wav2vec 2.0: A Framework for Self-Supervised Learning of Speech Representations

Alexei Baevski, Henry Zhou, Abdelrahman Mohamed, Michael Auli

Facebook AI

NeurIPS 2020

• Current speech recognition systems require thousands of hours of transcribed speech to reach acceptable performance

• Collecting large amount of labeled data is not available for most of 7,000 languages spoken worldwide

• Self-supervised learning has been very successful for natural language processing and computer vision

 Propose wav2vec 2.0, an end-to-end method for learning discrete speech units as well as contextualized representations from unlabeled speech

02 Methodology | Model Overview



1. Feature encoder

> Blocks of temporal convolution + layer normalization + GELU activation function

2. Quantization module

- > Discretize the encodings to a finite set of speech representations via **product quantization**
- Concatenate G chosen entries from each of G groups containing V entries
- > Gumbel softmax: probabilities for choosing the v-th entry in group g

$$p_{g,v} = \frac{\exp(l_{g,v} + n_v)/\tau}{\sum_{k=1}^{V} \exp(l_{g,k} + n_k)/\tau}$$

where τ is temperature factor, $n = -\log(-\log(u))$, u are uniform samples from $\mathcal{U}(0,1)$

3. Context network

Convolutional layer + GELU + Transformer encoders

02 Methodology | Comparison with Contrastive Predictive Coding (CPC)



1. Masking

> Randomly mask a proportion of p time steps in encodings and their subsequent M time steps

2. Objective

> Contrastive loss: given context network output c_t over masked time step t, identify the true

quantized q_t from itself and K distractors uniformly sampled from other masked time steps

$$\mathcal{L}_m = -\log \frac{\exp(sim(\boldsymbol{c}_t, \boldsymbol{q}_t)/k)}{\sum_{\widetilde{q} \sim Q_t} \exp(sim(\boldsymbol{c}_t, \widetilde{\boldsymbol{q}})/k)},$$

where cosine similarity $sim(a, b) = a^T b / ||a|| ||b||$, k = 0.1 is a temperature factor

Diversity loss: maximize the entropy of averaged softmax distribution to encourage the equal use of the V entries in each of the G groups

$$\mathcal{L}_d = \frac{1}{GV} \sum_{g=1}^G \sum_{\nu=1}^V \bar{p}_{g,\nu} \log \bar{p}_{g,\nu}$$

- Fine-tuning for ASR and phoneme representation
 - > Add a randomly initialized linear projection layer on top of the context network
 - > Vocabulary: 29 characters + 1 word boundary token for LibriSpeech dataset, 39 classes for TIMIT
 - SpecAugment
 - Minimize CTC loss

- Language model (LM) and decoding
 - ➤ 4-gram LM
 - > Transformer LM: 20 blocks, model dimension 1,280, FFN dimension 6,144, 16 attention heads

Datasets

- Pre-training: LibriSpeech-960 hours (LS-960), LibriVox-53.2k hours (LV-60k)
- Fine-tuning: LS-960, LibriSpeech-100 hours (LS-100), Libri-light (train-10h, train-1h, train-10min), TIMIT-5 hours
- Model configurations
 - Feature encoder: 7 blocks with temporal convolution of 512 channels, strides (5, 2, 2, 2, 2, 2, 2), kernel widths (10, 3, 3, 3, 3, 2, 2)
 - > Quantization module: G = 2, V = 320
 - > BASE: 12 Transformer blocks, model dimension 768, FFN dimension 3,072, 8 attention heads
 - > LARGE: 24 Transformer blocks, model dimension 1,024, FFN dimension 4,096, 16 attention heads

03 **Experiments** | Low-Resource Labeled Data Evaluation

- Word Error Rates (WER)
 - ▶ [42]: State-of-the-art method
 - LARGE model surpasses [42] with 1-hour

labeled data

- ➢ BASE vs. LARGE on LS-960
- ➤ LARGE LS-960 vs. LV-60k

Model	Unlabeled data	LM	dev		te	test	
			clean	other	clean	other	
10 min labeled							
Discrete BERT [4]	LS-960	4-gram	15.7	24.1	16.3	25.2	
BASE	LS-960	4-gram	8.9	15.7	9.1	15.6	
		Transf.	6.6	13.2	6.9	12.9	
LARGE	LS-960	Transf.	6.6	10.6	6.8	10.8	
	LV-60k	Transf.	4.6	7.9	4.8	8.2	
1h labeled							
Discrete BERT [4]	LS-960	4-gram	8.5	16.4	9.0	17.6	
BASE	LS-960	4-gram	5.0	10.8	5.5	11.3	
		Transf.	3.8	9.0	4.0	9.3	
LARGE	LS-960	Transf.	3.8	7.1	3.9	7.6	
	LV-60k	Transf.	2.9	5.4	2.9	5.8	
10h labeled							
Discrete BERT [4]	LS-960	4-gram	5.3	13.2	5.9	14.1	
Iter. pseudo-labeling [58]	LS-960	4-gram+Transf.	23.51	25.48	24.37	26.02	
1 00-0	LV-60k	4-gram+Transf.	17.00	19.34	18.03	19.92	
BASE	LS-960	4-gram	3.8	9.1	4.3	9.5	
		Transf.	2.9	7.4	3.2	7.8	
LARGE	LS-960	Transf.	2.9	5.7	3.2	6.1	
	LV-60k	Transf.	2.4	4.8	2.6	4.9	
100h labeled							
Hybrid DNN/HMM [34]	-	4-gram	5.0	19.5	5.8	18.6	
TTS data augm. [30]	-	LSTM			4.3	13.5	
Discrete BERT	LS-960	4-gram	4.0	10.9	4.5	12.1	
Iter. pseudo-labeling [58]	LS-860	4-gram+Transf.	4.98	7.97	5.59	8.95	
	LV-60k	4-gram+Transf.	3.19	6.14	3.72	7.11	
Noisy student 42	LS-860	LSTM	3.9	8.8	4.2	8.6	
BASE	LS-960	4-gram	2.7	7.9	3.4	8.0	
		Transf.	2.2	6.3	2.6	6.3	
LARGE	LS-960	Transf.	2.1	4.8	2.3	5.0	
	LV-60k	Transf.	1.9	4.0	2.0	4.0	

03 **Experiments** | High-Resource Labeled Data Evaluation

- WER on LS-960 labeled data
 - Weaker base model: LARGE -from scratch vs. ContextNet [17]
 - Vocabulary mismatch AM-char; LM-word
 - > No data balancing as used in [42]
 - > The proposed method and [42] are complimentary

Unlabeled	LM	dev		test	
data		clean	other	clean	other
-	CLM+Transf.	2.20	4.94	2.47	5.45
-	CLM+Transf.	2.10	4.79	2.33	5.17
-	Transf.	-	-	2.0	4.6
-	LSTM	1.9	3.9	1.9	4.1
-	LSTM	2.1	4.3	1.9	3.9
LV-60k	CLM+Transf.	2.10	4.79	2.33	4.54
LV-60k	CLM+Transf.	2.00	3.65	2.09	4.11
LV-60k	4-gram+Transf.	1.85	3.26	2.10	4.01
LV-60k	LSTM	1.6	3.4	1.7	3.4
-	Transf.	1.7	4.3	2.1	4.6
LS-960	Transf.	1.8	4.7	2.1	4.8
LS-960	Transf.	1.7	3.9	2.0	4.1
LV-60k	Transf.	1.6	3.0	1.8	3.3
	Unlabeled data - - - - - - - - - - - - - - - - - -	Unlabeled dataLM-CLM+TransfCLM+TransfTransfLSTM-LSTM-LSTMLV-60kCLM+Transf.LV-60k4-gram+Transf.LV-60kLSTM-Transf.LV-60kLSTMLV-60kTransf.LV-60kLSTM-Transf.LV-60kLSTM-Transf.LS-960Transf.LS-960Transf.LV-60kTransf.	Unlabeled data LM dd clean - CLM+Transf. 2.20 - CLM+Transf. 2.10 - Transf. - - LSTM 1.9 - LSTM 2.10 - LSTM 1.9 - LSTM 2.10 LV-60k CLM+Transf. 2.00 LV-60k 4-gram+Transf. 1.85 LV-60k LSTM 1.6 - Transf. 1.7 LS-960 Transf. 1.7 LS-960 Transf. 1.7 LV-60k Transf. 1.6	$\begin{array}{c c c c c c c c c c c c c c c c c c c $	$\begin{array}{c c c c c c c c c c c c c c c c c c c $



- Phoneme Error Rates (PER) on TIMIT
 - > Obtains state-of-the-art with a reduction of 23%/29% over the next best result on the

dev/test sets

	dev PER	test PER
CNN + TD-filterbanks [59]	15.6	18.0
PASE+ [47]	-	17.2
Li-GRU + fMLLR 46	_	14.9
wav2vec [49]	12.9	14.7
vq-wav2vec [5]	9.6	11.6
This work (no LM)		
LARGE (LS-960)	7.4	8.3

- Quantization of the context network input and the targets
 - ➢ BASE pre-trained on LS-960
 - Continuous inputs enable better context representations
 - > Quantized targets lead to more robust training



	avg. WER	std.
Continuous inputs, quantized targets (Baseline)	7.97	0.02
Quantized inputs, quantized targets	12.18	0.41
Quantized inputs, continuous targets	11.18	0.16
Continuous inputs, continuous targets	8.58	0.08

Contributions

- Presents wav2vec 2.0, a self-supervised learning framework for speech pre-training
- WER of 4.8/8.2 on LibriSpeech with 10 minutes of labeled data
- Outperforms the previous best result with 100 times less labeled data on the clean 100 hour LibriSpeech setup
- Achieves a new state of the art on full LibriSpeech-960 benchmark for noisy speech (test-other)

Thank you for watching !

Presenter: Wenxin Hou