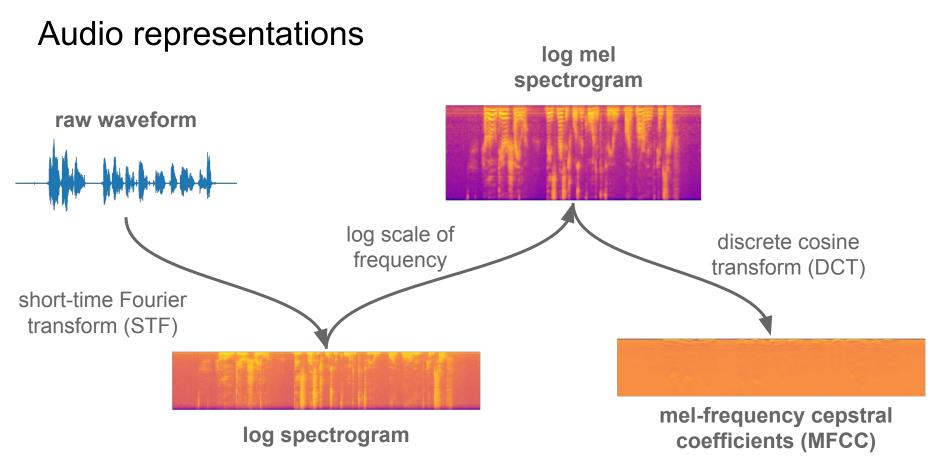
# Self-supervised learning for audio

Ildus Sadrtdinov, 07.03.23

#### What makes audio different?

- Audio signal is continuous (same as images)
- Audio signal is sequential (same as texts)
- Audio signal has high-frequency (even more than images)
- There are many informative feature representations available for audio



## Approaches to self-supervised learning

- Autoregressive (GPT-family)
- Contrastive (SimCLR, BYOL, CLIP)
- Masked modelling (\*BERT, MAE)

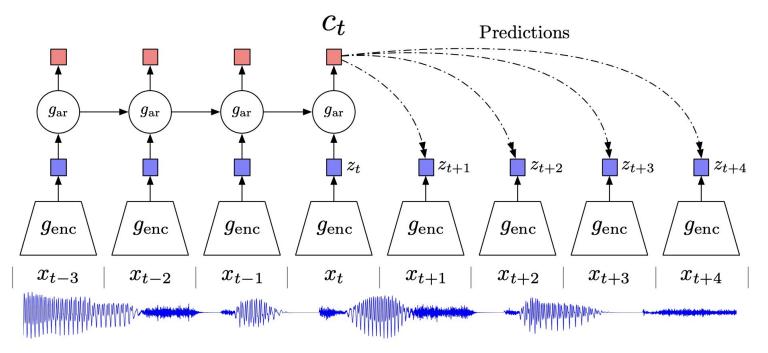
## SSL methods for audio

Model	Speech	Input format	Framework	Encoder	Loss	Inspired by
LIM [36]	/	raw waveform	(d)	SincNet	BCE, MINE or NCE loss	SimCLR
COLA [36]	X	log mel-filterbanks	(d)	EfficientNet	InfoNCE loss	SimCLR
CLAR [33]	X	raw waveform	(d)	1D ResNet-18	NT-Xent	SimCLR
(semi)		log mel-spectrogram		ResNet-18	+ cross-entropy	
Fonseca et al. [36]	X	log mel-spectrogram	(d)	ResNet, VGG, CRNN	NT-Xent loss	SimCLR
Wang et al. [88]	X	raw waveform	(d)	CNN	NT-Xent loss	SimCLR
		+ log mel-filterbanks		ResNet	+ cross-entropy	
BYOL-A [89]	X	log mel-filterbanks	(b)	CNN	MSE loss	BYOL
Speech2Vec [48]	/	mel-spectrogram	(a)	RNN	MSE loss	Word2Vec
Audio2Vec [91]	VX	MFCCs	(a)	CNN	MSE loss	Word2Vec
Carr [67]	/	MFCCs	(a)	Context-free network	Fenchel-Young loss	-
Ryan [68]	X	constant-Q transform	(a)	AlexNet	Triplet loss	-
		spectrogram				-
Mockingjay [92]	/	mel-spectrogram	(a)	Transformer	L1 loss	BERT
TERA [93]	/	log mel-spectrogram	(a)	Transformer	L1 loss	BERT
Audio ALBERT [94]	/	log mel-spectrogram	(a)	Transformer	L1 loss	BERT
DAPC [95]	/	spectrogram	(a)	Transformer	Modified MSE loss	BERT
			100 5		+ orthogonality penalty	
PASE [96]	/	raw waveform	(a)	SincNet + CNN	L1, BCE loss	BERT
PASE+ [97]	/	raw waveform	(a)	SincNet + CNN + QRNN	MSE, BCE loss	BERT
CPC [40]	/	raw waveform	(a)	ResNet + GRU	InfoNCE loss	-
CPC v2 [59]	/	raw waveform	(a)	ResNet + Masked CNN	InfoNCE loss	-
CPC2 [98]	/	raw waveform	(a)	ResNet + LSTM	InfoNCE loss	-
Wav2Vec [84]	/	raw waveform	(a)	1D CNN	Contrastive loss	-
VQ-Wav2Vec [85]	/	raw waveform	(a)	1D CNN + BERT	Contrastive loss	BERT
Wav2Vec 2.0 [81]	/	raw waveform	(a)	1D CNN + Transformer	Contrastive loss	BERT
HuBERT [99]	<b>/</b>	raw waveform	(c)	1D CNN + Transformer	Contrastive loss	BERT

#### Plan

- Contrastive Predictive Coding (CPC)
- Wav2Vec 2.0
- HUBERT
- Multi-format contrastive learning
- BYOL-A (BYOL for audio)

# Contrastive Predictive Coding (CPC)



$$f_k(x_{t+k}, c_t) \propto \frac{p(x_{t+k}|c_t)}{p(x_{t+k})}$$

$$f_k(x_{t+k}, c_t) = \exp\left(z_{t+k}^T W_k c_t\right)$$

$$f_k(x_{t+k}, c_t) \propto \frac{p(x_{t+k}|c_t)}{p(x_{t+k})} \qquad f_k(x_{t+k}, c_t) = \exp\left(z_{t+k}^T W_k c_t\right) \qquad \mathcal{L}_{\mathrm{N}} = -\frac{\mathbb{E}}{X} \left[\log \frac{f_k(x_{t+k}, c_t)}{\sum_{x_j \in X} f_k(x_j, c_t)}\right]$$

## Contrastive Predictive Coding (CPC)

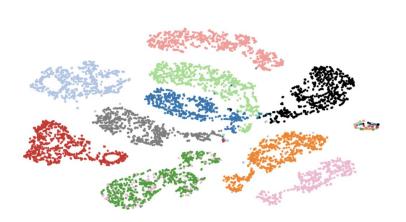


Figure 2: t-SNE visualization of audio (speech) representations for a subset of 10 speakers (out of 251). Every color represents a different speaker.

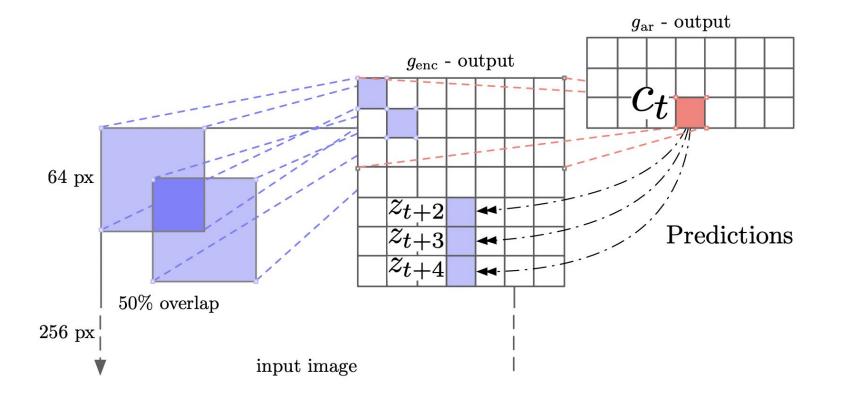
Method	ACC					
Phone classification						
Random initialization	27.6					
MFCC features	39.7					
CPC	64.6					
Supervised	74.6					
Speaker classification						
Random initialization	1.87					
MFCC features	17.6					
CPC	97.4					
Supervised	98.5					

Table 1: LibriSpeech phone and speaker classification results. For phone classification there are 41 possible classes and for speaker classification 251. All models used the same architecture and the same audio input sizes.

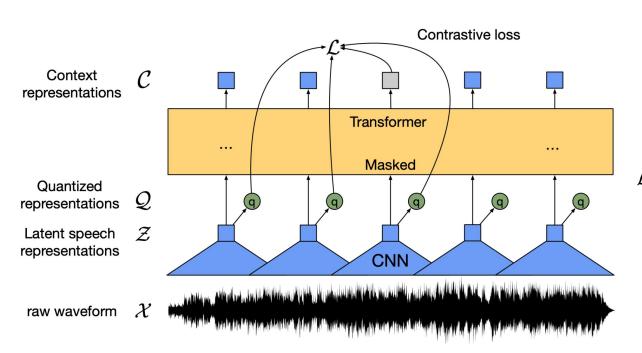
Method	ACC
#steps predicted	
2 steps	28.5
4 steps	57.6
8 steps	63.6
12 steps	64.6
16 steps	63.8
<b>Negative samples from</b>	
Mixed speaker	64.6
Same speaker	65.5
Mixed speaker (excl.)	57.3
Same speaker (excl.)	64.6
Current sequence only	65.2

Table 2: LibriSpeech phone classification ablation experiments. More details can be found in Section 3.1.

# Contrastive Predictive Coding (CPC)



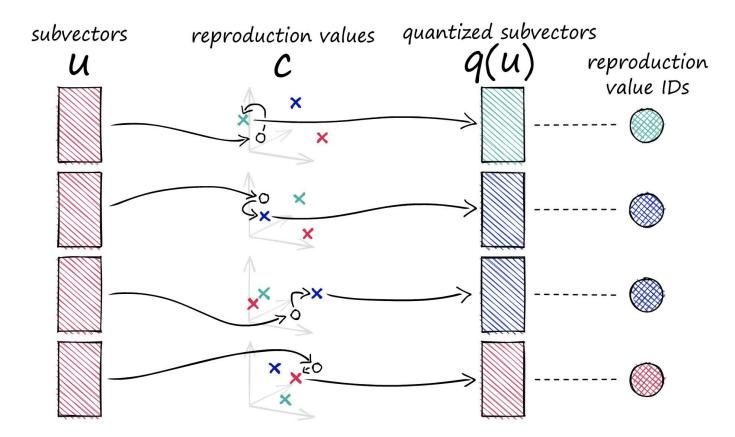
#### Wav2Vec 2.0



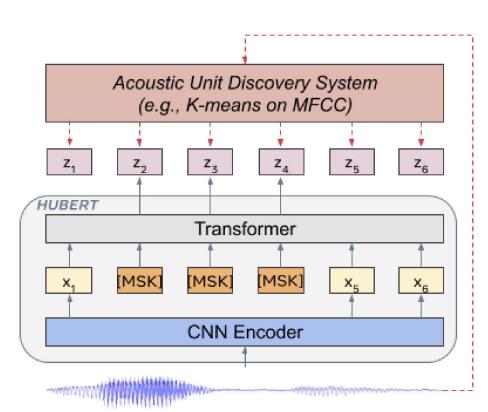
$$\mathcal{L} = \mathcal{L}_m + \alpha \mathcal{L}_d$$

$$\mathcal{L}_m = -\log \frac{\exp(sim(\mathbf{c}_t, \mathbf{q}_t)/\kappa)}{\sum_{\tilde{\mathbf{q}} \sim \mathbf{Q}_t} \exp(sim(\mathbf{c}_t, \tilde{\mathbf{q}})/\kappa)}$$

### **Product Quantization**

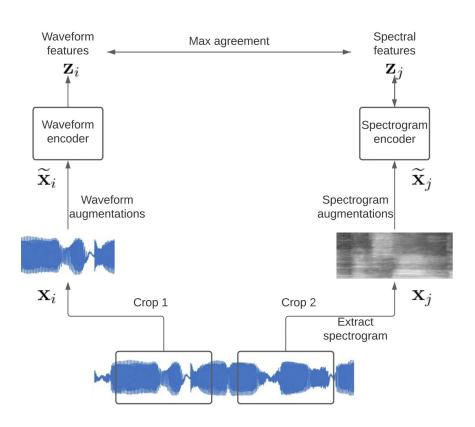


#### **HUBERT**



$$L = \alpha L_m + (1 - \alpha)L_u$$

$$p_f^{(k)}(c \mid \tilde{X}, t) = \frac{\exp(\sin(A^{(k)}o_t, e_c)/\tau)}{\sum_{c'=1}^{C} \exp(\sin(A^{(k)}o_t, e_{c'})/\tau)}$$



$$L_{i,j} = -\log \frac{\exp\left(\operatorname{sim}\left(\mathbf{z}_{i}, \mathbf{z}_{j}\right) / \tau\right)}{\sum_{k \neq i} \exp\left(\operatorname{sim}\left(\mathbf{z}_{i}, \mathbf{z}_{k}\right) / \tau\right)}$$

**Audio mixing** Small additive noise of any sort will not alter the original categories of the audio. Given two audio clips  $x_1$  and  $x_2$ , the mixed-up version is

$$\hat{\mathbf{x}}_1 = \alpha \mathbf{x}_1 + (1 - \alpha)\mathbf{x}_2 \tag{2}$$

where  $\hat{\mathbf{x}}_1$  inheritances labels from  $\mathbf{x}_1$ . In this work,  $\alpha$  is samples from  $\beta(5,2)$  distribution. This simulates various realistic noise conditions.

**Time masking** t consecutive time steps  $[t_0, t_0 + t)$  of the audio can be dropped out and it should not change the event classes, where  $t_0$  is randomly sampled. This can be applied both to raw audio and spectrograms.

**Frequency masking** A small amount of f frequency components  $[f_0, f_0 + f]$  on the spectrogram can be masked out without losing semantic information.

**Frequency shift** One can apply the truncated shift in frequency to the spectrograms by an integer number sampled from [-F, F], where F is the maximum shift size. Missing values after the shift are set to zero energy. Intuitively, this is a less expensive alternative of changing the pitch of the audio.

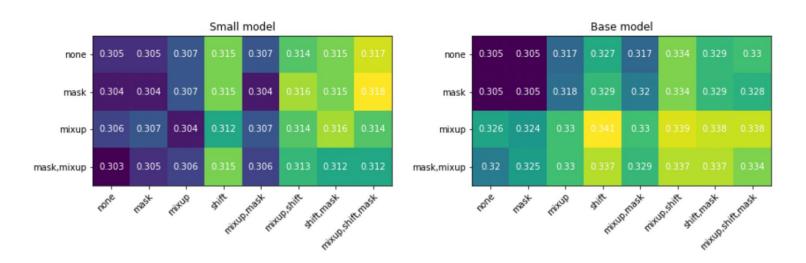
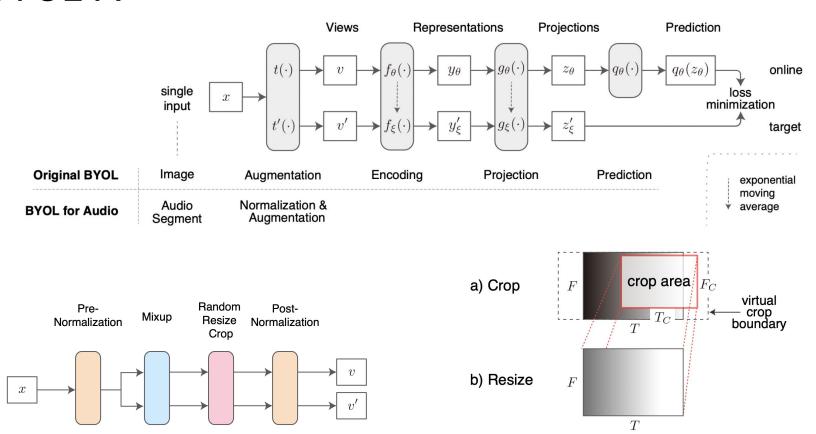


Figure 3: Validation mAP of the raw-audio-vs-log-mel models with different combinations of raw audio (along rows) and spectrogram (along columns) augmentations.

Table 3: Test performance of shallow model classification on AudioSet with fixed representations.

Model	Train inputs	<b>Eval inputs</b>	Test mAP
Triplet [20]	log-mel	log-mel	0.244
$L^{3}$ [22]	log-mel + video	log-mel	0.249
CPC [21]	waveform	waveform	0.277
$C^{3}$ [26]	log-mel + video	log-mel	0.285
MMV [28]	log-mel + video + text	log-mel	0.309
Ours	log-mel	log-mel	0.329
Ours	waveform	waveform	0.336
Ours	waveform + log-mel	log-mel	0.368
Ours	waveform $+ \log$ -mel	waveform	0.355
Ours	waveform $+ \log$ -mel	waveform + log-mel	0.376
Supervised [19]	waveform + log-mel	waveform + log-mel	0.439

#### **BYOL-A**



## **BYOL-A**

TABLE II
ABLATIONS OF BYOL-A AUGMENTATION MODULE WITH ACCURACY RESULTS, PRETRAINED WITH 1/10 AUDIOSET

Augmentation blocks used	NS	US8K	VC1	VF	SPCV2/12	SPCV2	Average	Degradation
Mixup+RRC (BYOL-A)	71.2%	77.0%	31.0%	83.1%	84.5%	87.2%	72.3%	
Mixup+Gaussian+RRC	69.5%	74.3%	25.2%	84.0%	82.8%	87.4%	70.5%	BYOL-A -1.8
Gaussian+RRC	69.7%	73.1%	29.2%	83.1%	78.0%	83.1%	69.3%	BYOL-A -3.0
RRC	69.4%	77.1%	34.5%	80.3%	71.4%	77.4%	68.4%	BYOL-A -3.9
Mixup	55.6%	69.4%	22.3%	78.3%	75.8%	82.0%	63.9%	BYOL-A -8.4
Gaussian	29.5%	31.2%	0.9%	57.9%	9.4%	10.3%	23.2%	BYOL-A -49.1

TABLE III
ABLATIONS OF NORMALIZATION BLOCKS WITH AVERAGE ACCURACY RESULTS, PRETRAINED ON 1/10 AUDIOSET

Method	Average	Degradation
BYOL-A	72.3%	
w/o Post-Norm	72.1%	BYOL-A -0.2
w/o Pre-Norm (mixup $\alpha = 0.05$ )	70.5%	BYOL-A -1.8
w/o Pre-Norm (mixup $\alpha = 0.1$ )	70.3%	BYOL-A -2.0
w/o Pre-Norm (mixup $\alpha = 0.4$ )	68.9%	BYOL-A -3.4