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NOISE REDUCTION BY MULTI-MICROPHONE CONSIDERING 3-D SPACE

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1 INTRODUCTION

A large number of current speech recognition systems for clean speech show great ability in ideal environments which have no noise and reverberation. However, there are many reports, e.g. [4] that performance of these speech recognition systems reduces extremely in real environments because noises and reverberations distort speech.

Therefore, this paper proposes a new method of noise reduction as a front-end processor of speech recognition systems to make speech recognition systems robust by reducing distortions of amplitude spectrums caused by noise.

The proposed method expanded noise reduction model [1] proposed by Akagi et al, and two paired microphones are used in the horizontal and perpendicular direction. This method reduces noise coming from a point of 3-D space.

2 ALGORITHM DESCRIPTION

The new noise reduction method proposed here uses a microphone array constructed with two microphone pairs as an input device. Here, the microphone of up and down is the main-microphone pair, the microphone pair of left and right is the sub-microphone pair and the center microphone is support. (See Fig.1)

2.1 FORMULATION

Let us assume that there are one microphone pair in a noisy environment as shown Fig.2.

$s(t)$ is a signal coming from the front, and $n(t)$ is a noise coming from the direction such that the difference of arrival time between two microphones locating at both ends of a 3ch. microphone array is 2δ . The arrival sound at each microphone is assumed as follow,

where $\varepsilon_1, \varepsilon_2$ is a certain small value.

Then, the objective signal is given by subtracting the estimated noise from the signal received by the main-microphone pair.

Noise direction estimation is done by crosscorrelation.

3 EXPERIMENTS AND RESULTS

Two experiments were conducted for evaluation. One is simulation experiments using operating on synthetic signals, and the other is real environment experiments using real signals that presented by speaker in a soundproof room (reverberation time: about 50 ms).

3.1 DATA AND CONDITION

Let us assume that speech (ATR, mht14348/bunri/) come from the front and noise (Sweep Tone: start frequency 1 kHz, end frequency 6 kHz, time 48000 point, Band Noise: center frequency 3.4 kHz, bandwidth 3 kHz, duration time 48000 point) come from direction shown in Table.1. Sampling frequency is 48 kHz.

Table 1: Arrival direction of noise

Simulation Experiments	Sweep Tone : Azimuth (right 30°), Angle of elevation (up 30°)	Data 1
	Sweep Tone : Azimuth (right 60°), Angle of elevation (up 30°)	Data 2
	Band Noise : Azimuth (right 30°), Angle of elevation (up 30°)	Data 3
Real Environment Experiments	Sweep Tone : Azimuth (right 45°), Angle of elevation (up 20°)	Data 4
	Band Noise : Azimuth (right 45°), Angle of elevation (up 20°)	Data 5

Data2 is adopted for performance test of this method when main-microphone and sub-microphone pairs have equal arrival time difference of noise.

Parameters are set up shown in Table.2.

3.2 ARRIVAL TIME DIFFERENCE OF NOISE

The arrival time difference of noise is calculated as

$$\begin{aligned}
\text{main} - \text{microphone pair} &: \delta_{ud} = (2d/c) \cdot \sin \beta \\
\text{sub} - \text{microphone pair} &: \delta_{lr} = (2d/c) \cdot \cos \alpha \cdot \cos \beta
\end{aligned} \tag{9}$$

α : Azimuth, β : Angle of elevation, c : Speed of sound, $2d$: Space of microphone pair (See Fig.1)

The arrival time difference and inaccurate frequency band are shown in Table.3.

Table 2: Parameter Establishment

Parameter	Establishment Value (Simulation)	Establishment Value (Real Environment)
Sampling frequency	48kHz	the same as left
Frame Length	2048 point (Sweep Tone) 1024 point (Band Noise)	4096 point
Frame Period	1024 point (Sweep Tone) 512 point (Band Noise)	2048 point
Window Types	Hamming (Direction Estimation) Triangle (Noise Reduction)	the same as left
ε_1	0.7 (Sweep Tone) 0.05 (Band Noise)	0.9
ε_2	0.5 (Sweep Tone) 0.05 (Band Noise)	0.7

Table 3: Arrival time difference of noise,inaccurate frequency band

Arrival direction of noise	Arrival time difference (main-pair,sub-pair)	Inaccurate frequency band (main-pair,sub-pair)
$\alpha : 30^\circ$ to the right, $\beta : 30^\circ$ to the up	7point、11point	3.4*n[kHz]、2.2*n[kHz]
$\alpha : 60^\circ$ to the right, $\beta : 30^\circ$ to the up	7point、7point	3.4*n[kHz]、3.4*n[kHz]
$\alpha : 45^\circ$ to the right, $\beta : 20^\circ$ to the up	5point、9point	4.8*n[kHz]、2.6*n[kHz]

3.3 RESULTS

For evaluation, the following signal-to-noise ration(SNR) is adopted.

$$SNR = 10 \log_{10} \frac{\sum_n s^2(t_n)}{\sum_n \{s(t_n) - \tilde{s}(t_n)\}^2} \quad (dB) \quad (10)$$

where $s(t_n)$ is an original sound wave and $\tilde{s}(t_n)$ is a noise-reduced sound wave.

Results of SNR are shown in Table.4 and 5.

Results of power spectrum of Data 1 and 2 are shown in Fig.3 .

Table 4: SNR (simulation experiments)

	Data 1	Data 2	Data 3
Noise-added speech wave	-10.45 (dB)	-10.45 (dB)	-3.61 (dB)
Noise-reduced speech wave of main-pair	-2.77 (dB)	-2.77 (dB)	11.27 (dB)
Noise-reduced speech wave of sub-pair	-4.97 (dB)	-2.77 (dB)	9.54 (dB)
Noise-reduced speech wave of this method	8.61 (dB)	-2.15 (dB)	18.84 (dB)

Table 5: SNR (Real environment experiments)

	Data 4	Data 5
Noise-added speech wave	-1.83 (dB)	-1.86 (dB)
Noise-reduced speech wave of main-pair	3.07 (dB)	2.05 (dB)
Noise-reduced speech wave of sub-pair	3.54 (dB)	3.68 (dB)
Noise-reduced speech wave of this method	4.64 (dB)	3.68 (dB)

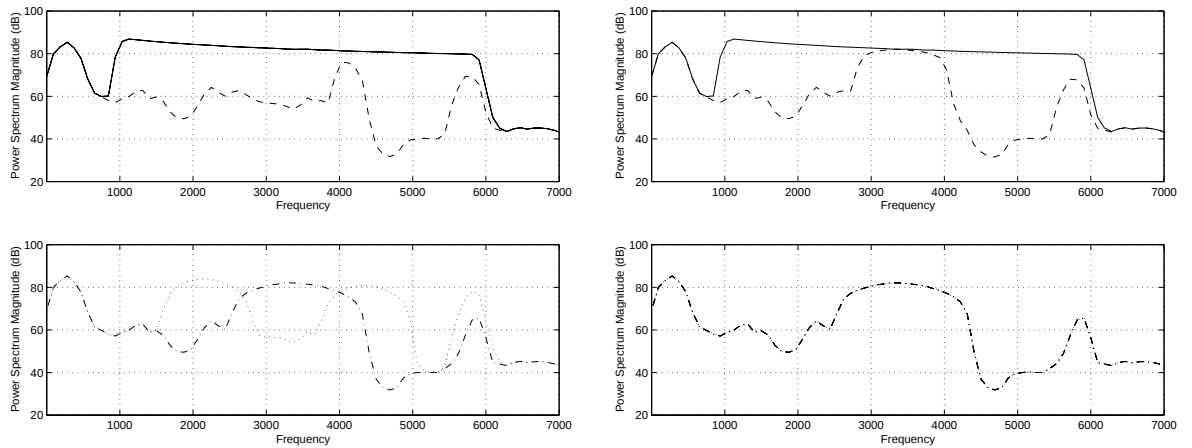


Figure 3: Power Spectrum (Fig.left : Result of Data1, Fig.right : Result of Data2) Solid line : Noise-added speech wave, Broken line : Noise-reduced speech wave of this method, Chain : Noise-reduced speech wave of main-pair, Dotted line : Noise-reduced speech wave of sub-pair

4 CONCLUSION

This paper proposed a method that reduces noise coming from a point of 3-D space with main and sub-microphone pairs.

SNRs can increase by 10 to 20 dB in simulations.

References

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