

## Self-Supervised Learning

Self-supervised learning (SSL) has achieved great successes in NLP and CV, especially for limited resource tasks.

It is because SSL utilizes a **large amount of unlabeled data** to learn universal representation, and the **universal representation** with outstanding generalizability, re-usability, and effectiveness can significantly benefit various downstream tasks.

The **common practice** of SSL:

1. Optimize the **pre-train model** with SSL objective on the large-scale unlabeled data.
2. Optimize the **downstream model** on the various downstream annotated dataset, where the input feature is the universal representation extracted from the pre-trained model.

As for SSL in **Speech**, we have witnessed great success of SSL in **content related task**, e.g. wav2vec 2.0/HuBERT SSL methods achieve state-of-the-art results in ASR.

It is unknown if we can also boost the performance of SSL for **speaker related task** (e.g. speaker verification, diarization task).

## Mask prediction loss

**Mask prediction loss**, proposed by HuBERT, is the **state-of-the-art** SSL method for **content** representation learning.

**Main idea**: conduct **iterative offline clustering** to provide target labels and perform BERT-like mask prediction loss.

**Steps**:

1. Conduct **k-means clustering** on the MFCC feature of input signals.
2. Set the clustering center of each input frame as the **pseudo target label**.
3. Train a Transformer-based model with the **mask prediction loss**, where the Transformer encoders are fed with the masked input features  $X$ , and predict the pseudo target label  $z_t$  in the masked region  $M$ :

$$\mathcal{L}_{\text{Content}} = -\sum_{t \in M} \log f(z_t | \tilde{X}, t)$$

4. Given the pre-trained model, we conduct k-means clustering on **the latent representations** generated by the pre-trained model, and start a new iteration from step 2.

## Utterance-wise contrastive loss

**Utterance-wise contrastive loss** is proposed to enhance **single-speaker** representation learning.

**Assume**: 1) each training utterance contains one active speaker. 2) Each utterance in the training batch belongs to a different speaker,

**Main idea**: the pre-trained model is optimized to **discriminate** the representations from the same utterance or the different utterance

**Methods**: For each training batch, we extract and quantize the latent representations from the internal layer of Transformer encoders, then perform **contrastive loss** over the **quantized representations** in the **mask regions**, where the representations within the same utterance are considered as positive instances, the representations from other utterances are considered as negative instances.

The **speaker information modeling loss**:

$$\mathcal{L}_{\text{Contrastive}} = -\left(\sum_{q_t^b \in \hat{Q}^b} \log \frac{\exp(\text{sim}(l_t^b, q_t^b)/\kappa)}{\exp(\text{sim}(l_t^b, q_t^b)/\kappa)+1} - \sum_{q_t^b \sim \hat{Q} \setminus \hat{Q}^b} \log \frac{1}{\exp(\text{sim}(l_t^b, q_t^b)/\kappa)+1}\right)$$

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

$$\mathcal{L}_{\text{Speaker}} = \mathcal{L}_{\text{Contrastive}} + \alpha \mathcal{L}_{\text{diversity}}$$

The final **UniSpeech-SAT pre-training loss**:

$$\mathcal{L}_{\text{UniSpeech-SAT}} = \mathcal{L}_{\text{Speaker}} + \beta \mathcal{L}_{\text{Content}}$$

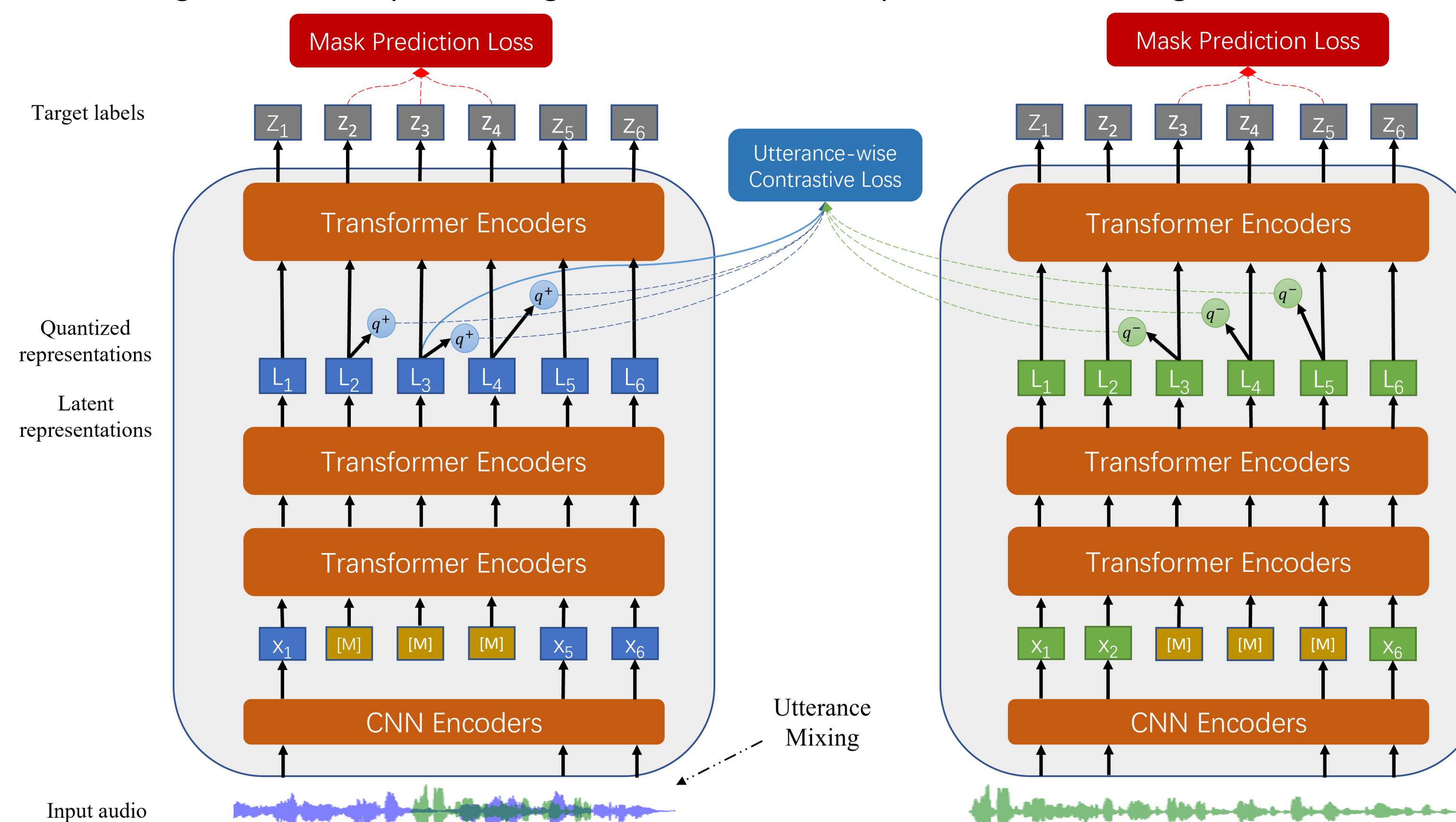
## UniSpeech-SAT

Can we apply SSL for both **content related task** and **speaker related task**?

In this work, we propose **Universal Speech** representation learning with **Speaker Aware pre-Training**, to improve existing SSL framework for speaker representation learning.

Specifically, **UniSpeech-SAT** includes the following methods:

1. Mask prediction loss (from HuBERT) -> **content** representation learning
2. Utterance-wise contrastive loss -> **single-speaker** representation learning
3. Utterance mixing augmentation -> **multi-speaker** representation learning
4. Large and diverse pre-training data -> **robust** representation learning



## Utterance mixing augmentation

**Utterance mixing augmentation** is proposed to further boost **multi-speaker** representation learning.

**Main idea**: simulate the **multi-speaker speech** for self-supervised pretraining when only single-speaker pretraining data is available.

**Methods**: For each training batch, we first randomly choose some utterances as the primary utterances. Then, for each primary utterance, we randomly choose an utterance from the same batch, crop a chunk of random length and mix it with the primary utterance in a random region.

With the utterance mixing method, the model is trained to extract **the information of the main speaker** from the mixed audio with the single-speaker information modeling loss and predict the **content information corresponding to the main speaker** with the content information modeling loss.

Note that we constrain the mixing portion in each utterance to be less than 50%, avoiding potential label permutation problem.

## Conclusion

In this work, we propose a **speaker aware pre-training method** which is complementary to current ASR oriented pre-training.

The evaluation on the SUPERB benchmark shows our universal speech representation achieves **state-of-the-art overall performance** and outperforms other baselines by a **large margin**.

This work is extended to a journal paper **WavLM** ([paper](#), [code](#)), where we sheds light on a general pre-trained model for **full stack speech processing task** and achieve 1) SOTA results on all the 10 tasks of SUPREB. 2) SOTA results on 4 typical speech tasks from different speech aspects: speaker verification, speech separation, speaker diarization and speech recognition.

## Large and diverse pre-training data

**Previous works** only use the **audiobook speech** for pre-training, which limits the generalizability of the pre-trained speech representation in diverse scenarios.

Towards **robust** speech representation learning, we **scale up unlabeled pre-training data to 94k hours** of public audios from **diverse domains**, including:

1. 10K hours Gigaspeech data, from audiobooks, podcasts and YouTube.
2. 24K hours VoxPopuli data, from European Parliament (EP) event recordings.
3. 60k hours LibriVox data, from audiobooks

## Universal Representation Evaluation with SUPERB

We evaluate our models on **SUPERB**, which is designed to provide a standard and comprehensive testbed for pretrained models on various speech tasks.

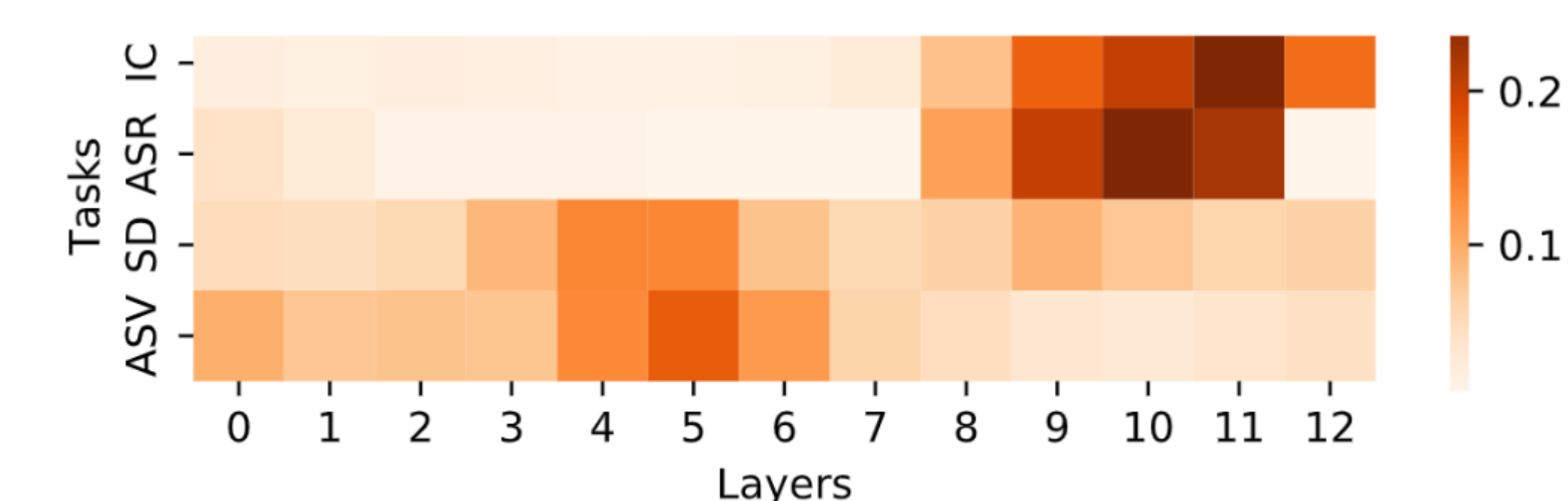
SUPERB Policies:

1. The design of **task specific layers** follows the SUPERB official implementations for each downstream task.
2. Pre-trained models are **frozen** to limit the space of the fine-tuning hyperparameter search
3. The task specific layers consume the **weighted sum results** of the hidden states extracted from each layer of the pre-trained model

## Universal Representation Evaluation Results

**Table 1**: Universal speech representation evaluation on SUPERB benchmark. The overall score is computed by ourselves: we multiply the QbE score with 100, replace each error rate score with (1 - error rate), and average the scores of all tasks.

Method	#Params	Corpus	Speaker			Content			Semantics			ParaL	Overall		
			SID	ASV	SD	PR	ASR (WER)	KS	QbE	IC	SF	ER			
FBANK	-	-	8.5E-4	9.56	10.05	82.01	23.18	15.21	8.63	0.0058	9.10	69.64	52.94	35.39	44.2
PASE+ [14]	7.83M	LS 50 hr	37.99	11.61	8.68	58.87	25.11	16.62	82.54	0.0072	29.82	62.14	60.17	57.86	57.5
APC [8]	4.11M	LS 360 hr	60.42	8.56	10.53	41.98	21.28	14.74	91.01	0.0310	74.69	70.46	50.89	59.33	67.6
VQ-APC [10]	4.63M	LS 360 hr	60.15	8.72	10.45	41.08	21.20	15.21	91.11	0.0251	74.48	68.53	52.91	59.66	67.2
NPC [11]	19.38M	LS 360 hr	55.92	9.40	9.34	43.81	20.20	13.91	88.96	0.0246	69.44	72.79	48.44	59.08	67.0
Mockingjay [12]	85.12M	LS 360 hr	32.29	11.66	10.54	70.19	22.82	15.48	83.67	6.6E-04	34.33	61.59	58.89	50.28	56.1
TERA [13]	21.33M	LS 360 hr	57.57	15.89	9.96	49.17	18.17	12.16	89.48	0.0013	58.42	67.50	54.17	56.27	64.2
modified CPC [2]	1.84M	LL 60k hr	39.63	12.86	10.38	42.54	20.18	13.53	91.88	0.0326	64.09	71.19	49.91	60.96	65.1
wav2vec [3]	32.54M	LS 960 hr	56.56	7.99	9.90	31.58	15.86	11.00	95.59	0.0485	84.92	76.37	43.71	59.79	71.5
vq-wav2vec [4]	34.15M	LS 960 hr	38.80	10.38	9.93	33.48	17.71	12.80	93.38	0.0410	85.68	77.68	41.54	58.24	69.3
wav2vec 2.0 Base [5]	95.04M	LS 960 hr	75.18	5.74	6.02	6.08	6.43	4.79	96.23	0.0233	92.35	88.30	24.77	63.43	80.3
HuBERT Base [6]	94.68M	LS 960 hr	81.42	5.11	5.88	5.41	6.42	4.79	96.30	0.0736	98.34	88.53	25.20	64.92	82.0
UniSpeech-SAT Base	94.68M	LS 960 hr	85.76	4.31	4.41	5.40	6.75	4.86	96.75	0.0927	98.58	88.98	23.56	66.04	83.0
- contrastive loss	94.68M	LS 960 hr	84.74	4.61	4.72	5.22	6.80	5.17	96.79	0.0956	98.31	88.56	24.00	65.60	82.8
- utterance mixing	94.68M	LS 960 hr	85.97	4.35	5.87	5.06	7.04	5.05	96.88	0.0866	98.10	88.50	24.52	65.97	82.7
UniSpeech-SAT Base+	94.68M	CD 94k hr	87.59	4.36	3.80	4.44	6.44	4.88	97.40	0.1125	98.84	89.76	21.75	68.48	84.0
wav2vec 2.0 Large [5]	317.38M	LL 60k hr	86.14	5.65	5.62	4.75	3.75	3.10	96.6	0.0489	95.28	87.11	27.31	65.64	82.1
HuBERT Large [6]	316.61M	LL 60k hr	90.33	5.98	5.75	3.53	3.62	2.94	95.29	0.0353	98.76	89.81	21.76	67.62	83.5
UniSpeech-SAT Large	316.61M	CD 94k hr	95.16	3.84	3.85	3.38	3.99	3.19	97.89	0.0836	99.34	92.13	18.01	70.68	85.6



**Fig. 2**: Weight Analysis.

**Table 2**: Results of UniSpeech-SAT Base+ with various mixing ratios on 94k hours training data.

Method	Ratio	Speaker	Content	Semantics	ParaL
		SD	ASR (WER)	IC	ER
HuBERT Base [6]	-	5.88	6.42	4.79	98.34
	0.0	5.04	6.39	4.76	99.24
UniSpeech-SAT Base+	0.2	3.80	6.44	4.88	98.84
	0.5	3.73	6.65	5.18	99.29