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Towards Efficient Recurrent Architectures: A Deep LSTM Neural Network Applied to Speech Enhancement and Recognition

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Abstract Long short-term memory (LSTM) has proven effective in modeling

sequential data. However, it may encounter challenges in accurately capturing long-term temporal dependencies. LSTM plays a central role in speech enhancement by effectively modeling and capturing temporal dependencies in speech signals. This paper introduces a variable-neurons-based LSTM designed for capturing long-term temporal dependencies by reducing neuron representation in layers with no loss

of data. A skip connection between nonadjacent layers is added to

prevent gradient vanishing. An attention mechanism in these

connections highlights important features and spectral components. Or

LSTM is inherently causal, making it well-suited for real-time processing without relying on future information. Training involves utilizing combined acoustic feature sets for improved performance, and the models estimate two time-frequency masks-the ideal ratio mask (IRM) and the ideal binary mask (IBM). Comprehensive evaluation using perceptual evaluation of speech quality (PESQ) and short-time objective intelligibility (STOI) showed that the proposed LSTM architecture demonstrates enhanced speech intelligibility and perceptual quality. Composite measures further substantiated performance, considering residual noise distortion (Cbak) and speech distortion (Csig). The proposed model showed a 16.21% improvement in STOI and a 0.69 improvement in PESQ on the TIMIT database. Similarly, with the LibriSpeech database, the STOI and PESQ showed improvements of 16.41% and 0.71 over noisy mixtures. The proposed LSTM architecture outperforms deep neural networks (DNNs) in different stationary and nonstationary background noisy conditions. To train an automatic speech recognition (ASR) system on enhanced speech, the Kaldi toolkit is used for evaluating word error rate (WER). The proposed LSTM at the front-end notably reduced WERs, achieving a notable 15.13% WER across different noisy backgrounds.

Keywords

Author Keywords: Deep learning; Speech enhancement; Speech recognition; Skip connections; LSTM; Acoustic features; Attention process

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