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Technical Note

A Design for a Collaborative Steering System of Microphone Array and Video Camera Toward Multi-Lingual Tele-Conference

Takanobu Nishiura,†,☆ Rainer Gruhn† and Satoshi Nakamura†

It is very important for multi-lingual tele-conferencing through speech-to-speech translation to capture distant-talking speech with high quality. In addition, the speaker image is also needed to realize a natural communication in a such conference. A microphone array is an ideal candidate for capturing distant-talking speech. Uttered speech can be enhanced and speaker images can be captured by steering a microphone array and a video camera in the speaker direction. However, to realize automatic steering, it is necessary to localize the talker. To overcome this problem, we propose collaborative steering of the microphone array and the video camera in real-time for a multi-lingual tele-conference through speech-to-speech translation. We conducted experiments in a real room environment. The speaker localization rate (i.e., speaker image capturing rate) was 97.7%, speech recognition rate was 90.0%, and TOEIC score was 530 – 540 points, subject to locating the speaker at a 2.0 meter distance from the microphone array.

1. Introduction

To achieve multi-lingual tele-conferencing through speech-to-speech translation, the high-quality sound capture of distant-talking speech is very important. However, background noise and room reverberations seriously degrade the sound capture quality in real acoustical environments. A microphone array is an ideal candidate as an effective method for capturing distant-talking speech. With the microphone array, the desired speech signal can be selectively acquired by precisely steering the directivity in the desired speech direction. Thus, it is also necessary to localize the speaker direction to realize high-quality sound capture of distant-talking speech.

Therefore, we use the delay-and-sum beamformer [1),2)] to realize robust high-quality sound capture of distant-talking speech in various environments and achieve real-time processing after estimating the speaker direction by the CSP (Cross-power Spectrum Phase analysis) coefficient addition method [3)] based on CSP method [4)], and a video camera is also automatically steered to the estimated speaker direction. By using the above methods, the speaker speech and image can be captured robustly and accurately. Next, we translate beamformed speech using ATR's Multi-lingual Automatic Translation System for Information Exchange (ATR-MATRIX) [5)]. It is an ideal candidate as an effective tool for translating speech-to-speech.

We propose a system with collaborative steering of the microphone array and the video camera as a step toward achieving multi-lingual tele-conferencing through speech-to-speech translation. **Figure 1** shows the setup of this system.

2. Key Technology for the System

Figure 2 shows an overview of the proposed system. In this system, a video camera and a microphone array are first automatically steered to the DOA (Direction Of Arrival) estimated by the CSP coefficient addition method after capturing speech with a microphone array. Next, speech beamformed by the steering directivity of the microphone array is translated and synthesized using ATR-MATRIX for multi-lingual tele-conferencing. The speaker image is captured by a video camera and shown at the same time. A natural multi-lingual tele-conference can be realized with this system. **Figure 3** shows the microphone array and the video camera used in the proposed system. In the following sections, we explain the key technologies for the proposed system in detail.

2.1 DOA Estimation by CSP Coefficient Addition Method

DOA must be estimated to collaboratively steer the microphone array and the video camera. Thus, we use the CSP coefficient addition method [3)] to estimate DOA. In the environment

† ATR Spoken Language Translation Research Laboratories
☆ Presently with Faculty of Systems Engineering, Wakayama University

Fig. 1 Speech-to-speech translation system of distant-talking speech.

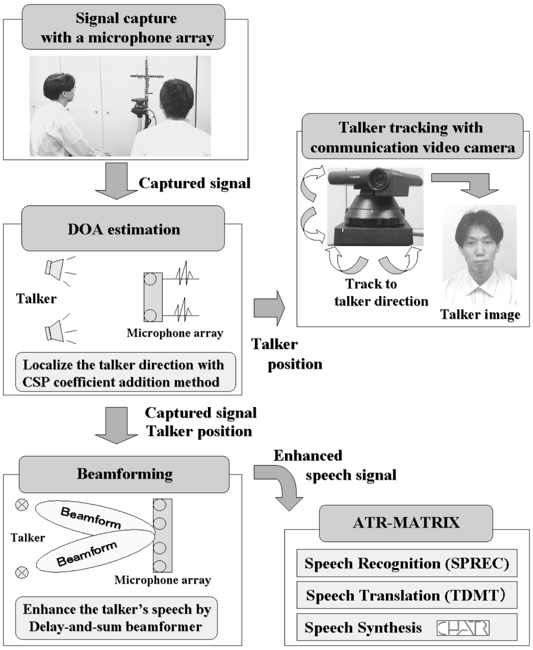

Fig. 2 Proposed system overview.

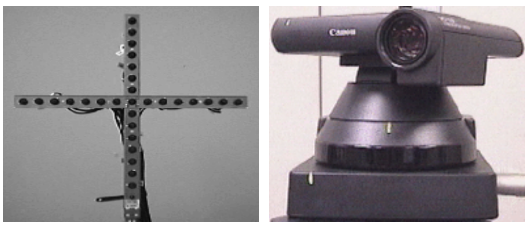

Fig. 3 Microphone array and video camera.

of **Fig. 4**, the CSP coefficients are derived from Eq. (1).

$$\mathrm{CSP}_{i_n,j_n}(k) = \mathrm{IDFT}\left[\frac{\mathrm{DFT}[s_{i_n}(t)]\mathrm{DFT}[s_{j_n}(t)]^*}{|\mathrm{DFT}[s_{i_n}(t)]|\,|\mathrm{DFT}[s_{j_n}(t)]|}\right], \quad (1)$$

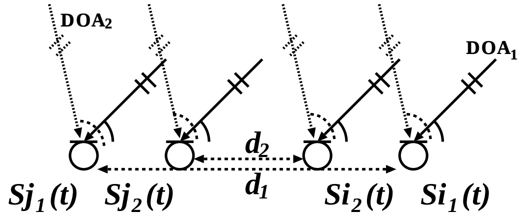

Fig. 4 DOA estimation with CSP coefficient addition method.

where t and k are the time index, DFT $[\cdot]$ (or IDFT$[\cdot]$) is the discrete Fourier transform (or the inverse discrete Fourier transform), and the symbol * is the complex conjugate. Then, CSP coefficients are added as shown in Eq. (2).

$$\mathrm{CSP}_{i,j}(\theta) = \sum_{n=1}^{N} \mathrm{CSP}_{i_n,j_n}(\theta),$$

$$\textit{subject to} \quad \theta = \cos^{-1}\left(\frac{c \cdot k/F_s}{d_n}\right), \quad (2)$$

where N is the number of additions, d_n is the distance between two adjacent transducers, c is the sound propagation speed, and F_s is the sampling frequency. The DOA can be estimated by finding the maximum values of the added CSP coefficients by Eq. (3).

$$\mathrm{DOA_n} = \underset{\theta}{\mathrm{argmax}}(\mathrm{CSP}_{ij}(\theta)). \quad (3)$$

The CSP coefficient addition method can estimate multiple DOAs. However, as this system has only one video camera, we selected the desired DOA based on signal energy of estimated DOAs. The CSP coefficient addition method is suitable for real-time processing because it can accurately estimate DOAs by simple calculation.

2.2 Automatic Video Camera Steering for Capturing Speaker Image

The video camera is automatically steered to the DOA estimated by the CSP coefficient addition method in order to automatically capture the speaker image and to facilitate multilingual tele-conferencing. In this system, the speaker image is automatically captured with a video camera as shown in Fig. 3. It can move not only in the horizontal direction but also in the vertical direction. Video camera steering is controlled through an RS232C port by a server computer. The video image is shown immediately on a monitor and translated speech is emitted from a loud speaker.

2.3 Microphone Array Steering for Speech Enhancement

Beamforming is necessary to capture distant-talking speech at high quality with a microphone array. In this paper, a delay-and-sum beamformer [1),2)] is used to form the directivity to the estimated DOA. the delay-and-sum beamformer can emphasize the desired sound signal from the estimated DOA because the sound signals captured with multiple transducers are added after synchronizing them.

2.4 Speech-to-Speech Translation with ATR-MATRIX

ATR-MATRIX [5)] consists of a speech recognition sub-system (ATR-SPREC), a language translation sub-system (TDMT), and a speech synthesis sub-system (CHATR) [6)]. The current implementation of our system deals with a hotel room reservation task/domain. The speech recognition sub-system recognizes speech that is beamformed with the microphone array. Then, the language translation sub-system translates the recognized speech. Finally, the translated speech is synthesized by CHATR.

2.4.1 Speech Recognition with ATR-SPREC

The speech recognizer module was built based on ATR-SPREC, a speech recognition software toolkit developed at ATR. ATR-SPREC has the following settings:

- Acoustic model: Shared-state context-dependent (triphone) HMMs produced by the ML-SSS algorithm.
- Language model: Multi-class composite N-gram.
- Search engine: A decoder featuring multi-pass search and word graph output.

2.4.2 Language Translation with TDMT

The language translator module uses Transfer Driven Machine Translation (TDMT) technology and can deal with various expressions in spoken languages because it uses not only sentence structures but also translation examples. The basic mechanisms of TDMT are as follows:

- Extraction of partial linguistic structures (patterns) from an input sentence.
- Example-based and pattern-by-pattern transfer to a target language.
- A search for the most likely combination of transferred patterns.

2.4.3 Speech Synthesis with CHATR

Speech synthesis is essential for realistic multi-lingual teleconferencing through speech-to-speech translation. CHATR [6)] produces natural synthetic speech by selecting and re-sequencing wave units from a CHATR-specific speech database. We used CHATR as a speech synthesizer for realizing the proposed system. Since the current configuration of our system has male and female acoustic modules, the CHATR speech synthesis sub-system can output either male or female voices.

Table 1 System components.

AD converter	Thinknet DF4448
Microphone	Hoshiden KUC1333
Microphone array	Onkyo Sokki OMA520 29 transducers, horizontal:15, vertical:15 2.125 cm spacing
Microphone amplifier	Thinknet MA2016
Server computer	COMPAQ XP-1000 × 2 CPU: 500 MHz, Memory: 512 MB
OS	Tru64 UNIX V5.0A
Video camera	CANON VC-C3

Table 2 System conditions.

AD conversion	
Sampling frequency	16 kHz
Quantization	16 bit
DOA estimation and Beamforming	
Frame length	128 msec. (shift: 64 msec.)
Window	Hamming window
ATR-MATRIX	
Frame length	25 msec. (shift: 10 msec.)
Feature vector	MFCC 12 orders, ΔMFCC 12 orders, Δlog-power 1 order
Window	Hamming window
Video camera	
CCD pixel size	1/4 inch
Moving performance	Pan: angle: 180° speed: 1°–76°/s, Tilt: angle: 55° speed: 1°–70°/s

3. System Specifications

Tables 1 and **2** show the proposed system's specifications. This system uses two workstations (server computers). One is for multi-channel signal capture, DOA estimation, beamforming, and video camera steering. The other is for speech-to-speech translation. The two computers are connected by a LAN (Local Area Network) and communicate with each other by socket protocol. If we conduct multi-lingual tele-conferencing, two sets of this system will be needed. Although the video camera can move in both horizontal and vertical directions, movement in the vertical direction is slower

Table 3 System performances.

DOA estimation rate	97.8%
Speech recognition rate	90.0% (91.4%)
TOEIC score	530 $\sim$ 540 pts. (546 pts.)

( ) shows performance with closed talking microphone.

than that in the horizontal direction because of the video camera performance. An AD converter is connected to the server computer by SCSI, and the video camera is connected through an RS232C port.

4. System performance

The proposed system was evaluated in an acoustic experimental room. Reverberation time of this room is $T_{[60]} = 0.27$ seconds and ambient noise level is 24.3 dBA. Two speakers engage in mutual talk to as in a tele-conference using one system. Also, we evaluated the proposed system by assuming that one speaker speaks Japanese and the other speaker listens in English because we can only realize Japanese to English translation at this time. Two speakers are located at positions along $30°, 60°, 90°, 120°$, and $150°$ directions and 2 meters distance from the microphone array as shown in Fig. 1. **Table 3** shows our experimental results. These results are averaged from 4 subjects (1 female and 3 males). A hotel room reservation task consisting of 42 dialogues was used as test data. With the proposed system, DOA estimation rate (i.e., speaker image capturing rate) was 97.8%, and speech recognition rate was 90.0%, compared to 91.4% with a closed talking microphone. We also evaluated speech translation performance with TOEIC score according to reference [7]. TOEIC score can be calculated with the translation paired comparison method which is proposed by F. Sugaya in 2000. As a result, we confirmed that the proposed system may achieve about 530–540 points while the system with a closed talking microphone is 546 points. Next, we investigated the system response speed. As a result, we confirmed that DOA can be estimated within 0.192 seconds, the video camera is steered automatically with about 0.2 seconds delay after capturing speech, and beamforming still takes more than about 0.064 seconds after estimating DOA. Thus, we can conclude that it will take about 0.256 seconds delay after capturing speech to ATR-MATRIX. We could confirm that the proposed system achieves high speech-to-speech translation performance, although it is slightly less effective than a system with a closed talking microphone. We could also confirm a system response speed of within 0.256 seconds for the collaborative steering of the microphone array and video camera.

5. Conclusions

In this paper, we proposed collaborative steering of a microphone array and video camera as a step toward achieving multi-lingual tele-conferencing through speech-to-speech translation. And, we realized a system that can process in real-time. In the future, we will have to consider barge-in and high quality capture of speech in noisy reverberant environments.

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