



Improving Channel Decorrelation for Multi-Channel Target Speech Extraction

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Abstract

Target speech extraction has attracted widespread attention. When microphone arrays are available, the additional spatial information can be helpful in extracting the target speech. We have recently proposed a channel decorrelation (CD) mechanism to extract the inter-channel differential information to enhance the reference channel encoder representation. Although the proposed mechanism has shown promising results for extracting the target speech from mixtures, the extraction performance is still limited by the nature of the original decorrelation theory. In this paper, we propose two methods to broaden the horizon of the original channel decorrelation, by replacing the original softmax-based inter-channel similarity between encoder representations, using an unrolled probability and a normalized cosine-based similarity at the dimensional-level. Moreover, new combination strategies of the CD-based spatial information and target speaker adaptation of parallel encoder outputs are also investigated. Experiments on the reverberant WSJ0 2-mix show that the improved CD can result in more discriminative differential information and the new adaptation strategy is also very effective to improve the target speech extraction.

Index Terms: Multi-channel, target speech extraction, channel decorrelation, scaling adaptation

1. Introduction

Target speech extraction (TSE) aims to extract only a target signal from the mixed speech [1]. Generally, most TSE approaches require additional target speaker clues to drive the network towards extracting the speech of that speaker. Many previous works have focused on the TSE tasks [2–6]. Although these works have achieved great success on close-talk target speech extraction tasks, the performance of far-field speech extraction is still far from satisfactory due to the reverberation, noise, etc. In that case, the additional multi-channel spatial information usually can be used to improve the separation ability.

Many contributions have been paid to exploit the spatial information between multi-channel recordings for TSE. For example, the direction aware SpeakerBeam [7] combines an attention mechanism with beamforming to enhance the signal of the target direction; the neural spatial filter [8] uses the directional information of the target speaker to extract the corresponding speech; the time-domain SpeakerBeam (TD-SpeakerBeam) [9] incorporates the inter-microphone phase difference (IPD) [10] as additional input features to further improve the speaker discrimination capability. All of these approaches showed promising results, which indicates that the multi-channel information can provide an alternative guider to discriminate the target speaker better. Actually, one of the keys of TSE is still

speech separation that refers to extract all overlapping speech sources from the mixed speech [11]. In order to further improve the separation ability, many strategies for exploiting the multi-channel information have been recently proposed, such as normalized crosscorrelation (NCC) [12], transform-average-concatenate (TAC) [13], and inter-channel convolution difference (ICD) [14], etc. Therefore, how to effectively exploit the multi-channel spatial information for TSE is crucial.

Considering that the mixed speech in an N -dimensional embedding space can be decomposed into N representations at the dimensional-level. Given multi-channel input mixtures, each dimensional-level representation among different microphone channels not only has high correlations, but also has some differentiations, both of these correlations and differentiations characterize the inter and intra speaker properties in the mixture. Our previous work in [15] proposed a channel decorrelation (CD) mechanism with parallel encoder target speaker adaptation to leverage these information for improving TSE systems. This decorrelation is performed on each dimension of all the multi-channel encoder representations of input mixtures, it is used to extract the inter-channel differential spatial information to learn difference between individual source signals of input mixture. Results in [15] have already shown that our original CD significantly improved the TSE performance over IPD features, however, the performance gains over the parallel encoder architecture [16] are still limited.

Therefore, in this study, we revisit the principle of our original channel decorrelation algorithm, two improvements are proposed to replace the original softmax-based inter-channel similarity calculation, one is using an unrolled probability and the other is using a normalized cosine-based similarity. Both of them are performed at the dimensional-level of encoder representations, and they aim to expand the dynamic range of the inter-channel differential probability to capture the fine-grained representation for a better spatial information utilization. Moreover, new combination strategies of the improved CD and target speaker adaptation of parallel encoder outputs are also investigated. All of experiments are performed on the publicly available multi-channel reverberant WSJ0 2-mix dataset. Results show that both of our improved channel decorrelation and the new adaptation strategy are very effective to improve the target speech extraction.

2. Related Works

2.1. Conv-TasNet

The Conv-TasNet is a time-domain speech separation technique proposed in [17]. It has become the mainstream speech separation approach because of its competitive performance over most of time-frequency domain speech separation algorithms. This separation structure has attracted widespread attention and

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further improved in many recent works [18–21]. Conv-TasNet consists of three parts: an encoder (1d convolution layer), a mask estimator (several convolution blocks), and a decoder (1d deconvolution layer). The waveform mixture is first encoded by the encoder and then is fed into the temporal convolutional network (TCN) [22] based mask estimator to estimate a multiplicative masking function for each source. Finally, the source waveforms are then reconstructed by transforming the masked encoder representations using the decoder. More details of Conv-TasNet can be found in [17].

2.2. TD-SpeakerBeam

TD-SpeakerBeam is a Conv-TasNet based target speech extraction approach that has been recently proposed in [9]. As shown in Fig.1 (a), it introduces an auxiliary network with an encoder and a convolution block to transform a pre-saved target speaker enrollment waveform to a target speaker embedding vector. This vector is then used in a scaling adaptation layer [23] to drive the network to extract the corresponding speech. The encoder and convolution block in the auxiliary network have the same structure as they used in Conv-TasNet. The speaker adaptation is performed between the first and second convolution blocks of the mask estimator. And in the adaptation layer, the mixture representation matrix is re-weighted by an element-wise multiplication with the repeated target speaker embedding vectors. Furthermore, by concatenating the IPD features after the adaptation layer, the performance of TD-SpeakerBeam can be further improved.

In Fig.1 (a), y_1 , \hat{x}^s , a^s , and e^s are the mixture waveform of the first channel, the extracted target speech waveform, the enrollment utterance of the target speaker, and the target speaker embedding vector respectively. We use this IPD configuration as one of our baseline.

3. Channel Decorrelation

3.1. Original CD

Assuming there is a N -dimensional embedding space, each channel of the multi-channel mixture waveform can be encoded into a $N \times T$ embedding matrix, where T is the length of that matrix. Each dimensional-level vector of the mixture embedding can be seen as one type of fine-grained representation that corresponds to different local traits of the whole input mixture. Our channel decorrelation is proposed to construct such a latent embedding space to learn dimensional-level discriminate feature representations between the encoded representations of multi-channel input mixtures, by taking the advantage of spatial configuration of multi-channel microphone-arrays. Taking two channels for example, the CD process is shown in Fig.1 (b). It accepts two encoded mixture representations \mathbf{W}_1 and \mathbf{W}_2 , and then outputs the differential spatial information \mathbf{W}_{cd} between two channels. The specific calculation of \mathbf{W}_{cd} is as follows:

First, compute the cosine correlation between \mathbf{W}_1 and \mathbf{W}_2 in each corresponding dimension (row). Note that \mathbf{W}_1 and \mathbf{W}_2 are two matrices, i.e.,

$$\mathbf{W}_i = [\mathbf{w}_i^1, \mathbf{w}_i^2, \dots, \mathbf{w}_i^N]^T, i = 1, 2 \quad (1)$$

where $\mathbf{W}_i \in \mathbb{R}^{N \times T}$ is the input of the i th channel of CD mechanism, N is the output dimension of convolutional encoder, T is the length of convolutional encoder output. $\mathbf{w}_i^j \in \mathbb{R}^{1 \times T}$, $j = 1, 2, \dots, N$, is the j th dimension vector of the i th channel. \mathbf{T} is the operation of transpose.

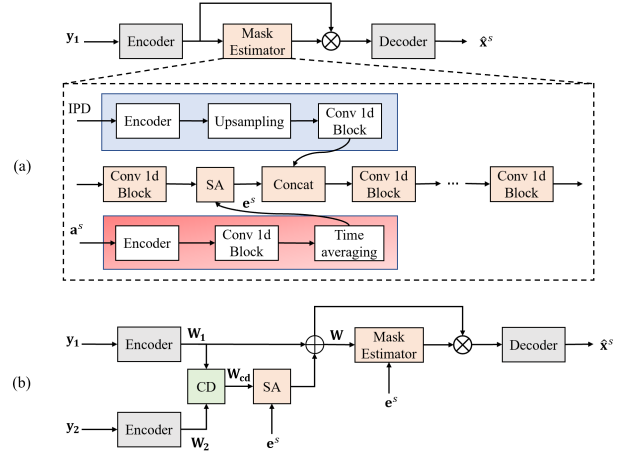


Figure 1: *Conv-TasNet based structures that incorporated with IPD features and channel decorrelation (CD) mechanism for the target speech extraction task. All network modules are the same as Conv-TasNet. “SA” indicates scaling adaptation.*

The cosine correlation of j th dimensional vector between the first channel and the second channel is calculated as,

$$\phi_{1,2}^j = \frac{\langle \mathbf{w}_1^j, \mathbf{w}_2^j \rangle}{\|\mathbf{w}_1^j\|_2 \|\mathbf{w}_2^j\|_2}, j = 1, 2, \dots, N \quad (2)$$

where $\langle \cdot \rangle$ is the inner product of two vectors, $\|\cdot\|_2$ represents the Euclidean norm. The vectors involved in the operation are normalized to zero mean prior to the calculation.

Then, calculate the cosine correlation between the vectors of each dimension of \mathbf{W}_1 and \mathbf{W}_2 in turn, and concatenate them to get a similarity vector $\phi_{1,2}$,

$$\phi_{1,2} = [\phi_{1,2}^1, \phi_{1,2}^2, \dots, \phi_{1,2}^N]^T \quad (3)$$

where $\phi_{1,2} \in \mathbb{R}^{N \times 1}$ represents the similarity of two encoded mixture representations in each dimension in a latent space.

Next, introduce an auxiliary vector \mathbf{a} with the same size as $\phi_{1,2}$ and values are all 1, i.e.,

$$\mathbf{a} = [1, 1, \dots, 1]^T \quad (4)$$

where $\mathbf{a} \in \mathbb{R}^{N \times 1}$ can be regarded as the cosine correlation between first channel \mathbf{W}_1 itself, which represents the reference vector of $\phi_{1,2}$. Then, a softmax operation is used to calculate the probability between each corresponding dimensional element in $\phi_{1,2}$ and \mathbf{a} as $p_{1,2}^j$ to get a similarity probability vector $\mathbf{p}_{1,2} \in \mathbb{R}^{N \times 1}$ as,

$$p_{1,2}^j = \frac{e^{\phi_{1,2}^j}}{e + e^{\phi_{1,2}^j}}, j = 1, 2, \dots, N \quad (5)$$

$$\mathbf{p}_{1,2} = [p_{1,2}^1, p_{1,2}^2, \dots, p_{1,2}^N]^T$$

Next, subtract $\mathbf{p}_{1,2}$ from \mathbf{a} to get a vector $\mathbf{s}_{1,2} \in \mathbb{R}^{N \times 1}$ that represents differentiated scores between channels, and repeat $\mathbf{s}_{1,2}$ to the same size as \mathbf{W}_2 to get the differentiated score matrix $\mathbf{S}_{1,2} \in \mathbb{R}^{N \times T}$, i.e.,

$$\mathbf{s}_{1,2} = \mathbf{a} - \mathbf{p}_{1,2} \quad (6)$$

$$\mathbf{S}_{1,2} = [\mathbf{s}_{1,2}, \mathbf{s}_{1,2}, \dots, \mathbf{s}_{1,2}]$$

Finally, the differentiated spatial information \mathbf{W}_{cd} between channels can be obtained by multiplying \mathbf{W}_2 by $\mathbf{S}_{1,2}$ as

$$\mathbf{W}_{cd} = \mathbf{W}_2 \odot \mathbf{S}_{1,2} \quad (7)$$

where $\mathbf{W}_{cd} \in \mathbb{R}^{N \times T}$, \odot denotes element-wise multiplication. In addition, to better guide the network towards extracting speech of target speaker, we also perform the target speaker scaling adaptation on \mathbf{W}_{cd} to exploit the target speaker-dependent spatial information, as shown in Fig.1 (b).

3.2. Unrolled CD

Note that in Equation (5), the softmax-based inter-channel similarity $p_{1,2}^j$ ranges from $\frac{1}{e^2+1}$ to $\frac{1}{2}$ or 0.12 to 0.5, which results in a $\mathbf{s}_{1,2} \in [0.5, 0.88]$ differentiated score range. It means that the original CD tends to assign a bigger but coarse weight on the inter-channel differential information than the correlations. This is incompatible with our motivation. Intuitively, ideal inter-channel differential information of the same utterance from different microphone channels should be less but fine and discriminative enough.

In order to alleviate such a problem, we modify the Equation (5) as

$$p_{1,2}^j = \frac{2e^{\phi_{1,2}^j}}{e + e^{\phi_{1,2}^j}}, j = 1, 2, \dots, N \quad (8)$$

By doing so, we can achieve a $[0.24, 1]$ inter-channel similarity range and the resulting differentiated score range becomes $[0, 0.76]$. We call this modification as “unrolled CD”. By expanding the dynamic range of $\mathbf{s}_{1,2}$ distribution, the unrolled CD may result in more fine and discriminative inter-channel differential information.

3.3. Cosine-based CD

Another method to address the limitation of original CD that mentioned in section 3.2 is to directly normalize the cosine-based inter-channel similarity to $[0,1]$ using the conventional cosine probability transformation $(1 + \phi_{1,2}^j)/2$. In this way, the resulting differentiated score is defined as

$$\mathbf{s}_{1,2} = \frac{\mathbf{a} - \phi_{1,2}}{2} \quad (9)$$

And the dynamic range of $\mathbf{s}_{1,2}$ becomes $[0, 1]$. We call this modification as “cosine-based CD”.

3.4. CD with Parallel Encoder Speaker Adaptation

Besides the improvements of the original CD algorithm, we also investigate new combination strategies of the CD-based spatial information and target speaker adaptation of parallel encoder outputs. Subfigure (a) and (b) in Fig.2 demonstrate two new different combination strategies, in both of which the output of CD only plays a role of effective inter-channel discriminative spatial information as the IPD used in TD-SpeakerBeam. These two proposals are motivated by the speculation that the distribution of \mathbf{W}_{cd} is very sparse. Such a sparse distribution may not contain enough differential spatial information to well characterize the target speaker identity.

4. Experiments

4.1. Dataset

Our experiments are performed on the multi-channel reverberant WSJ0-2mix corpus [24]. Multi-channel recordings are gen-

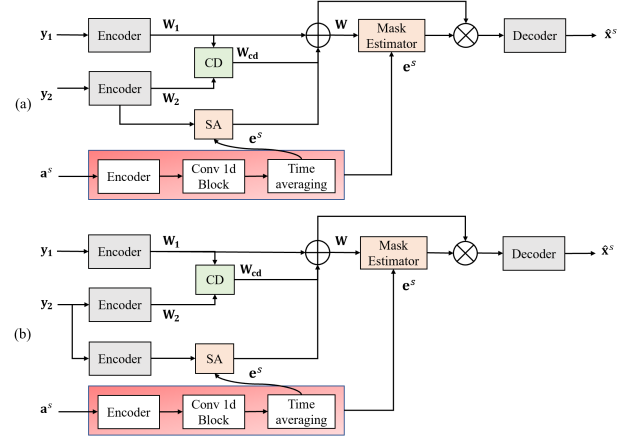


Figure 2: Different combination strategies of the CD-based spatial information and target speaker adaptation of parallel encoder outputs.

erated by convolving clean speech signals with room impulse responses simulated with the image method for reverberation time of up to about 600ms [9]. The dataset consists of 8 channel recordings, but to have a fair comparison with the state-of-the-art baselines, we also use only two channels in experiments.

We use the same way as in [25] to generate adaptation utterances of the target speaker. The utterance is anechoic and is selected randomly that different from the one in the mixture. The size of training, validation, and test sets are 20k, 5k, and 3k utterances, respectively.

4.2. Configurations

All experiments are performed based on our previously released open source of the original CD-based TSE in [15, 26]. We still use the same hyper-parameters as our the original CD [15]. However, it’s worth noting that in this study, we only use the scale-invariant signal-to-distortion ratio (SI-SDR) [27] as the training objective function, instead of the multi-task loss with a cross-entropy speaker identification loss used in our previous work of [15]. For evaluating the performance, besides the signal-to-distortion ratio (SDR) of BSSEval [28] and SI-SDR, we also report the PESQ [29] and STOI [30] to measure the speech quality and intelligibility.

4.3. Results in SDR/SI-SDR

Table 1 shows the performance comparison among different frameworks in SDR/SI-SDR for female-female (FF), male-male (MM), female-male (FM) and the whole (Avg) conditions of the reverberated WSJ0-2mix evaluation sets. The first ten systems, from “TSB [9]” to “CD-Old-IPD”, are all mentioned in our previous work of [15], but their performances are slightly worse (absolute $< 0.2\text{dB}$ SDR) than the numbers we reported in [15]. Because they are all only trained using the SI-SDR instead of the multi-task loss.

“TSB [9], TSB-IPD [9]” are the state-of-the-art TD-SpeakerBeam (TSB) without and with IPD features. Because the source code of TSB is not publicly available, we implemented this algorithm in [26] and the reproduced results shown in “TSB (our), TSB-IPD (our)” are slightly better than the ones in [9]. Instead of using IPD to exploit the multi-channel spatial information, the parallel encoder (Para-Enc) [16] was proposed

Table 1: *SDR/SI-SDR (dB) performance of different target speech extraction systems. “SA” represents the speaker adaptation is also performed on the parallel encoder output. “FF”, “MM”, “FM” and “Avg” represent the female-female, male-male, female-male and average conditions. All the systems are trained only using the SI-SDR loss.*

System	SA	FF	MM	FM	Avg
TSB [9]	-	9.13 / -	9.47 / -	12.77 / -	11.17 / -
TSB-IPD [9]	-	10.17 / -	10.30 / -	12.49 / -	11.45 / -
TSB(our)	-	9.43/8.84	10.02/9.52	12.54/12.06	11.26/10.76
TSB-IPD(our)	-	10.01/9.46	10.51/10.02	12.80/12.31	11.65/11.15
Para-Enc	-	10.83/10.21	11.52/10.99	13.12/12.61	12.26/11.72
Para-Enc	✓	11.31/10.68	11.91/11.41	13.31/12.79	12.56/12.02
CD-Old	-	11.16/10.53	11.67/11.10	13.20/12.69	12.41/11.85
CD-Old	✓	11.53/10.92	12.02/11.52	13.60/13.09	12.78/12.26
CD-Old-Tied	✓	11.41/10.82	11.83/11.27	13.53/13.02	12.67/12.13
CD-Old-IPD	✓	11.20/10.58	11.79/11.27	13.39/12.88	12.55/12.01
CD-Unrolled	-	11.41/10.78	12.10/11.60	13.64/13.14	12.81/12.28
CD-Unrolled	✓	11.33/10.70	12.27/11.77	13.78/13.28	12.92/12.39
CD-Cosine	-	11.30/10.62	12.09/11.57	13.61/13.12	12.77/12.24
CD-Cosine	✓	11.53/10.89	12.08/11.56	13.57/13.06	12.78/12.25
CD-Para-a	✓	11.71/11.10	12.28/11.76	13.80/13.28	13.00/12.46
CD-Para-b	✓	11.41/10.74	12.15/11.64	13.68/13.17	12.84/12.30

to directly sums each mixture encoded representation of multi-channels as the input of mask estimator, and it achieved much better results than IPD. Therefore, our study is also based on this parallel encoder structure, and “Para-Enc” is taken as the strongest baseline for all of the proposed CD-based systems.

“CD-Old, CD-Old-Tied” and “CD-Old-IPD” are the improved parallel encoder-based TSE systems using our previously proposed CD mechanism without and with parameters sharing between encoders, CD together with IPD features respectively. We see that the “CD-Old” achieves the best results over other two systems. Sharing encoder parameters in CD does not bring any extra benefits. And incorporating IPD features degrades the performance a little bit, this may results from the spatial information mismatch between IPD features and CD output, because IPD is extracted in the frequency domain, while the CD is performed in the time-domain.

Comparing “CD-Old” with “Para-Enc” baseline, we see that our original CD does improved the average SDR/SI-SDR from 12.26/11.72 to 12.41/11.85. However, these performance gains are very limited. By introducing the unrolled and cosine-based CD, significant performance gains are obtained, and the unrolled way achieves the best result. Absolute 0.55/0.56 dB (from 12.26/11.72 to 12.81/12.28) average SDR/SI-SDR gains have been obtained by “CD-Unrolled” over the “Para-Enc” baseline. And these gains are consistent for all the same-gender (FF, MM) and cross-gender (FM) evaluation conditions.

Moreover, besides the improvements of CD mechanism, the proposed additional scaling speaker adaptation (SA) on the encoder output also improves the parallel encoder-“Para-Enc” baseline. By comparing the “Para-Enc” with (perform the SA on the W_2) and without “SA”, we obtain an absolute 0.3dB average SDR/SI-SDR improvement. Consistent gains are also achieved in the “CD-Old” system (as Fig.1 (b)). However, it’s interesting to find that when we generalize the “SA” on W_2 of the improved CD, “CD-Unrolled” or “CD-Cosine”, the performance improvements become very limited, this may due to

the fact that the differential spatial information extracted by the improved CD is too sparse to characterize the target speaker identity. This differential information is more useful to enhance the mixture separation. Therefore, by changing the combination strategy between CD and SA of parallel encoder outputs as in Fig.2 (a) and (b), we obtain two corresponding results in system “CD-Para-a” and “CD-Para-b”. Compared with “CD-Unrolled (w/o SA)”, “CD-Para-a” still can achieve a further absolute 0.3 dB SDR/SI-SDR improvement (from 11.41/10.78 to 11.71/11.10) on the “FF” evaluation condition, although the improvements on other conditions are relatively small. That’s to say, incorporating the improved CD and SA in an appropriate way can maximize their individual advantage for improving a TSE system. Compared with the “Para-Enc” baseline, the best proposed system “CD-Para-a” achieves 8.12/8.71%, 6.59/7.00%, 5.18/5.31%, and 6.03/6.31% relative improvements in SDR/SI-SDR for female-female, male-male, female-male, and average conditions respectively. It indicates that the improved CD with SA can significantly enhance the target speech extraction, especially for the same-gender mixtures.

4.4. Results in PESQ/STOI

Table 2: *PESQ/STOI performance of different target speech extraction systems. “SA” represents the speaker adaptation is also performed on the parallel encoder output.*

System	SA	PESQ	STOI
Mixture	-	1.964	0.742
Para-Enc	-	2.987	0.910
Para-Enc	✓	3.047	0.915
CD-Unrolled	-	3.107	0.917
CD-Unrolled	✓	3.143	0.918
CD-Para-a	✓	3.142	0.920

Table 2 shows the PESQ and STOI performance comparison among different systems. Compared with “Para-Enc (w/o SA)”, the “CD-Unrolled (w/o SA)” achieves 0.120 and 0.007 improvements in PESQ and STOI respectively. Furthermore, we also see that the additional speaker adaptation performed on the parallel encoder output is also beneficial for improving the speech quality and intelligibility.

5. Conclusion

In this study, two methods are proposed to improve our original channel decorrelation algorithm. By expanding the dynamic range of decorrelation distribution, the improved algorithm can capture better fine-grained inter-channel differential information. Furthermore, new combination strategies of the improved decorrelation and additional target speaker adaptation of encoder outputs are also investigated. Extensive experimental results demonstrate that both the improvements on the original channel decorrelation and target speaker adaptation are very effective to build a much better target speech extraction system. Our future work will focus on generalizing the proposed algorithms to other related tasks.

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7. References

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