Towards efficient self-supervised representation learning in speech processing

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Abstract

Self-supervised learning has achieved impressive results in speech processing, but current models are computationally expensive, generating environmental concerns because of their high energy consumption. Therefore, we propose an efficient self-supervised approach to address high computational costs, using a single GPU during 24 to 48 hours of pretraining. The proposed approach combines linear, convolutional, and self-attention layers with several optimizations, including dynamic batching, flash attention, mixed-precision training, gradient accumulation, and acoustic feature extraction with input preprocessing. Computational cost estimations for our proposed model represent up to two orders of magnitude improvements in computational efficiency against existing speech models.

1 Introduction

Self-supervised models generate impressive results when learning latent representations, but their training is computationally expensive (Peng et al., 2023). Yet, their results in speech processing are astounding because downstream tasks strongly benefit from their learned representations (Mohamed et al., 2022; Parcollet et al., 2023b).

Self-supervised approaches for speech representation learning can be based on consistency or self-training (Zhang et al., 2020). Whether using consistency or self-training, large training costs represent a challenge. Indeed, most existing models require several GPUs for days to pretrain their neural architectures. This requirement causes several limitations. First, it hinders the training and deployment of speech models in computing platforms with low resources, such as edge devices and mobile platforms (Gaol et al., 2023; Mohamed et al., 2022). Secondly, reproducibility is challenging, as few labs can afford large computational resources (Lin et al., 2023). Last but not least, it creates environmental concerns because of the high energy consumption during training (Parcollet et al., 2023b).

To address those limitations, we propose an efficient self-supervised model to learn speech representations. Instead of focusing on the model performance in downstream tasks, the proposed model focuses primarily on computational costs, limiting the resources available for pretraining. We set a pretraining limit based on cramming (Geiping and Goldstein, 2023): we use a single GPU for 24 to 48 hours to train the model.

2 Related work

Several models have been recently proposed for self-supervised learning of speech representations, including CombinedSSL (Zhang et al., 2020), Mockingjay (Liu et al., 2020), Spiral (Huang et al., 2022), Data2vec2(Baevski et al., 2023), and DinoSR (Liu et al., 2023a). But two approaches have clearly emerged (Mohamed et al., 2022): Hidden unit BERT (HuBERT) (Hsu et al., 2021) and wav2vec2 (Baevski et al., 2020b). However, self-supervised models are quite costly, requiring a lot of computational resources for training. One alternative to reduce training costs is knowledge distillation (Allen-Zhu and Li, 2023), where a small student model learns from a large teacher model, which has been pretrained previously (Peng et al., 2023).

Using knowledge distillation, LightHuBERT (Wang et al., 2022) improves HuBERT with a once-for-all transformer model. The teacher is a HuBERT base model, while the student learns by predicting masked inputs in an iterative process. The transformer in LightHuBERT comprises subnets with sharable weights and several configuration parameters, enabling a random search to adjust the model to different resource constraints.

The student architecture in knowledge distillation methods is manually designed, and it does not

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change during training. However, modifying student architectures can have a considerable impact on model results, even for student architectures with similar sizes (Ashihara et al., 2022). Therefore, a joined distillation and pruning approach for speech SSL has been recently proposed, using HuBERT (DPHuBERT) or WavLM (DPWavLM) as the teacher models (Peng et al., 2023).

Yet, knowledge distillation approaches need a pretrained teacher model because student models can not be trained standalone (Chen et al., 2023). Thus, computational costs do not improve as they should include teacher model training. In contrast, MelHuBERT (Lin et al., 2023) proposes a simplified version of HuBERT that has twelve self-attention layers and a weighted sum of all the layers for downstream tasks. The input is a 40-dimensional Mel log spectrogram, so input sequences are shorter, reducing the multiplication and addition calculations by 33% (Lin et al., 2023).

There are also efforts to improve the wav2vec architecture. Proposed approaches improving wav2vec include squeezed and efficient wav2vec2 with disentangled attention (SEW-D) (Wu et al., 2022) and stochastic squeezed and efficient wav2vec2 (S-SEW) (Vyas et al., 2022).

Despite existing efforts to improve self-supervised model efficiency, there is still room to reduce the computational costs of self-supervised models. Computational costs create challenges when using these models in mobile devices and for training on very large datasets (Mohamed et al., 2022; Parcollet et al., 2023b). They also hinder the development of new approaches, the study of other training recipes, and the reproduction of experimental results, as few researchers can afford the cost (Chen et al., 2023; Lin et al., 2023; Parcollet et al., 2023b; Wang et al., 2023). Besides, computational costs have environmental implications, as training requires considerable amounts of energy (Parcollet et al., 2023b).

Likewise, few existing self-supervised models use half-precision numbers, even though this technique can half the memory requirements and accelerate the arithmetic computations on recent GPUs (Micikevicius et al., 2018). A similar issue happens with dynamic batching (Gaol et al., 2023; Tyagi and Sharma, 2020), a procedure that avoids wasting computing resources on the padded portion of speech mini-batches. Also, most models use standard self-attention layers, though efficient alternatives have been proposed recently, without using approximations (Dao et al., 2022; Parcollet et al., 2023a).

We address these limitations in the following section, proposing an efficient model for self-supervised learning of speech representations.

3 Efficient self-supervised approach

In this section, we describe our proposed model: efficient self-supervised learning (ESSL). We also summarize the optimizations used to improve model efficiency.

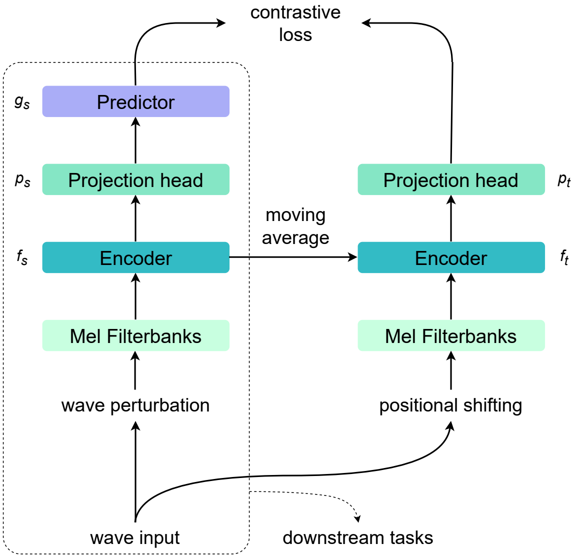

Figure 1: Neural architecture for our proposed ESSL approach, based on a teacher – student configuration (Huang et al., 2022).

3.1 Model architecture

The architecture uses a teacher – student configuration based on recent work for speech processing (Huang et al., 2022). The student part comprises an encoder, a projection head, and a predictor, while the teacher part comprises an encoder and a projection head (Figure 1). Following a conformer configuration (Gulati et al., 2020), the encoder has 3 convolutional layers, followed by 2 self-attention layers, 2 convolutional layers, and 10 self-attention layers. Projection heads are linear layers, and the predictor has 3 convolutional layers (Huang et al., 2022). Self-attention layers use relative position embeddings to better capture the sequence ordering of input sequences (Chen et al., 2022).

Pretraining relies on a contrastive loss to force the student latent representation to converge to the latent representation of the teacher part of the model, updating teacher weights with an exponential moving average of student weights (Chen et al., 2020; Huang et al., 2022). A contrastive loss in the teacher – student configuration is defined as follows (Chen et al., 2020; Huang et al., 2022):

$$\phi(a, b) = \frac{a^T b}{\|a\| \, \|b\|} \tag{1}$$

$$\mathcal{L} = -\sum_{i=1}^{T} \log \frac{e^{\phi(z_i, z_i')/\tau}}{\sum_{j \in D_i} e^{\phi(z_i, z_j')/\tau}} \tag{2}$$

where z is the latent representation from the student network and z' is the latent representation from the teacher network, τ represents a temperature parameter, and D_i is the set of distractors for the z_i representation.

Regularization for the proposed model includes dropout, SpecAugment (Park et al., 2019), random positional shifting (Huang et al., 2022), and multicondition training (Chiba et al., 2019) through noise addition. For noise addition, audio data comes from the DNS 2021 challenge (Reddy et al., 2021), adding noise audio to the utterances in the input dataset. Noise addition is performed randomly, with a probability of 0.5 (Huang et al., 2022).

After noise addition, random positional shifting is also used on the input sequences. Random shifting avoids the model exploiting positional information from input sequences. The shifting of input sequences forces the model to focus on speech data, and the sequences for the teacher model are readjusted before calculating the pretraining loss (Huang et al., 2022). Likewise, SpecAugment randomly masks the input audio sequence in the time and frequency domains (Park et al., 2019). Masks use zero values in the time domain, while Gaussian noise replaces the speech data of the masks in the frequency domain. Finally, dropout is applied in the self-attention layers of the model (Park et al., 2019).

3.2 Model optimizations

Optimizations in our proposed model include flash attention (Dao et al., 2022), mixed precision training (Micikevicius et al., 2018), dynamic batching (Tyagi and Sharma, 2020), gradient accumulation (Huang et al., 2023), and acoustic feature extraction (AFE) with input preprocessing (Parcollet et al., 2023b). AFE comprises the first part of the neural model, processing the input signal before feeding it to the subsequent layers. The best-performing approaches for AFE combine Mel Filterbanks for preprocessing the raw waveform before the convolutional module (Parcollet et al., 2023b), as we do in ESSL.

Batch sizes have a considerable impact on training performance (Chen et al., 2023; Hsu et al., 2021). To deal with the high memory requirements of large batch sizes with a single GPU, gradients are accumulated for a few training steps before applying them to update the parameters of the model (Huang et al., 2023). This approach enables the increase in batch size to get close to batch sizes used in large models (Liu et al., 2023a).

Another optimization involving training batches is dynamic batching (Ravanelli et al., 2021). Based on the duration of each audio file, dynamic batching packages one or several files into a single batch, keeping the total batch duration under a specified maximum duration. By doing so, dynamic batching minimizes the amount of padding that fixed batch sizes must use. This optimization reduces the amount of RAM required to train a model. It also eliminates the GPU iterations wasted when processing the padding data in fixed batch sizes.

Concerning the number format for model parameters and data, mixed precision training uses the floating point 16 (FP16) format. FP16, also known as half-precision, diminishes the size of the model and the batches, using less RAM during training than the floating point 32 (FP32) commonly used in computations. FP16 also enables faster training in the GPU, without affecting the convergence of the model (Micikevicius et al., 2018; Narayanan et al., 2021).

Lastly, FlashAttention (Dao et al., 2022) improves the efficiency of self-attention layers by focusing on the optimization of the input-output (IO) memory operations in the GPU. In general, GPUs have two kinds of memories. A small SRAM is associated with each kernel, and a large high-bandwidth memory, which is slower and is shared between all the kernels. Memory-intensive operations, like the matrix operation of the self-attention layers, have their bottleneck at the read-write RAM access. In contrast, compute-intensive operations have their bottleneck in the number of arithmetic operations that must be realized. As self-attention is primarily a memory-intensive operation, FlashAttention reduces the number of IO operations by tiling, assigning a matrix operation to a single kernel, and saving some results from the forward pass to share in the subsequent backward pass.

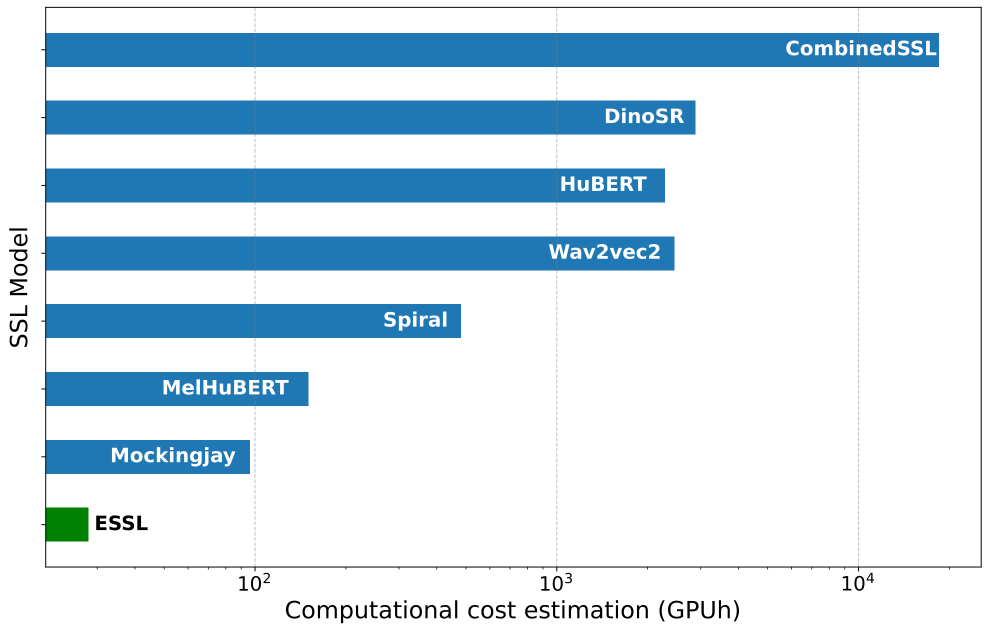

Figure 2: Cost estimation for pretraining speech SSL models. ESSL represents a remarkable reduction in computational costs against existing models.

4 Results and discussion

All experiments run on a single GPU, an NVIDIA GeForce RTX 3090 Ti, with 24 GB of memory and 1.56GHz of base clock rate. Considering training data, LibriSpeech 960h provides speech utterances for unsupervised pretraining. Finetuning for Automatic Speech Recognition (ASR) is performed with LibriSpeech 100h (Panayotov et al., 2015), using a CTC loss (Yan et al., 2023). Regarding training configuration, pretraining requires 60k iterations, which is equivalent to 15k pretraining steps because we do four gradient accumulations. The learning rate warms up the first 8% of the iterations to a maximum of 3e-4. For finetuning, 160k iterations are performed. This is equivalent to 40k finetuning steps with four gradient accumulations. The learning rate warms up the first 10% of the iterations to a maximum of 3e-5 (Huang et al., 2022). Code is publicly available[1] to facilitate the replication of experimental results.

Efficiency gains of ESSL are remarkable (Figure 2). Though metrics degrade against large speech models (Table 1), the computational cost estimation represents a fifth of recent work (Lin et al., 2023), diminishing from 150 GPUh to only 28 GPUh, and about a third of recent work (Liu et al., 2020). When doing a comparison against large models, their computational cost estimations are around one or two orders of magnitude larger. For example, Spiral takes 480 GPUh, which is 17 times larger than our proposed approach. Similarly, CombinedSSL takes 18432 GPUh, which is 658 times larger than ESSL.

SSL Model	ASR
Mockingjay (Liu et al., 2020)	15.48
wav2vec (Schneider et al., 2019)	11.00
vq-wav2vec (Baevski et al., 2020a)	12.80
wav2vec2 Base (Baevski et al., 2020b)	4.79
HuBERT Base (Hsu et al., 2021)	4.79
Spiral Base (Huang et al., 2022)	3.30
WavLM Base (Chen et al., 2022)	3.40
CombinedSSL (Zhang et al., 2020)	1.60
ESSL	10.69

Table 1: WER for LibriSpeech test-clean dataset (Yang et al., 2021). Models are pretrained with LibriSpeech 960h. ASR results use a language model for decoding.

As mentioned, batch size is crucial for training speech processing models (Chen et al., 2023). Using dynamic batching, half-precision, and gradient accumulation enables ESSL to get close to the batch sizes used in large speech models – but using one GPU only. The batch size has 18 minutes of audio data. With 4 gradient accumulations, it gets to 72 minutes. This size is close to batch sizes used

[1] https://github.com/Orange-OpenSource/essL

in recent speech models, such as 47 minutes in HuBERT, 96 minutes in wav2vec2, or 187 minutes in WavLM (Liu et al., 2023a).

Perturbations on input speech sequences are also crucial for the performance of ESSL. Removing them makes WER degrade from 29.91% to 40.08% (Table 2). This drop in performance indicates the importance of SpecAugment, random positional shifting, and multicondition training through noise addition in the pretraining process.

Other experiments to analyze ESSL include random initialization and MelHuBERT configuration. For experiments with MelHuBERT configuration, we used 40 Mel Filterbanks, with a 20ms hop length (Lin et al., 2023). Though training steps can be up to 36% faster given shorter input sequence lengths, WER drops considerably, going from 29.91% down to 51.09%. Concerning random initialization, we discarded pretrained weights and finetuned from a model with random weights. Results suggest finetuning only is not enough for speech processing. A WER of 99.7% highlights the importance of pretraining in final ESSL results.

Configuration	dev-other	dev-clean
ESSL	**28.18**	**10.38**
- w/o perturbations	40.08	17.88
- w/ 40 Mel Filterbanks	51.09	26.41
- random initialization	99.70	99.78

Table 2: Analysis of different configurations for ESSL. Results include WER performance on LibriSpeech dev-other and dev-clean datasets.

5 Limitations

Very-low data settings are challenging. The limited availability of data hinders research in speech processing for under-resourced languages (Liu et al., 2023b; Shi et al., 2021). We tested finetuning ESSL for ASR with the Librilight dataset (Kahn et al., 2020). Librilight has 10 hours, 1 hour, and 10 minutes datasets to finetune models, in contrast with the 100 hours available in LibriSpeech. Results indicate ESSL struggles in very-low data settings, with a WER of 97.30% in LibriSpeech dev-other (Table 3). This performance degradation is too high to perform ASR for under-resourced languages.

Method	dev-clean	dev-other
Librilight 10 min		
ESSL	96.41	97.30
wav2vec2 Base	8.9	15.7
HuBERT Base	9.1	15.0
Librilight 1 hr		
ESSL	96.05	96.41
wav2vec2 Base	5.0	10.8
HuBERT Base	5.6	10.9
Librilight 10 hr		
ESSL	70.45	81.62
wav2vec2 Base	3.8	9.1
HuBERT Base	3.9	9.0

Table 3: WER results for LibriSpeech dev-other and dev-clean datasets, using the Librilight very-low data settings of 10 minutes, 1 hour, and 10 hours for model finetuning.

6 Conclusion

In this work, we proposed ESSL, an efficient approach for self-supervised learning of speech representations. ESSL addresses high computational costs by combining several model optimizations and fixing a limit on computational resources available for pretraining. Estimations of computational cost reduction reveal up to two orders of magnitude improvements against existing speech SSL models. Overall, ESSL is a step in the process of reducing computational costs in SSL models, enabling their training in edge devices, facilitating the development of new approaches, and making them more environmentally friendly.

For future work, we will investigate our efficient approach for other speech processing tasks, including intent classification, keyword spotting, query by example, and other downstream tasks. We will also explore architectural modifications to improve model performance in very-low data settings.

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