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研究課題名（和文）　次世代補聴器のための先端両耳雑音抑制技術

研究課題名（英文）　Advanced binaural noise reduction for next generation hearing aids

研究代表者
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研究成果の概要：In this research, we proposed a two-stage binaural speech enhancement with Wiener filter (TS-BASE/WF) based on the equalization-cancellation (EC) model. In the proposed TS-BASE/WF, the interfering signal is first estimated by equalizing and cancelling the target signal based on the EC model, and a time-variant Wiener filter is then applied to enhance the target signal given the noisy mixture signals. Main advantages of the proposed TS-BASE/WF are: (1) effective in dealing with non-stationary multiple-source interfering signals; (2) successful in preserving the perceptual impression of acoustic scenes. Experimental results in various spatial configurations confirm that the TS-BASE/WF outperforms the traditional binaural speech enhancement algorithms in both speech enhancement and sound localization.

交付額

（金額単位：円）

	直接経費	間接経費	合　計
2007 年度	1,900,000	0	1,900,000
2008 年度	1,300,000	390,000	1,690,000
年度			
年度			
年度			
総　計	3,200,000	390,000	3,590,000

研究分野：総合領域
科研費の分科・細目：情報学・知覚情報処理・知能ロボティクス
キーワード：両耳雑音抑圧法、二段階両耳音声強調、心理音響 EC モデル、補聴器

１．研究開始当初の背景

The traditional speech enhancement algorithms used for hearing aids yield monaural output signals, which do not preserve the binaural cues in them. The next-generation hearing aids should be able to preserve the binaural cues which can improve the recognition rate in noise due to the spatial release from masking.

２．研究の目的

The aim of this research is to develop a binaural speech enhancement algorithm for hearing aids based on the research knowledge in speech science and signal processing.

３．研究の方法

This research was conducted along the following steps:

(1) We investigated the state-of-the-art modeling techniques of binaural hearing in psychoacoustics, which including the cross-correlation based model, the auditory-nerve activity model and the equalization-cancellation (EC) model. The EC model was finally adopted in this research due to its high ability in discriminating the target signal and interference signals.

(2) Based on the EC model, a two-stage binaural speech enhancement approach with Wiener filter (TS-BASE/WF) was proposed. In the proposed TS-BASE/WF, the interfering signal is first estimated by equalizing and cancelling the target signal based on the EC model, and a time-variant Wiener filter is then applied to enhance the target signal given the noisy mixture signals. The block diagram of the proposed TS-BASE/WF is shown in Fig. 1.

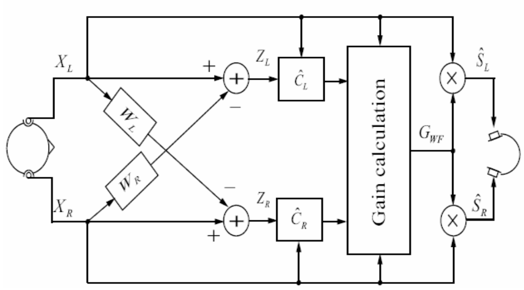


Fig. 1 Block diagram of TS-BASE/WF

(3) As for the binaural processing, the proposed TS-BASE/WF was evaluated in the sense of speech enhancement and binaural-cue preservation in various spatial configurations.

The ability in speech enhancement was examined through subjective listening tests using the mean opinion score (MOS). The results are plotted in Fig. 2. It is observed that in comparison with the traditional binaural speech enhancement systems, the proposed TS-BASE/WF algorithm exhibits the significant increases in MOS improvements, that is, the speech quality improvements.

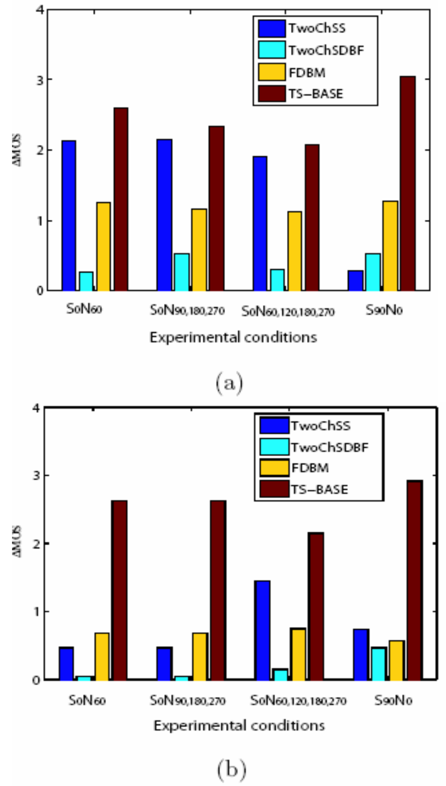
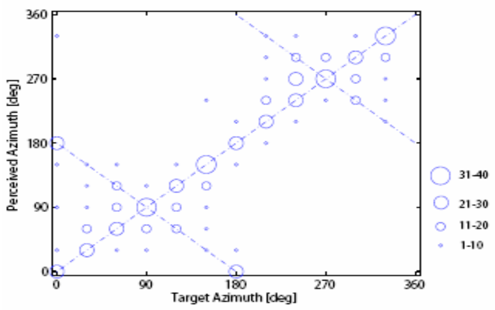


Fig. 2 MOS improvements of the studied algorithms at the left ear (a) and the right ear (b) in the different acoustical conditions.

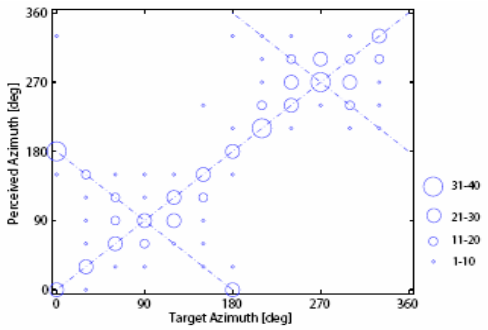
The ability in binaural-cue preservation was assessed in terms of perceptual sound localization in different spatial scenarios. The localization results are plotted in Fig. 3. The perceived directions are distributed along the diagonal line, that is, closely consistent with those of the real input signals. The front-back confusion is evidently observed as well. In comparison with the results in the tested two spatial conditions, the variances of the perceived directions for the target signals in the one-noise-source condition are slightly lower than those in the three-noise-source conditions.

４．研究成果

In this research, we proposed a two-stage binaural speech enhancement with Wiener filter (TS-BASE/WF) based on the psychoacoustic EC model for high-quality speech enhancement. The effectiveness of the proposed TS-BASE/WF in suppressing interfering signals and enhancing target signals was proved by the



(a)



(b)

Fig. 3 Results of perceptual sound localization results in the one-noise-source conditions ($S_{0:30:330}N_0$) (a) and the three-noise-source conditions ($S_{0:30:330}N_0$) (b). .

comprehensive evaluations. All evaluations demonstrated that in comparison with the traditional algorithms, the proposed TS-BASE/WF yields the highest speech quality and provides the accurate sound localization performance in different acoustic spatial configurations. These characteristics of the TS-BASE/WF system enable it promising to be applied for hearing aids system.

Supported by this research grant, we have published 1 patent and 3 Journal papers, 14 conference papers which are listed in the following.

５．主な発表論文等

（研究代表者、研究分担者及び連携研究者には下線）

〔雑誌論文〕（計　3　件）

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〔産業財産権〕
○取得状況（計１件）

名称：音声強調処理システム
発明者：鈴木陽一，坂本修一，李軍鋒，本郷哲
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国内外の別：国内

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