustness methods

	MFCC	AFE	MAS-MF (3)
Clean	63.45	79.34	76.35
Car	37.27	72.56	68.03
Babble	30.31	61.58	63.13
Restaurant	34.30	55.65	58.27
Street	26.13	58.31	57.72
Airport	31.85	60.55	61.36
Train	26.95	60.88	59.56
Average	35.75	64.12	63.49

RR (%) is the relative error rate reduction over the MFCC baseline

Experiments and results (4/4)





- For the unprocessed case, the environmental noise results in a significant PSD mismatch over *the entire modulation frequency range* [0 50 Hz]
- The PSD mismatch can be considerably suppressed after performing either of AFE and MAS-MF

Conclusions

- This work provides a novel feature extraction method, MAS-MF, for robust speech recognition
 - Joint normalization of real and imaginary acoustic spectra
 - The spectral distortion is **reduced via MF**
 - MF is nonlinear and has little to do with learning
 - **Recognition accuracy** can be obviously promoted
- Several directions as to the future work
 - combining MAS-MF with other robustness methods
 - examining MAS-MF on the state-of-the-art deep neural network (DNN) scenario

Thank you for your attention

