Speech Separation in the Vehicle Environment Based on FastICA Algorithm

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Abstract—The speech interaction in-vehicle was mainly realized by the speech recognition. The human-machine interaction around was usually disturbed by the noise, and the speech received by the receiver was not the original pure speech, so compared to the pure environment, the accuracy of the speech recognition declined so sharply that it could not meet the demand of the practical application of human-machine interaction. So the speech recognition was required to have the strong adaptability and processing capacity to the speech with noise. In this paper, the FastICA algorithm in signal process and statistics was studied and used to separate the driver’s speech in the vehicle environment, and realizes the pretreatment in recognizing driver’s speech. The effectiveness of the method had been validated in actual vehicle experimental configuration.

Index Terms—integrated voice-data communication, intelligent control, speech processing, human-machine interfaces, fast independent component analysis (Fast ICA)

I. INTRODUCTION

With the development of science and technology and the higher requirement of vehicle performance, vehicles had more and more devices, and the human-machine interaction became more and more important. Considering the driving safety, the drivers had to focus on the road status and keep their hands on the steering wheel. If drivers could control the GPS navigation system through the speech, such as setting destinations, searching places, and controlling et al., they could avoid using their hands and eyes, and this had very important meanings for freeing the drivers’ hands. Using speech as a means of human-machine interaction could avoid their dependence on hands, and it also had very important practical meanings on convenience and safety.

The most significant advantage of the speech communication interface was to keep the drivers’ hands on the steering wheel and eyes on the road. To realize the speech recognition technology, it needed very high performance processor and software. The condition was that the hardware already existed, but the software needed to be improved. Many companies such as Lernout & Hauspie (L&H), Nuance, IBM, Dragon and Speech Works had already developed speech recognition software which could run on various platforms [1]. The key technologies of the speech recognition system were creating the command vocabulary and training the vocabulary. And it required high recognition accuracy. When the vocabulary was large or there were more users the accuracy would decline. In fact in vehicle, the vocabulary was small and the user was only the driver, so the emphasis of the speech recognition system was to improve the performance of the system through improving the key technology.

Delphi Automotive Systems had provided hand-free speech recognition system since 1996, and it also provided telematics series products cooperated with BMW. The navigation service such as Jaguar ASSIST, Wingcast and Wireless-Car provided by Ford Company was used in Explorer, Mercury Sable and Jaguar Sedan [2]. The Sarah Auto PC, which showed in Paris Motor Show by the Citroen in 1998, starts a new situation in vehicle communication technology [3]. It based on Microsoft WINCE2.0 operating system, controlled by speech, reacted to driver’s voice command and dialogize with handlers through speech synthesis system. In the same year, the General Motors Company vigorously developed its Onstar electronic system, and made it a speech recognition system from the basic lane driving assistant system. In 2000, the General Motors Company produced the first speech controlling wireless internet automobile. Simultaneously, some limousine such as “Jaguar S Series” installed the simple speech control system, through which the driver could regulate the air-conditioning and sound by speech instruction. In 2001, the Audi AG showed the Multi Media Interface(MMI) control technology in Frankfurt Motor Show, which integrated the electronic system in the car and realized the speech control. In 2003, Dr. Sapphire Company developed the vehicle audio multimedia, which controlled telephone, sound and navigation system by speech recognition. In the same year, the Mercedes-Benz’s new generation vehicle type of E-Class used speech recognition control technology to control the navigation system and the vehicle phone. In 2004, the scientists in
The speech interaction in-vehicle was mainly realized by speech recognition. The speech recognition was a technology to implement corresponding control through recognizing speaker’s voice or identification to correctly judge the speech connotation. It was the key technology of using speech in human-machine interaction. In recent years, it was used in many fields. Depending on speech recognition hardware and tool software to recognize speaker’s voice, it controlled corresponding actuators. At present, the speech recognition in laboratory quiet environment was very mature, and the recognition accuracy could reach 90%, which could satisfy the normal application. These systems were large vocabulary continuous speech recognition system such as Via Voice of IBM, Whisper of Microsoft and SPHINX.H of CMU [6]. But the coming multimedia age urgently required the speech recognition system to use practically, and this required the speech recognition to have the strong adaptability and processing capacity to the speech with noise.

Because the human-machine speech interaction in-vehicle disturb by the noise in environment, the received speech was already not the original pure speech. Compared to the pure environment, the recognition accuracy decreased sharply, and could not meet the demand of practical human-machine interaction. The difficulty of the speech recognition was how to solve the noise. Now there was much study about noise eliminating. The common methods could be summarized in four aspects: spectral subtraction, environment structured technology, correcting recognizer model (not correcting speech signal) to adapt noise, and establishing noise model. But these methods could not eliminate the noise influence completely, and they needed further improvement. This kind of speech recognition system in the paper distinguished the specific speaker, and was often very accurate. It demanded user to train the speech recognition system to get the recognition result. Realizing noise eliminating and suppression, enhancing the speech were the basic and key contents for improving speech recognition accuracy. After the front end processing on the speech inside vehicle with noise, such as noise eliminating and speech enhancing, the speech as pure as possible was got, and then the speech recognition in noise environment was done by using the mature speech recognition technology [7].

In 1990, Bregman proposed the computer separation method which aimed at separating speech signal in his published book [8]. In 1995, after the founding of the first Computational Auditory Scene Analysis (CASA) study team, the computer speech separation aiming at speech became a hot research. In the same year, A. J. Bell and T. J. Sejnowski published the milestone literature in ICA theory and technology developing. Then scholars began to use ICA technology to separate the synthetic speech signal with noise, and ICA technology began to use in multi-channel speech signal separation [9]. But ICA separation algorithm existed limitations in actual environment. In October, 2003 at Montreal in Canada, a speech separation prospect study and discussion team initiated and composed by National Natural Science Foundation, and other fund committee also participated. The team communicated with all speech study fields and focused on discussing the future of the speech separation. In November 2004, at Quebec in Montreal, a study and discussion team about speech separation and understanding in complex speech environment initiated by Air Armed Forces Natural Science Research Association and National Natural Science Foundation [10]. But ICA algorithms still have problems in the following aspects.

a) the estimation of the number of blind source signal. At present the speech blind separation algorithm assumes that the number of the observed mixture signals is equal to or greater than the number of the source signals. For unknown source signals and the number of the observed signals less than the source signals, it is difficult to analyze and needs further study.

b) The space location of the source signals. When the sensors’ position of the source signals and observed signals are not in the same plane but have the spatial relationship, there will be energy masking and information masking, and then the space location only can be done by the limited information in the observed signals.

Therefore, studying and proposing the front end processing technology in the driving environment was very important. Based on the FastICA algorithm in the signal processing and statistics, the paper used the method to separate the deriver’s speech, and realized the front end processing to recognize the driver’s speech.

The remaining parts of this paper were organized as follows: Section II described the FastICA algorithm. Section III described the configuration of experiments in the actual vehicle environment, analyzed the experimental results, and the conclusions were given in Section IV.

II. ICA AND FASTICA ALGORITHM

As we know that independent component analysis (ICA) was a popular technology for solving the BSS problems. Paper [11] addressed some of the first works to
apply ICA in symbol demodulation. ICA relied on higher-order statistics which were typically the fourth-order statistic kurtosis to solve the BSS problems. Papers [12] and [13] addressed the issue of delay estimation with ICA. Jammer mitigation with a BSS principle was first discussed in [14], where an ICA method called JADE and second-order methods were used to mitigate a temporally correlated jammer. The paper [15] also addressed the issue of blind beam forming to jammer mitigation. These methods were very popular in other domains, and they were attracted attention in the field of communication engineering. Some person thought the reason was that some of these methods had an artificial neural network background.

In the vehicle environment we introduced ICA technology into speech separation. Combining an ICA element to standard techniques enabled a robust and computationally efficient structure. In the paper, we introduced a switching techniques based on BSS/ICA effectively to combat interference [14]. BSS was an idea of signal processing where mixtures of several sources were separated without the knowledge of the mixing processes. Several schemes existed for interference suppression based on a similar principle. Considering the following linear model, the popular framework for solving the BSS problem was ICA.

\[
\text{Morig} = \text{AO} + \text{v} \quad (1)
\]

Usually, we must make the fundamental restrictions before the mixed-signal separation. First, the components of \( O \) are statistically independent. Second, at most one component of \( O \) is Gaussian distributed. Third, the mixing matrix \( A \) is full rank. In the model the linear mapping was called the mixing matrix. The model assumed some noise \( v \) considered to be Gaussian. Solution to the linear source separation problem was not possible, if there was no information on some of the variables \( A \) or \( O \), in addition to the observed data \( \text{Morig} \).

\[
\begin{bmatrix}
\text{M}_{\text{orig},1} \\
\text{M}_{\text{orig},2} \\
\vdots \\
\text{M}_{\text{orig},K}
\end{bmatrix}, \quad
\begin{bmatrix}
\text{O}_1 \\
\text{O}_2 \\
\vdots \\
\text{O}_N
\end{bmatrix}, \quad
\text{v} = \begin{bmatrix}
\text{V}_1 \\
\text{V}_2 \\
\vdots \\
\text{V}_K
\end{bmatrix}
\quad (2)
\]

\( \text{Oi} = [\text{Oi}(1) \ldots \text{Oi}(t)] \) (3)

The symbol \( t \) represented time, but could represent some other variable, e.g., space. The model consisted of \( N \) sources of \( t \) samples, i.e.

\[\text{Morig,}i = [\text{Mi}(1) \ldots \text{Mi}(t)] \quad (4)\]

Usually, the observations \( \text{Morig} \) consisted of \( K \) mixtures of the sources. And it was assumed that there were at least as many observations as sources i.e., \( K \times N \).

\[
A = [a_1, a_2 \ldots a_N] \quad (5)
\]

If the mixing matrix \( A \) was known and the noise was negligible, the sources could be estimated by finding the matrix \( B \), the inverse of the mixing matrix \( A \). The full-rank assumption was the necessary and sufficient condition for the existence of the pseudo-inverse of \( A \).

\[
\text{BMorig} = \text{BAO} = \text{O} \quad (6)
\]

If there were as many observations as sources, then \( A \) was square and has full-rank. So that, \( B = A^{-1} \). When there were more observations than sources, the existed several matrices \( B \) that satisfy the condition \( BA = I \). In this case, the choice of \( B \) depended on the components of \( O \) which we were interested in. For cases where there were less numbers of observations than sources, a solution did not exist unless further assumptions were made. Now the rank of \( A \) was less than the number of sources. There were some redundancies in the mixing matrix, and hence further information was required.

The algorithm could achieve optimization by regulating separation matrix which used the stochastic gradient algorithms. On the assumption that the independent component analysis data model satisfied the restrictions, the convergence rate of FastICA was at least quadratic. With the help of nonlinear functions, FastICA finds the independent components of non-Gaussian distribution directly. The characteristic of FastICA ensured that it could achieve optimization by a suitable nonlinear function; especially it could gain the algorithm which was robustness.

We processed BSS for the system using FastICA algorithm in the paper as figure 1, the detailed execution was as follows.

1. Initializes \( S(0) \)
2. Calculates \( S(k) = C^{-1}E \{ M(S(k-1)^{3}M) \} - 3S(k-1) \)
3. Divides \( S(k) \)
4. Outputs \( S(k) \)
5. Calculates \( S(k) = C^{-1}E \{ M(S(k-1)^{3}M) \} - 3S(k-1) \)
6. If \( |S(k)TS(k-1)| \) don't approach 1 closely, then, makes \( k = k + 1 \), returns to step 2. Otherwise, outputs \( S(k) \).

Figure 1. The process of algorithm
This speech signal from the composite signal. Separating a speech signal each time, we must remove the iterative procession would terminates. Besides, after were equal to each other or have small difference, then separated had the unit energy. If two neighboring $S_j(k)$ separated. After each iterations we must carry on processing to $S_j(k)$ the columns of the mixed matrix $O$, added a projection operation. Then divided $S_j(k)$ with $||S_j(k)||$. The initial random vectors at the beginning also run the project before the implementation of recursion. In order to avoid deterioration of estimated $S_j(k)$, the projection could be cancelled after a certain number of iterations. Therefore, we combined with the FastICA algorithm, optimized the separation matrix $W$ by the stochastic gradient algorithm. This algorithm processed the iterations by the determinations of the greatest negative entropy, gained the following formula:

$$S_j(k)=S_j(k)-OO^T S_j(k)$$

(7)

We projected the value of current $S_j(k)$ to the columns of the mixed matrix $O$, added a projection operation. Then divided $S_j(k)$ with $||S_j(k)||$. The initial random vectors at the beginning also run the project before the implementation of recursion. In order to avoid deterioration of estimated $S_j(k)$, the projection could be cancelled after a certain number of iterations. Therefore, we combined with the FastICA algorithm, optimized the separation matrix $W$ by the stochastic gradient algorithm. This algorithm processed the iterations by the determinations of the greatest negative entropy, gained the following formula:

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(7)

Where, $S_j(k)$ was the row vector corresponding to the $j$th speech signal within matrix $S$ that iterated $k$ times. Continue the iteration like this, until all signals were separated. After each iterations we must carry on processing to $S_j(k)$, to ensure the results which been separated had the unit energy. If two neighboring $S_j(k)$s were equal to each other or have small difference, then the iterative procession would terminates. Besides, after separating a speech signal each time, we must remove this speech signal from the composite signal.

III. EXPERIMENTAL RESULTS

To handle the noise disturbing to the speech recognition system in vehicle environment, we needed the sound data in vehicle environment to provide the basis for the subsequent speech recognition. According to the vehicle structure and working characteristics, we analyzed the noise sources and made the concrete experiment environment under the new automobile industry national standard <GB-T18697-2002 Acoustics-Measurement of noise inside motor vehicles>.

In this experiment, the distance between the vehicle and large objects was longer than 20m, so the vehicle sound radiated could only become part of the inside vehicle noise by the reflection of the road but not the buildings, walls and other similar large objects. The temperature must between -50℃ and +350℃. The wind speed along the measurement road at the height of 1.2m must never exceed 5m/s. And other meteorological conditions mustn’t influence the measurement result. The noise in-vehicle was greatly influenced by the roughness of the road surface condition, and the flat road surface could generate smooth noise inside the vehicle [16]. So in the experiment, the selected road was flat asphalt road without joints and unevenness. The road surface was dry and had no sundries such as snow, dirt, stones and leaves. Then there was no way to increase the sound level inside vehicle. The vehicle was no-load in the experiment while collecting noise, there was no other load except the deriver, testers and test devices. In the experiment, the skylight, all the windows, air inlets, air outlets and assist devices such as windshield wiper, heater devices, fans, and air conditioning were closed as they were not the main sources of the noise inside vehicle. The adjustable seat should adjust to the middle of the horizontality and vertical. The back of the seat should be vertical. And the adjustable headrest should be in the middle. The specific experiment environment settings list in Table I.

<table>
<thead>
<tr>
<th>Measurement Date:</th>
<th>July 16, 2008</th>
</tr>
</thead>
<tbody>
<tr>
<td>Weather:</td>
<td>Cloudy Day</td>
</tr>
<tr>
<td>Measurement Place:</td>
<td>high-tech zone, Changchun, China</td>
</tr>
<tr>
<td>Temperature(℃):</td>
<td>22 celsius degree</td>
</tr>
<tr>
<td>Road Situation:</td>
<td>Flat asphalt road</td>
</tr>
<tr>
<td>Wind Speed(m/s):</td>
<td>Southerly, level 1.0, 0.5m/s</td>
</tr>
<tr>
<td>Vehicle Style:</td>
<td>BORA1.6, M1</td>
</tr>
<tr>
<td>Manufacture Date:</td>
<td>1985-10</td>
</tr>
<tr>
<td>Engine Type:</td>
<td>4 Cylinder</td>
</tr>
<tr>
<td>Driven mileage(km):</td>
<td>33000</td>
</tr>
<tr>
<td>Type:</td>
<td>BJH026729</td>
</tr>
<tr>
<td>Rated Passengers No or Maximum Total Mass (kg):</td>
<td>5passengers/1830</td>
</tr>
<tr>
<td>Rated Power Capability (kw):</td>
<td>74</td>
</tr>
<tr>
<td>Rated Speed (r/min):</td>
<td>2500</td>
</tr>
<tr>
<td>Sensitivity Range:</td>
<td>-26±3dB, RL= 2.2KΩ</td>
</tr>
<tr>
<td>Frequency:</td>
<td>50-16000 Hz</td>
</tr>
<tr>
<td>Operation Voltage Range:</td>
<td>2.2V-5.0V (DC)</td>
</tr>
<tr>
<td>Max. Sound Pressure Level:</td>
<td>115dB S.P.L</td>
</tr>
<tr>
<td>S/N Ratio:</td>
<td>More than 58dB 1kHz, 0dB=1V/Pa, A-weight</td>
</tr>
<tr>
<td>Sensitivity Reduction:</td>
<td>3.0V-2.2V, Sensitivity Variation less than 3dB</td>
</tr>
<tr>
<td>Sound level Meter Style:</td>
<td>TDJ824</td>
</tr>
<tr>
<td>Accuracy Level:</td>
<td>±1.5dB 94dB@1kHz</td>
</tr>
<tr>
<td>Verification Effective Date:</td>
<td>July 1, 2008</td>
</tr>
<tr>
<td>Two Laptops:</td>
<td>IBM R60</td>
</tr>
<tr>
<td>Recording Software:</td>
<td>Adobe Audition 1.5</td>
</tr>
</tbody>
</table>

At present in the blind source speech separation algorithm, it was assumed that the number of the mixture signals was equal to or more than the source signals. As we only needed to separate the driver’s speech, all the sounds except the driver’s were noise. And two microphones could meet the demand of using ICA algorithm. For the speech interface was mainly used to receive the driver’s speech, the noise inside vehicle related to the position of the driver. All the microphones used in the noise measurement experiment must install in a certain form. Under the microphone, a thin sponge pad (about 8mm) must be laid to make sure that the vehicle vibration would not influence the microphone. The installation of the microphones should be so tight that they could not make a relative motion for the vehicle. A relative motion meant having amplitude about 20mm.
The vertical distance from microphone to its fixed position on vehicle must be longer than 0.15m. The two microphones must point to the driver’s mouth in their most sensitive direction [17]. One place of the microphones was the dashboard in front of the driver, and the other was sun visor. The vertical coordinate of the microphone was $0.70 \pm 0.05$ m above the intersection of the seat surface without driver and the backrest surface. The horizontal coordinate of the microphone was the center plane of the seat. In the driver’s seat, the left distance between the horizontal coordinate and the seat center plane was $0.20 \pm 0.02$ m. The position of the microphones related to the seat and the practical position of the microphones show in figure 2 and figure 3.

In the experiment, according to the specific speech command, a section of words was designed which completely composed of speech commands. Then microphone A recorded a driver’s speech commands. Then a selected pure music played in the CD player. Microphone A recorded the CD music. Then microphone A and B recorded the mixture sound at the uniform speed 40km/h while the CD played and the driver spoke. The mixture sound was composed of driver’s speech, CD music and the noise inside vehicle. The content of the driver’s speech was door, window, wiper, CD, air conditioning, direction indicator lamp, and fog light. The music in the CD was the High Mountain and Running River. The sound pressure level in the vehicle was 62.5dB with background music, and 69.6dB with driver’s command, and 67.6dB with both. The collected sound stored in the format WAV. And the file length was 4.0s. The figure 4 and figure 5 were two mixture sounds of driver’s speech and CD music. The figure 6 and figure 7 were the separated driver’s speech and CD music by FastICA.
In the four figures above, they showed that the time domain of the signals did not change before and after separation. But the amplitude of the signals increased after separation by the FastICA algorithm. Through listening, it was found that the volume of the speech significantly increases. The spectrograms before and after separation were showed in figure 8, 9, 10 and 11.
In the four figures, they clearly showed that the FastICA algorithm not only reserved most speech signal energy, but also eliminated the background noise effectively. In figure 10, the reserved sound was driver’s speech and the eliminated sounds were noise inside vehicle and CD music. In figure 11, the reserved sound was CD music and the eliminated sounds were noise inside the vehicle and the driver’s speech. The spectrogram after separation was more similar to the pure original one.

Subject test method was used to test the separated driver’s speech and CD music in the experiment. And the evaluation standard was mean opinion score which was widely accepted to test the subject acceptance level [18]. As the driver’s speech and the CD music recorded while the vehicle was running, the MOS result of the original driver’s speech was 4.39 and the original CD music was 4.51. So the FastICA algorithm was better. The concrete items were in Table II.

### Table II. The Test Result of MOS

<table>
<thead>
<tr>
<th>Evaluation Group</th>
<th>The original driver’s speech by FastICA</th>
<th>The separated driver’s speech by FastICA</th>
<th>The original CD music by FastICA</th>
<th>The separated CD music by FastICA</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>4.50</td>
<td>3.50</td>
<td>4.60</td>
<td>3.90</td>
</tr>
<tr>
<td>2</td>
<td>4.30</td>
<td>3.60</td>
<td>4.50</td>
<td>3.80</td>
</tr>
<tr>
<td>3</td>
<td>4.30</td>
<td>3.50</td>
<td>4.50</td>
<td>3.80</td>
</tr>
<tr>
<td>4</td>
<td>4.20</td>
<td>3.30</td>
<td>4.30</td>
<td>3.70</td>
</tr>
<tr>
<td>5</td>
<td>4.30</td>
<td>3.20</td>
<td>4.50</td>
<td>3.80</td>
</tr>
<tr>
<td>6</td>
<td>4.30</td>
<td>3.20</td>
<td>4.50</td>
<td>3.60</td>
</tr>
<tr>
<td>7</td>
<td>4.50</td>
<td>3.50</td>
<td>4.50</td>
<td>3.50</td>
</tr>
<tr>
<td>8</td>
<td>4.70</td>
<td>3.50</td>
<td>4.50</td>
<td>3.50</td>
</tr>
<tr>
<td>9</td>
<td>4.20</td>
<td>3.50</td>
<td>4.50</td>
<td>3.50</td>
</tr>
<tr>
<td>10</td>
<td>4.60</td>
<td>3.50</td>
<td>4.60</td>
<td>3.50</td>
</tr>
<tr>
<td>Mean</td>
<td>4.39</td>
<td>3.43</td>
<td>4.51</td>
<td>3.58</td>
</tr>
</tbody>
</table>

Figure 12. The numerical analysis diagram of the MOS test result

In the MOS test of the experiment, 10 people who never learned BSS or studied the speech signal process were selected as listeners. Then they listened to the original driver’s speech, the original CD music, the separated driver’s speech, and the separated CD music. After listening, they scored the test sound by their satisfaction. And the satisfaction level was divided into 5. At last, mean opinion score of every test sound was calculated. Table II was the specific data of the MOS test result. And figure 12 was the numerical analysis diagram of the MOS test result.

In the experiment, it showed that the effect of the separated driver’s speech and CD music by the FastICA was only understood and barely accepted. It was satisfied in mood and rhythm but not the degree. The score of the original driver’s speech and CD music were 4.39 and 4.51, not 5 mainly because the sound were collected when the vehicle was running, which means there were bigger noise interference with the recorded sound.

### IV. Conclusion

In the paper, the ICA method in the statistics and signal process were applied to the vehicle practical driving environment innovatively. And the method separated the driver’s speech successfully and it was the front end process for further recognition of the driver’s speech effectively. The reasonable real BORA vehicle experiment validated the performance of the method.

Compared the spectrograms and the Spectrum language before and after separation, and tested by MOS, the FastICA could obtain a better effect in practical vehicle speech separation algorithms. And the method could be the front end process for further recognition.

The next work was to construct a whole high recognition accuracy vehicle speech interface system based on the method in the paper, and the system could meet the demand of the practical application.

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