

Unsupervised End-to-End Learning of Discrete Linguistic Units for Voice Conversion

<u>Andy T. Liu</u>, Po-chun Hsu, Hung-yi Lee

National Taiwan University

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Highlights

- We present an **unsupervised end-to-end ASR-TTS autoencoder framework**, where we discover discrete subword units from speech without using any labels.
- Contributions:
 - Present a discrete encoding method that outperforms continuous encodings.
 - Able to dientangle speech content from speaker style automatically.
 - Achieved many-to-many voice conversion without using any parallel data.
- In our subjective and objective evaluations, we show that **VC quality is improved** when compared to continuous representations (Chao et. el).
- In ZeroSpeech 2019, the proposed method achieved **2nd place** in terms of low bitrate.

Proposed Method (1/2) - Discrete linguistic units discovery



In this **unsupervised end-to-end** manner, **discrete linguistic units** are learned and **represented as multilabel binary vectors (MBVs)**.

Proposed Method (1/2) - MBV: discrete vectors of zeros one ones



Proposed Method (1/2) - VC using the ASR-TTS autoencoder



Why Multilabel-Binary Vectors ? (MBV)

Both one-hot and MBV are discrete, however

each dimension of **one-hot** vector corresponds to a **linguistic unit** (phoneme), while each dimension of **MBV** may corresponds to a **pronunciation attribute**.



This makes MBV more data efficient than one-hot vectors as a linguistic unit, We also verified that one-hot vectors is incapable for this task.



How can the model automatically learn how to disentangle speech content form speaker identity?

If the bottleneck dimension of the ASR-Encoder is set just right, so that there is just enough capacity to encode the content, ideal VC output can be achieved.

> This is later formally proved by Qian et. al (ICML 2019), where they used RNN + downsampling to form the bottleneck, **in comparison we use discrete encodings to form this bottleneck.**

Bottleneck Visualization



Step 0: Given the trained ASR-TTS autoencoder framework shown previously



Step 1: Add a TTS-Patcher on top of it to improve VC quality



Step 2: Train the TTS-Patcher (Generator) in the framework of GAN







Experiment - Setups

• The ZeroSpeech 2019 Challenge provides two datasets:

- Development English Set
- Testing Suprise Set

• For our experiments:

We use only the English set (Voice / Unit set, no parallel data are used) for training, and evaluate our model on the English Test set.

• For ZeroSpeech 2019 Challenge:

Tune our model's hyperparameters with the Development English set, and use those hyperparameters to train our Surprise language model.



Chou et al. (2018) introduce a **classifier** on the latent code, which needs to be trained in a **GAN** setting.

However, **we did not use additional supervision on the latent code** and achieved improved results.



 Table 2: Comparison of different latent representations.

Incapable for this task **Types of encodings** Dim Acc 1024 43.3% One-hot continuous 1024 84.1% A disentangled representation continuous 128 79.9% should produce voice similar to 1024 78% continuous (with add'l loss) the target speakers and leads to continuous (with add'l loss) 128 81.3% higher classification accuracy. 92.3% 1024 Ours (MBV) Ours (MBV) 128 93.9%

Experiment - Subjective Evaluation

• Human participants are required to grade each method on a 1 to 5 scale under two measures:

• Naturalness

Whether the converted speech is human-like.

• Similarity

Whether the converted speech's has similar speaker characteristics to the target speaker. Table 3: *Results of subjective human evaluation. All methods used an encoding dimension of 1024 if not specified otherwise.*

Types of encodings	naturalness	similarity
continuous	3.80	2.14
continuous (with add'l loss)	3.21	2.58
Ours (MBV with dim 6)	1.61	1.51
Ours (MBV)	3.36	3.06
Ours (with adv. training)	2.57	3.15

Experiment - Subjective Evaluation

The setting we used to compete in the ZeroSpeech Challenge

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Surprise Set Leaderboard - ZeroSpeech 2019 Challenge



Surprise Set Leaderboard - ZeroSpeech 2019 Challenge



Experiment - Encoding Dimension Analysis

The proposed method achieves lower "bit rate" and "distinct units" with comparable ABX scores.

More analysis can be found in our paper.

 Table 4: Performance of different encoding dimensions.

-	Method	Dim	CER	BR	ABX	distinct
	Baseline	200	1.000	71.98	35.90	65
-	Cont.	1024 128	0.036	138.45 138.45	31.83 33.96	16849 16849
_		1024	0.196	138.45	32.02	16849
		512 256	0.313	138.45 138.45	32.82 32.52	16849 16849
		128	0.430	138.45	31.58	16849
	Ours	64 32	0.717 0.797	138.35 134.80	32.57 31.82	16772 14591
		16	0.887	105.96	35.62	3723
		8 7	0.998	55.97	37.71	94
		6 5	1.000 1.000	48.78 41.32	39.60 41.79	51 28

Conclusion

- The proposed encoding method MBV offers a strong bottleneck for content extraction in VC.
- As a result strong VC performance is achieved as speaker identity is eliminated from extracted encodings, while speech content is preserved.
- In the ZeroSpeech 2019 Challenge Surprise Dataset Leaderboard, the proposed method achieved outstanding results in terms of low bitrate.



Q&A