



Conformer-based Hybrid ASR System for Switchboard Dataset

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ICASSP, May, 2022

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Overview

1. Introduction
2. Our Proposed Conformer Acoustic Model
3. Experimental Setup
4. Experimental Results
5. Conclusion & Outlook

Hybrid ASR system & Conformer Architecture

- Hybrid neural network (NN)-hidden Markov model (HMM) automatic speech recognition (ASR) systems [Bourlard & Morgan 93] have achieved state-of-the-art performance on different tasks [Zhou & Michel⁺ 20, Lüscher & Beck⁺ 19, Kitza & Golik⁺ 19].

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⇒ **We present and evaluate a competitive conformer-based hybrid model training recipe**

Efficient Training With Time Down-/up-sampling

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⇒ **We apply time downsampling for efficient training and use transposed convolutions to upsample the output sequence**

Standard Conformer Architecture [Gulati & Qin⁺ 20]

- One conformer block consists of 3 types of modules: feed-forward (FFN) module, multi-head self-attention (MHSA) module, convolution (Conv) module
- Let x be the input sequence to conformer block i , then the equations of conformer block can be defined:

$$x_{FFN_1} = x + \frac{1}{2}FFN(x)$$

$$x_{MHSA} = x_{FFN_1} + MHSA(x_{FFN_1})$$

$$x_{Conv} = x_{MHSA} + Conv(x_{MHSA})$$

$$x_{FFN_2} = x_{Conv} + \frac{1}{2}FFN(x_{Conv})$$

$$\text{ConformerBlock}_i = \text{LayerNorm}(x_{FFN_2})$$

Our Proposed Conformer Acoustic Model

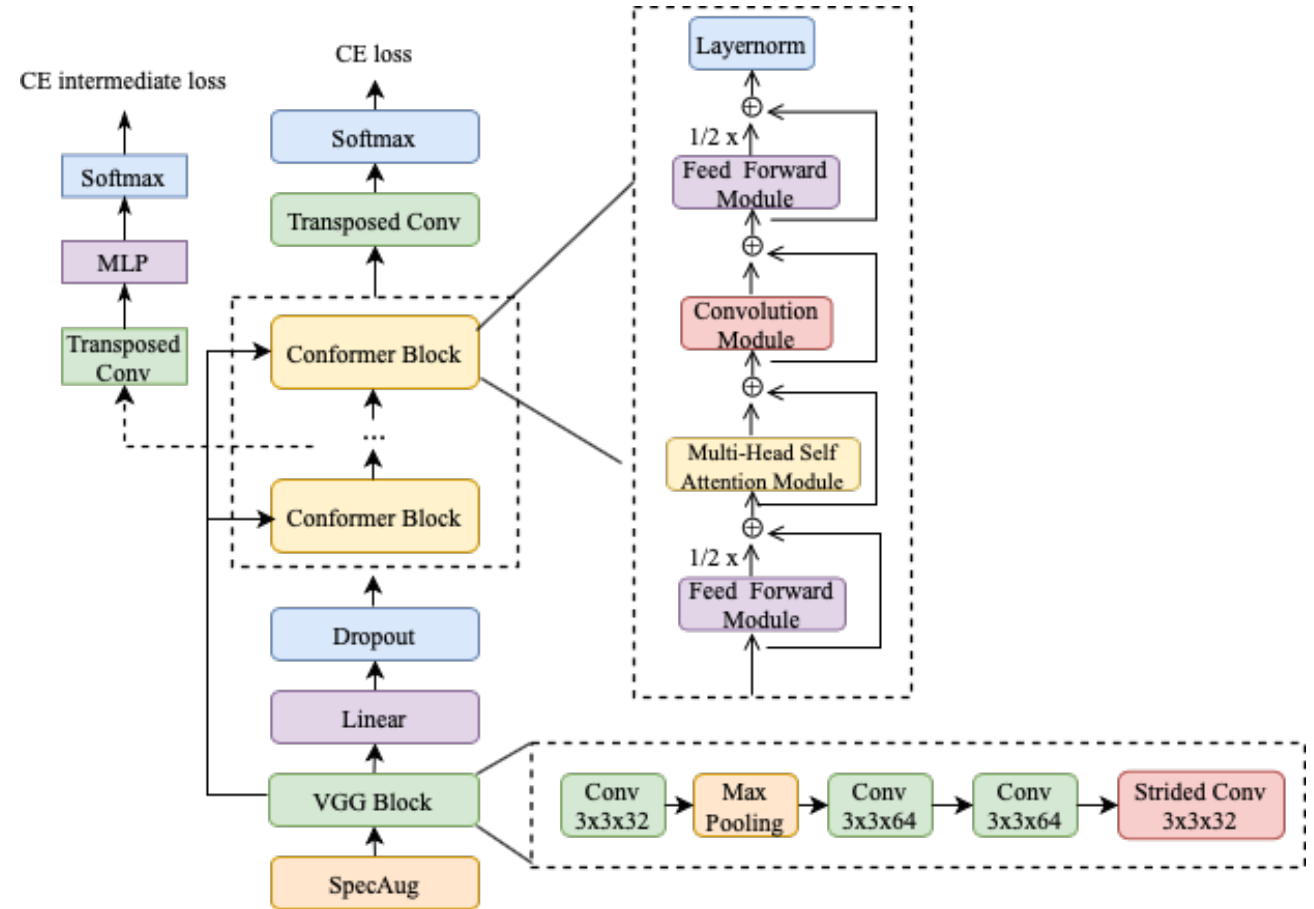
Time down-/up-sampling

- Use a strided convolution as part of the VGG network for downsampling.
- Use transposed convolution to upsample again to the frame-wise target alignment length before output

Intermediate loss[Tjandra & Liu⁺ 20]:
add intermediate losses at different layers

LongSkip: connect the output of the VGG network to the input of each conformer block [Huang & Liu⁺ 17]

Parameter sharing: share the parameters of intermediate loss layers and the ones of transposed convolution layers



Experimental Setup

Data

- Switchboard 300h dataset (English telephone speech)
- Hub5'00 (Switchboard + CallHome) as development set and Hub5'01 as test set.

Baseline

- Input: 40-dimensional Gammatone features
- Output units: 9001 state-tied CART (Classification and Regression Tree) labels
- Target: frame-wise alignment generated from a HMM-GMM system
- Model size: 12 conformer blocks
- Model dimension: attention dimension of each MHSA module is 512 with 8 attention heads, dimension of the feed-forward module is 2048
- Regularization: dropout with 10%, focal loss with factor 2

Experimental Setup

Language Model

- Use 4-gram count-based language model (LM) and LSTM LM in single pass decoding
- Use transformer (Trafo) LM for rescoring

LM	PPL (word-level) on Hub5'00
4-gram	79.5
LSTM	51.3
Transformer	48.1

Sequence Discriminative Training

- Lattice-based state-level minimum Bayes risk (sMBR) criterion
- Lattices generated using a bigram LM
- sMBR loss scale 0.9 and CE loss scale 0.1

Experimental Results

Kernel size & Number of Conformer Blocks (with 4-gram LM)

- Kernel size has a significant effect on WER
- Using smaller kernel size for depth-wise convolution is better
- We gain performance as we use deeper network

- Kernel size for depth-wise convolution

Kernel size	WER [%]		
	Hub5'00		
	SWB	CH	Total
6	8.4	17.1	12.8
8	8.1	16.8	12.5
16	8.2	17.6	12.9
32	8.4	18.0	13.2

- Number of Conformer blocks comparison

L	Params. [M]	WER [%]		
		Hub5'00		
		SWB	CH	Total
6	42	8.5	18.0	13.3
8	59	8.1	17.3	12.7
12	88	8.1	16.8	12.5

Experimental Results

Time Downsampling Factors and Variants (with 4-gram LM)

- Set the filter size and the stride of the transposed convolution as time reduction factor
- Choose down-sampling factor 3 by considering tradeoff between speed and performance
- Strided convolution applied at the end of the VGG network works best
- Time downsampling factor comparison
- Time downsampling variants comparison

Factor	Train time [h]	WER [%]		
		Hub5'00		
		SWB	CH	Total
2	1.28	8.3	16.4	12.4
3	0.92	8.1	16.8	12.5
4	0.86	8.4	17.9	13.2
5	0.73	8.7	18.6	13.7

Method	WER [%]		
	Hub5'00		
	SWB	CH	Total
BLSTM+maxpool	8.2	17.0	12.7
VGG-layer2	8.4	17.7	13.1
VGG-layer4	8.1	16.8	12.5

* VGG-layerX refers to strided convolution as X^{th} layer of VGG network, BLSTM+maxpool refers to one BLSTM layer with 512 units followed by time max-pooling layer

Experimental Results

Ablation Study of Training Methods (with 4-gram LM)

- SpecAugment is the most important method giving **20% relative improvement**
- Using intermediate loss is important for better convergence and gives **7% relative improvement** in WER
- Sharing parameters between transposed convolutions helps
- Other training methods have marginal improvements

Training method	WER [%]		
	Hub5'00		
	SWB	CH	Total
Baseline	8.1	16.8	12.5
- SpecAugment	9.8	21.5	15.7
- Intermediate loss	8.9	18.1	13.5
- Share transp. conv params.	8.5	17.3	12.9
- LongSkip	8.1	17.2	12.7
- Focal Loss	8.1	17.0	12.6
+ Share MLP params.	8.2	16.9	12.5

Experimental Results

Comparison between Conformer and BLSTM AM (with 4-gram LM)

- The BLSTM-based model consists of 6 BLSTM layers following a well-optimized setup as here [Kitza & Golik⁺ 19]
- With comparable number of parameters, conformer AM outperforms BLSTM AM by around **9% relative**

AM	LSTM dim.	Params. [M]	Hub5'00
BLSTM	500	41	14.2
	600	57	13.8
	700	76	13.8
	800	96	13.7
	1000	146	13.3
Conformer	-	88	12.5

Experimental Results

Overall Results

- **8.5% relatively better** on Hub5'00 compared to BLSTM hybrid system with LSTM LM
- Outperforms a well-trained RNN-T model with much fewer epochs.
- On par with a well-optimized BLSTM attention system [Tüske & Saon⁺ 20] on Hub5'01 test set
- The state-of-the-art conformer attention-based system trains much longer and uses cross-utterance LM

Work	#Epochs	Approach	AM	LM	seq. train	WER [%]	
						Hub 5'00	Hub 5'01
[Kitza & Golik ⁺ 19]	-	Hybrid	LSTM	4-gram LSTM	yes	13.9 11.7	-
[Zhou & Berger ⁺ 21]	100	RNN-T	LSTM	LSTM Trafo	no	11.5 11.2	11.5 11.2
[Tüske & Saon ⁺ 20]	250	LAS	LSTM	LSTM	no	9.8	10.1
[Tüske & Saon ⁺ 21]	250	LAS	Conf.	- LSTM Trafo	no	9.9 8.6 8.4	10.1 8.5 8.5
ours	27	Hybrid	Conf.	4-gram LSTM 4-gram LSTM Trafo	no yes	12.5 11.3 11.9 10.7 10.3	12.1 10.5 11.4 10.1 9.7

Summary

- For the first time, a training recipe for a conformer-based hybrid model is evaluated
- We combined different training methods from the literature that boosted the word-error-rate
- We applied time down-sampling using strided convolution to speedup training and used transposed convolution as a simple method to upsample again
- We observed SpecAugment and intermediate loss layers are necessary to achieve good performance
- Our model outperforms the BLSTM-based hybrid model significantly

Follow up work

- We extend this training recipe as well as use speaker adaptation to improve the WER 11% relative i.e. from 10.3 to 9.2 on Hub5'00 with Transformer LM [Zeineldeen & Xu⁺]

Thank you for your attention

Any questions?



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