A Predefined Command Recognition System Using a Ceiling Microphone Array in Noisy Housing Environments

Yoko Sasaki *a*b y-sasaki@aist.go.jp Satoshi Kagami *b*c*a s.kagami@aist.go.jp Hiroshi Mizoguchi *a*b hmr@s.noda.tus.ac.jp Tadashi Enomoto *d enomoto.tadashi@b5.kepco.co.jp

Abstract

This paper describes a speech recognition system that detects basic voice commands for a mobile robot operating in a home space. The system recognizes arbitrary timed speech with position information in a noisy housing environment. The microphone array is attached to the ceiling, and localizes sound source direction in azimuth and elevation, then separates multiple sound sources using Delay and Sum Beam Forming (DSBF) and Frequency Band Separation (FBS) algorithm. We implement the sound localization and separation method on our 32 channel microphone array. The separated sound source is recognized using an open source speech recognizer. These sound localization, separation and recognition functions are implemented as online processing in real world. We define four indices to evaluate the performance of the recognition system, and the efficiency in a noisy environment or with distant sound sources is confirmed from experiments in varied conditions. Finally, an application for a mobile robot interface is reported.

1. INTRODUCTION

Auditory system is useful for a robot system to communicate with human, or for the initial recognition of environmental change. To achieve efficient auditory functionality in the real world, combination of sound localization, separation and recognition in a total system is needed.

There is an increasing amount of research developing robotic audition systems for human-robot interaction, and varied methods are proposed for each function of auditory system. Study on Blind Source Separation (BSS) based on Independent Component Analysis (ICA) is well performed. A real-time sound separation system [1] which has two microphones was proposed and showed efficient performance on hands-free sound recognition. It is applied to robot as human-robot verbal conversation interface. The method is known for high performance sound separation with small system. On the other hand, assumption for frequency independence between sound sources causes limitation for reverberant field or handling multiple sound sources.


Several methods are proposed for implementation in real world. GMM-based speech end-point detection method and outlier-robust generalized sidelobe canceler was implemented using a 12ch microphone array [4]. Missing feature algorithm is proposed in order to achieve robust localization and segmentation method [5].

As for total auditory system, real-time auditory system [6] is reported with simultaneous speech recognition system implemented on a robot embedded microphone array. It works for multiple sound sources in near range and shows efficient recognition rate for known speech source coming from front of the robot head using a preliminary measured impulse response and optimized parameters for recognition. Nakadai et. al reported an application of simultaneous multiple sounds recognition system using this system [7].

In this paper, we present for a newly developed online voice command recognition system in noisy environments using a microphone array. In order to achieve the detection of verbal command from arbitrary time and position in a home space, following functions are needed: 1) works as online system, 2) without using environmental information, 3) robust recognition to various sound sources including non-speech sound sources. We propose a command recognition system using beamforming based sound localization and separation method implemented on a microphone array in each room of a home space. The extracted sound sources are recognized using a open source recognizer for close-talking microphone. The proposed system doesn’t need preliminary measured environmental information, and can adapt to various environmental conditions such as multiple target sound sources which require simultaneous recognition, or distant target sound sources together with noisy sound in near range of microphone array. We define four indexes to evaluate the recognition system and experiments to measure performance in varied conditions are performed. Finally, an application of the voice command recognition system for controlling mobile robot in a home environment is shown.

*a : School of Science and Technology, Tokyo University of Science
b : Digital Human Research Center, National Institute of Advanced Industrial Science and Technology (AIST)
c : CREST, Japan Science and Technology Agency (JST)
d : Kansai Electric Power Company, Inc.
2. Sound Localization and Separation

This section describes our approach to localize and separate multiple sound sources using microphone array.

2.1. FBS Based 2D Sound Localization

Sound localization method has two main parts. First, Delay and Sum Beam Forming (DSBF) to enhance focused direction’s signal to get sound pressure distribution, named spatial spectrum. Second, Frequency Band Selection (FBS)[9] as a kind of binary mask to filter out louder detected sound’s signal and localize smaller power sound source simultaneously. The localization system detects multiple sound sources from the highest power intensity to the lowest at each time step.

Aligning the phase of each signal amplifies the desired sounds and attenuates ambient noise. Let O be the microphone array center, and $C_j$ the focus position on the $O$ centered hemisphere which has a larger radius than array size. Focus position is set by a linear triangular grid on the surface of sphere[9] to localize sound source position in azimuth and elevation. Using the median point on the surface of sphere to localize sound source position. Using the median point on the surface of sphere to localize sound source position. Using the median point on the surface of sphere to localize sound source position. Using the median point on the surface of sphere to localize sound source position.

Let $t$ be time and $(\theta, \phi)$ be azimuth and elevation of $C_j$. Delay time $\tau_{ji}$ for $i$-th microphone ($1 \leq i \leq N$) is defined by microphone arrangement. Observed wave $y_j$ in all directions is expressed as equation (3):

$$y_j(\omega, t) = \sum_{i=1}^{N} W_i(\omega, \theta, \phi) \times x(t + \tau_{ji})$$ \hspace{1cm} (1)

where $N$ is number of microphones and $W_i$ is corrective weight of microphone’s directivity.

We apply FBS method after DSBF to detect multiple sound sources. FBS assumes that the frequency components of each signal are independent and is a kind of binary mask that segregates targeted sound sources from mixed signal by selecting the frequency components judged to be from the targeted sound source.

The process is as follows. Let $X_a(\omega)$ be the frequency components of DSBF-enhanced signals of position $a$ and $X_b(\omega_j)$ be those of position $b$. Selected frequency component $X_{as}(\omega)$ for position $a$ is expressed as in Equation (2):

$$X_{as} = M_a X_a \Rightarrow \begin{cases} 1 & \text{if } X_a(\omega) \geq X_b(\omega) \\ 0 & \text{otherwise} \end{cases} \hspace{1cm} (2)$$

This process rejects the attenuated noise signal from the DSBF-enhanced signal. The segregated waveform is obtained by inverse Fourier transform of $X_{as}(\omega)$. When frequency components of each signal are independent, FBS completely separate the desired sound source. This assumption is usually effective for human voice or everyday sounds of limited duration.

The spatial spectrum for directional localization, which indicates sound pressure distribution over a frame length $N$ is described as follows:

$$Q_k(\theta, \phi) = \frac{\sum_{i=0}^{\omega} (\prod_{k=0}^{K-1} (1 - M_k(i))) \times \sqrt{|Y(t)|^2}}{\sum_{i=0}^{\omega} (\prod_{k=0}^{K-1} (1 - M_k(i)))}$$ \hspace{1cm} (3)

where $Y(\omega)$ is Fast Fourier Transform of DSBF enhanced signal $y$, and $M_k$ is separation vector for the $k$-th loudest sound source generated by FBS.

Fig. 1 shows calculation flow of FBS based multiple sound localization. In DSBF phase, the system scans the formed beam of DSBF on each spherical grid, and obtains spatial spectrum which indicates sound pressure distribution on hemisphere. In FBS phase, the system first detects the loudest sound direction as the maximum peak of spatial spectrum, then it filters out the loudest sound signal by FBS and localize the second stronger sound source, and so on.

2.2. FBS Based Multiple Sound Separation

Sound source separation algorithm is almost the same as localization. It segregates the sound sources of detected directions from localization stage. For robust recognition, mask $M$ in equation (2) is revised, and mask vector $M'$ for separation is expressed as equation (4):

$$M'_a(\omega) = \begin{cases} 1 & \text{if } X_a(\omega) \geq X_b(\omega) \\ 0.5 & \text{else if } X_a(\omega) \geq 0.5X_b(\omega) \\ 0 & \text{otherwise} \end{cases}$$ \hspace{1cm} (4)

For recognition stage, sound existence interval is extracted by commonly used power based Voice Activity Detection (VAD) function from separated sound sources. It assumes there are intervals of silence before and after speech, and detects beginning and end time of speech with reference to maximum power during the separated stream. This simple VAD is sufficient for our proposed system in that assumption, because VAD for our system
is mainly used to reject broken streams during a separation interval. It is applied for separated sound streams, then each stream always contains some sound signals, and noise level is stationary during a separated sound stream.

3. COMMAND RECOGNITION SYSTEM

This section describes overview of our command recognition system for online implementation.

3.1. System Overview

For online implementation, determining when the system should stop separation and start recognition is an important problem. Our system decides separation and recognition interval $T_{\text{max}}$ from the longest sentence in command dictionary. By separating past $T_{\text{max}} + T_p (T_p \geq T_{\text{max}})$ with a cycle of $T_{\text{max}}$, the system can detect arbitrary timed voice command at more than one of separated sound stream.

Fig. 2 shows calculation flow of the command recognition system. Localization module outputs azimuth and elevation pairs at each cycle of FFT data length, and the instantaneous sound source localization keeps running continuously. Sound source separation and recognition modules are running with $T_{\text{max}}$ intervals. From instant localization results of past $T_{\text{max}} + T_p$, it estimates the number of sound sources and their positions, then segregates each detected sound source. VAD is applied for each separated source, and extracted sound sources are input to recognizer, if the data is not interrupted by a separation interval.

3.2. Microphone Array

The proposed command recognition system is tested using 32 channel microphone array unit attached on the ceiling. Fig. 3 shows the microphone array and its microphone arrangement. The microphone array has 32 omni-directional electret condenser microphones and can sample 32 data channels simultaneously. Sampling frequency is 16(kHz) and resolution is 16(bit). It localizes azimuth omni-directionally (0 to 359(deg)), and elevation from directly below (0(deg)) to horizontal direction(90(deg)).

For implementation, our system set 160 grids on surface of hemisphere initially, and each grid is divided into four smaller triangles for fine search at detected source positions. Using 32ch array, the system works well for multiple sound sources and power difference in varied environments without using environmental information.

Fig. 3 32 channel microphone array on ceiling

Six microphone array units are attached to the ceiling of the experimental house "Holone". Each unit works independently and recognition results with position information are sent to a mobile robot. Fig. 4 shows pictures of the microphone array units and the arrangement in each room.

Fig. 4 Microphone array units in experimental house "Holone"

4. EXPERIMENT

This section shows the experimental result of proposed command recognition system installed to 32 channel cel-
ing microphone array. For implementation, calculation interval $T_{\text{max}}$ is set to 3.0(sec), and data length parameter $T_p$ is set to 3.5(sec).

### 4.1. Evaluation of Sound Source Localization

Accuracy of two dimensional sound source localization is evaluated. The experimental room has a reverberation time ($T_{60}$) of 450 (msec) and background noise level($L_A$) is 32 (dB). For calculation, the frame length is 1024 points (64 msec) for one instance of localization. For evaluation of 2D directional localization, angular error is defined as the intervector angle between direction of the estimated sound source and that of the real sound source. As shown in Fig. 3, the microphone array has rotationally-symmetric arrangement on one plane, and the performance of localization in the azimuth direction is uniform.

Experiments are performed in two different conditions. The first condition (C1) is 1 sound source at different elevation angles. A loud speaker playing male speech is used as the sound source. Distance $(r)$ between microphone array and speaker is 2.0(m), and the sound source’s SNR is about 15(dB) from background noise.

The second condition (C2) is 2 sound sources at different distances. One loud speaker is set at $(r, \theta, \phi)=(2(m), 180(\text{deg}), 60(\text{deg}))$ from the array center, and it plays female speech(not including command sentences) or classical music as noise source. The other loud speaker is set at 90(\text{deg}) for azimuth angle and is varied over horizontal distance 1, 3, 5, 7 and 10 (m). Height from the microphone array stays constant at 1.15(m). Volume of the 2 sources is the same, and the SNR is about 15(dB) from background noise.

Fig. 5 shows the result of condition C1 to evaluate performance shift between elevation angle. The result is average angle error of 200(sec) data for each direction. The angular error is 9 to 18(\text{deg}) and is not dependent on elevation angle. For more than 45(\text{deg}) of elevation angle, azimuth error is smaller than elevation and angle error is close to elevation error. This indicates that localization performance of elevation angle is weaker than azimuth direction and angle error is mainly caused by elevation error for more than 45(\text{deg}) of elevation.

### 4.2. Evaluation Index of Recognition System

Four indexes are defined to evaluate the recognition system.

- **Word Correct Rate**: proportion of correctly recognized words to total recognized words
- **Task Achievement Rate**: proportion of correct recognized command at target position and timing to total command utterance
- **Error Recognition Rate**: proportion of erroneously recognized command to total number of separated sound sources
- **Target Separation Rate**: proportion of separated sound sources at correct position and time to total command utterances

Word Correct Rate shows quality of separated source for recognition, and error is mainly caused by phenotypic variation of spoken Japanese. For example, both "(place) (ni) (mo q te i q te)" and "(place) (e) (mo q te i ke)" mean "bring it to (place)", but second and third words are registered independently in the word dictionary, and the index is 1/3 in this instance. Task Achievement Rate excludes such difference. It counts the recognized sentence when the meaning is correct, and the index shows the efficiency of an application. Error Recognition Rate contains two different errors, one is error recognition of known voice command, and the other is false positive recognition of non-command sound source. Both errors affect applications using the recognition system, and the index also shows efficiency of application. Target Separation Rate counts separated sound sources after VAD, whose position and timing corresponds to real events. The index
shows performance of proposed localization and separation method.

As shown in Fig. 2, separated sound sources can overlap, and the system sometimes recognizes one voice command in two intervals. Such overlapped recognition is excluded from calculating Task Achievement Rate and Target Separation Rate.

In addition, calculation cost is evaluated by calculating elapsed time for three separate parts. First is the time elapsing from the start of voice command phonation to the start of separation module (A). Second part is from the start of separation module to output of separated sound sources after VAD (B). Third part is elapsed time of command recognition (C). A+B+C gives the total time from start of voice command phonation to the output of the recognition result.

4.3. Basic Evaluation of Recognition System

The performance of the recognition system is evaluated. Experimental conditions are same as in section IV-A.

The result of condition C1 is shown in Fig. 7. At 0 and 90 (deg) in elevation angle, target separation rate (pink line with square mark) is less than 90 (%), on the other hand, the performance has no large difference over elevation angles. Task recognition rate (green line with X mark) is similar to target separation rate, and the word correct rate is higher than 87(%). The error recognition rate is near 0 (%), and the result suggests that the localization performance directly below the microphone array(0(deg)) and at horizontal direction (90(deg)) is a little worse than at other areas, on the other hand, the system shows constant performance over elevation angle.

Fig. 7 Results of elevation changes (1 sound source)

The result of distance evaluation is shown in Fig. 8. The experimental setup is condition C2 explained in section IV-A. The evaluation result is shown in Fig. 8. For less than 7 (m) distance, target separation rate is 100 (%). At 7 (m) distance, sound pressure level of received signal at microphone array is -5 (dB) compared to the noise sound source set at 2(m) distant from the array. This result shows the efficiency of the proposed sound localization and separation system for multiple sound sources with different levels of sound pressure: On the other hand, the task recognition rate is drops with distance for distances more than 7 (m). Error recognition rate is less than 10 (%) and doesn’t change with distance, but is higher than result in condition C1 overall. Error is mainly caused by noise sound source, which contains sounds not registered in the command dictionary. Degradation of sound separation for distant sound sources does not affect to error recognition rate.

4.4. System Evaluation for Three Sound Sources

The performance of 3 sound sources are evaluated. Three experiments are performed as follows:

- EXP-A: command at (2.0(m), 90(deg), 60(deg)), classical music at (2.5(m), 45(deg), 72(deg)) and female speech at (1.4(m), 180(deg), 45(deg))
- EXP-B: command at (2.0(m), 90(deg), 60(deg)), classical music at (2.5(m), 45(deg), 72(deg)) and command at (2.0(m), 180(deg), 60(deg))
- EXP-C: command at (2.0(m), 60(deg), 60(deg)), female speech at (1.4(m), 135(deg), 45(deg)) and command at (3.2(m), 180(deg), 72(deg))

Positions are described as (distance, azimuth, elevation), and the volume of each sound source is same level.

Table 1 shows the evaluation indexes of 3 sound sources condition. EXP-B and EXP-C have two command utterances and the indexes are calculated as a combined value. The indexes of EXP-B are a little smaller than EXP-A, but there is not so much of a difference. The result of EXP-C is worse than the others. The performance degradation is mainly caused by the low recognition rate for a distant command utterance at (3.2(m), 180(deg), 72(deg)). The system failed to detect this sound source, affected by the presence of female speech close to microphone array. High percentage of error recognition rate is mainly caused by error recognition for the female speech. This result suggests that the recognition system has greater performance for detected commands and multiple command input at same time, on the other hand, it is susceptible to unknown signal input.

4.5. Processing Time of the Recognition System

Table 2 shows the average elapsed time of each experiment. The system has a Pentium-4 3.0 GHz processor operated using Debian linux 2.4.27. The result of one sound source is average of all 1400(sec) data of condition C1, which has arbitrary timed utterances. The results of
two and three sound sources are the average of 800(sec) data for each condition.

Table 2 Elapsed time from start of voice command phonetion

<table>
<thead>
<tr>
<th>number of sound source</th>
<th>1</th>
<th>2</th>
<th>3</th>
</tr>
</thead>
<tbody>
<tr>
<td>(A) time before start separation (sec)</td>
<td>2.01</td>
<td>2.05</td>
<td>2.02</td>
</tr>
<tr>
<td>(B) elapsed time for separation (sec)</td>
<td>0.86</td>
<td>1.24</td>
<td>1.37</td>
</tr>
<tr>
<td>(C) elapsed time for recognition (sec)</td>
<td>0.61</td>
<td>0.54</td>
<td>0.61</td>
</tr>
<tr>
<td>(A+B+C) total processing time (sec)</td>
<td>3.48</td>
<td>3.83</td>
<td>4.00</td>
</tr>
</tbody>
</table>

Processing time changes with environmental condition because calculation cost of sound separation and recognition is depends on number of sound sources. The time before the start of separation is 2.024(sec), averaged over all conditions, and average time of command utterance is 2.34(sec) (1.08(sec) at minimum and 2.89(sec) at maximum). This indicates that calculation interval $T_{max}$ is valid for the tested application.

4.6. Voice Command Recognition in Housing Environment

This section describes experimental results using the ceiling microphone array at housing environment "Holone". System performance is evaluated in bedroom (unit 6 in Fig. 4 a)). The room has reverberation time ($T_{60}$) of about 600(msec) and background noise level is about 40 (dB). 11 experiments are performed. The experimental conditions are shown in Table 3. Data length of each experiment is 70 (sec). Positions of sound sources are described as (azimuth, elevation) angles. The height of the sound sources from the microphone array is stationary at 1.5 (m), and distance between microphone array and sound source is different in each condition. Command set A and B is arbitrary timed command utterance which randomly selected from the command dictionary, and female speech doesn’t contains command sentence in dictionary. All experimental conditions have 2 sound sources, and SNR between sources is about 0(dB). In EXP-1 to 4, two sound sources have enough distance. The condition in EXP-5 to 8 is 30 (deg) in azimuth interval between 2 sources and elevation of command utterance is 60 (deg). The condition in EXP-9 to 11 is 30 (deg) in azimuth interval between 2 sources and elevation of command utterance is 30 (deg).

The evaluation result is shown in Table 4. Excluding
EXP-9, target separation rate is near 100 (%) for the whole experiment. In EXP-9, it has minimum distance between 2 sound sources, and the system sometimes fail to detect two sound sources independently. The worst score of task achievement rate is 62.9 (%) in EXP-9. Error recognition rate becomes high when noise sound source contains speech compared to when it is classical music.

Directional localization errors as an intervector angle between estimated and real sound positions of EXP-4, 9 and 11 are shown in Fig. 9. Angular error of EXP-4 and 11 is constant during the experiment, and result of EXP-4 which has larger intervals between two sound sources performed better than result of EXP-11. Angle error of EXP-9 is varied in time. This is attributed to the other sound source near the target sound source.

4.7. Application for Mobile Robot

The recognition system is applied to a mobile robot to confirm the ability of the system. The mobile robot has autonomous mobility in a known environment[11]. Input of a laser range finder mounted on the robot is used for localization, a particle filter based method locating a position and orientation of the robot in a map. Optimized A* algorithm [12] is used to plan a trajectory of the robot to the goal position. The mobile robot is given the recognized verbal command with sound position information from a ceiling microphone array system.

Fig. 10 shows video clips of the experiment at "Holone". A user watching TV in living room tells the robot(currently in the bed room) to go to the kitchen(a). Living room microphone array detects the user utterance and sounds from TV, and separates them from each other. When the robot receives the command, it plans the path from the bedroom to the kitchen and starts moving(a,b). In the kitchen, a user places a coffee cup on the robot and orders to the robot to go to the study(c). Then, the robot starts moving to the study(d). In the study, a different user takes the coffee cup from the robot, and released the robot by saying "Thank you."(e). Then the robot goes back to the bedroom(f, g). In the experiment, the proposed recognition system recognized the command utterances, even when the TV is on as a noise sound source.
5. CONCLUSIONS AND FUTURE WORKS

The paper reported an online voice command recognition system using a microphone array. DSBF and FBS method is used for multiple sound localization, and separation. The method is implemented to the 32ch microphone array attached to the ceiling. The separated sound sources are recognized using an open source recognizer “Julian” with a word limited command dictionary, then recognized command with sound position and time information are sent to the mobile robot.

The system localizes sound source position in azimuth and elevation, and recognizes separated sound sources simultaneously. The system can recognize arbitrary timed voice commands in noisy environments using a 32 channel microphone array unit, and the system outputs the recognized command with sound position information. The system is tested on a compact microphone array unit and can be applied to robot embedded microphone array. By using DSBF and FBS method for multiple sound localization and separation, the microphone array system can localize sound source position in azimuth and elevation, with 10(deg) angle error on average. The proposed system works without environmental information such as impulse response, and the evaluation result in two different reverberation conditions shows similar performance.

We defined four indexes to evaluate the performance of the proposed auditory system. The experimental results show the robustness for multiple sound sources or distant sound sources. Target separation rate is more than 95(%) on average, and task achievement rate is more than 86(%) on average for two sound sources in the "Holone" experiments. And as shown in section IV-C, the task achievement rate is more than 70(%) within a 10(m) radius with a noise sound source at near range.

In this paper, word dictionary is limited simple command for mobile robot to prevent error recognition. Recognizer could works with a handled size of word dictionary for separated sound sources when input sound sources are only human voice included in dictionary. On the other hand, error recognition rate becomes high, when unknown sound sources exists. Future research is needed to detect human voice and reduce error recognition for non-modeled sound sources.

References