# **DPT-FSNET: DUAL-PATH TRANSFORMER BASED FULL-BAND AND SUB-BAND FUSION NETWORK FOR SPEECH ENHANCEMENT**

Sub-band models have achieved promising results due to their ability to model local patterns in the spectrogram. Some studies further improve the performance by fusing sub-band fusion model was not fully explored. This paper proposes a dual-path transformer-based full-band and sub-band full-band information, respectively. The features utilized by our proposed method are more interpretable than those utilized by the time-domain dual-path transformer. We conducted experiments on the Voice Bank + DEMAND and Interspeech 2020 Deep Noise Suppression (DNS) datasets to evaluate the proposed method. Experimental results show that the proposed method outperforms the current state-of-the-art.

(1)

## Motivations

- full-band and sub-band feature modeling.



improved transformer.

### Encoder

DPTPM

The intra-transformer processing block models the sub-band of the input features, which acts on the second dimension of D  $D_{b}^{intra} = [f_{b}^{intra}(D_{b-1}^{inter}[:,:,i], i = 1, \dots, F)]$ 

The inter-transformer processing block is used to summarize the information from each sub-band of the intra-transformer output to learn the global information of the speech signal, which acts on the last dimension of D

$$D_b^{inter} = [f_b^{inter}(D_b^{intra}[:, j, :], j = 1, \dots, T)]$$
(2)

### Decoder

The feature from the DPTPM output is passed through the decoder to obtain the estimated complex ratio mask. The enhanced complex spectrum is obtained by the element-wise multiplication between encoder's input and the mask.

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# Abstract

## **Results & Discussion**

We use a small-scale (Voicebank+DEMAND) and a large-scale dataset (DNS dataset) to evaluate the proposed model. In both of the above datasets, we compared our proposed algorithm with the current state-of-the-art, as shown in Tables 1 and 2.

SE-Conformer       3.13       0.95       4.45       3.55       3.82       -       CTS-Net       3.02         .earnable Loss Mixup       3.26       -       4.49       3.27       3.91       20.32       GaGNet       -	7 (2.78) 92.62 (96. 2 (2.94) 92.70 (96. (3.17) - (97.13	$\begin{array}{cccccccccccccccccccccccccccccccccccc$
SE-Conformer 3.13 0.95 4.45 3.55 3.82 - CTS-Net 3.02	7 (2.78)    92.62 (96. 2 (2.94)    92.70 (96.	$\begin{array}{c} 11) & 15.75 (17.29) \\ 66) & 15.58 (17.99) \end{array}$
	7 (2.78) 92.62 (96.	11) $15.75(17.29)$
DEMUCS 3.07 0.95 4.31 3.40 3.63 33.5 FullSubNet 2.9		$11) 1\Gamma 7\Gamma (17.00)$
T-GSA 3.06 - 4.18 3.59 3.62 - PoCoNet 2.83	3 (2.75) - (-)	- (-)
TSTNN 2.96 0.95 4.33 3.53 3.67 0.92 DTLN	- (-) 84.68 (94.	76) 10.53 (16.34)
MetricGAN         2.86         -         3.99         3.18         3.42         1.90         NSNet         2.3	7 (2.15) 90.43 (94.	47) 14.72 (15.61)
Noisy 1.97 0.91 3.34 2.44 2.63 - Noisy 1.82	2 (1.58) 86.62 (91.	52) 9.03 (9.07)
Method WB-PESQ STOI CSIG CBAK COVL Para. (M) Method WE	B-PESQ STOI (%	6) SI-SDR (dB)

Table 1. Comparison with other state-of-the-art systems on the VCTK+DEMAND dataset. Table 2. Comparison with other state-of-the-art systems on the DNS with\_reverb

 $(no\_reverb)$  test sets.

To further validate the effectiveness of our method, we performed two ablation analysis experiments.

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Method	WB-PESG	STOI	Method	WB-PESQ	STOI
CED + Dual-path former	2.97	0.95	STFT + CED + Original Transfomer	3.04	0.95
STFT + CED + Sub-band former	3.20	0.95	STFT + Improved Transfomer	3.11	0.95
STFT + CED + Full-sub former	3.33	0.96	STFT + CED + Improved Transfomer	3.33	0.96

Table 3. Ablation analysis results in terms of feature modeling on the VBD dataset. Table 4. Ablation analysis results in terms of model structure on the VBD dataset. In Table 3: By comparing exp.3 and exp.2, it can be seen that using two transformers to model sub-band and full-band information separately improves the performance over using two identical transformers to model only sub-band information. Moreover, the results of exp.3 are much better than exp.1, which proves that the frequency domain feature is more effective than the time domain feature for the

dual-path transformer.

In Table 4: By comparing exp.3 and exp.1, we can find that the performance of the improved transformer is much better than that of the original transformer. Performance can be improved by combining a convolutional encoder/decoder with the transformer as shown in exp.3 and exp.2.

# Conclusions

In this paper, we propose a dual-path transformer-based full-band and sub-band fusion network for speech enhancement in the frequency domain. Inspired by the full-band and sub-band fusion models, we explore features that are more efficient for dual-path structures with the intra part in the dual-path transformer models the sub-band information, and the inter part models the full-band information. Experimental results on the Voice Bank + DEMAND dataset and DNS dataset show that the proposed method outperforms the current state of the art at a relatively small model size.

# References

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