

# Towards an ASR Approach Using Acoustic and Language Models for Speech Enhancement

Khandokar Md. Nayem and Donald S. Williamson

Department of Computer Science, Indiana University, IN, USA

knayem@iu.edu, williads@Indiana.edu



- Poor performance and unwanted distortions in noisy conditions require further improvements.

## Motivation

- Healy et al. 2018 propose the ideal quantized mask (IQM), which converts SE from a regression problem to a classification problem.
- The quantized mask, however, does not consider spectral correlations along the frequency axis.
- A language model (LM) can be a better way to incorporate linguistic property of speech in an end-to-end speech approximation system.
- Hence, a spectral LM which focuses on spectral properties of human speech can be an alternate approach.



- Ten participants (9 male, 1 female) who are native English speakers over the age of 18 participated.

## **Proposed Speech Enhancement Model**

- Rightmost branch predicts the quantized class probability for the t-th time frame with two losses, a cross-entropy loss  $(\mathcal{L}_{cls})$ , and a regression loss  $(\mathcal{L}_{reg})$ .
- Left branch performs deep clustering with  $(\mathcal{L}_{DC})$  to separate speech from noise.
- Loss function is defined as,
  - $\mathcal{L}_{DC} = \|V^T V\|^2 2\|V^T Y\|^2 + \|Y^T Y\|^2$
  - $\mathcal{L} = (1 \lambda_1)\mathcal{L}_{DC} + \lambda_1\lambda_2\mathcal{L}_{cls} + \lambda_1(1 \lambda_2)\mathcal{L}_{reg}$

where V is embedding matrix, Y is source hot-vector, and  $\lambda_1$  and  $\lambda_2$  are hyperparameters.



• We constrain and quantize the unbounded continuous valued speech  $|S_{t,k}| \in$  $\{0, \Re^+\}$  by,

> $\left|S_{t,k}\right|^{q} = \mathcal{Q}_{\chi}(\mathcal{C}_{[0,r]}(\left|S_{t,k}\right|))$  $Q_{\chi}(|S_{t,k}|) = \chi \cdot \operatorname{argmin}_{i}(\{0, \chi, 2\chi, \cdots, (D-1)\chi\} - |S_{t,k}|)$

where r is the max spectrum value,  $\chi$  is the quantization step size,  $\mathcal{C}_{[0,r]}(\cdot)$  is a scaling function,  $Q_{\chi}(\cdot)$  is a quantization function which converts the range constrained magnitude spectrogram into total  $\mathcal{D}\left(=\frac{r}{r}\right)$  number of bins.

- Unlike traditional word or phoneme level-based language model (LM), we propose an alternative view of a LM, where we consider each quantization level as a word. We consider bi-gram LM, which we refer to as the Quantized Spectral Model (QSM).
- We consider a mean QSM (mQSM) where the probabilities are computed across all frequency channels and each entry (d) refers to the transition probability between two-time consecutive quantized levels.

 $mQSM = P(d_{t+1,:}|d_{t,:})$ 

• The enhanced speech sequence  $|S_{1:T,:}|^q$  is of the optimal quantized class sequence which is calculated using:

$$\left|\hat{S}_{1:T,:}\right|^{q} = \operatorname*{argmax}_{d_{1,:}} \prod_{i=1}^{r} P(M_{i,:} | d_{i,:}) P(d_{i,:} | d_{i-1,:})$$

## **Experiments and Results**

- Train and evaluate using IEEE male (single speaker, 720 utterances) and TIMIT (multiple speakers, 6300 utterances) speech corpora.
- Noise types: speech-shaped noise (SSN), cafeteria, factory, and babble.
- Trained in 3 SNR levels (-3, 0, 3 dB), tested in additional 2 SNR levels (-6, 6 dB).

		IEEE corpus		7	IMIT corpus	
	PESQ	SI-SDR	ESTOI	PESQ	SI-SDR	ESTOI
Mixture	1.86	1.8	0.53	1.81	-2.57	0.5
Chi++ <sub>IQM2</sub>	2.18	0.34	0.64	2.06	0.4	0.6
Chi++ <sub>IQM3</sub>		0.41	0.68	2.08	0.43	0.64
Chi++ <sub>IQM4</sub>		0.63	0.71	2.14	0.52	0.68
Chi++ <sub>IQM8</sub>	2.37	0.72	0.73	2.1	0.53	0.69
Chimera	2.4	0.81	0.75	2.16	0.49	0.69
Chi++ <sub>tPSA</sub>	2.46	0.84	0.76	2.25	0.74	0.72
Chi++ <sub>quant</sub>		0.82	0.75	2.2	0.63	0.67
Chi++ <sub>mQSM,greedy</sub>	2.45	0.88	0.8	2.26	0.81	0.74
Chi++ <sub>fQSM,greedy</sub>	2.46	0.93	0.82	2.27	0.84	0.74
Chi++ <sub>mQSM,bS</sub>	2.48	0.97	0.83	2.3	0.89	0.75
Chi++ <sub>mQSM,bS</sub>	2.48	1.04	0.83	2.34	0.95	0.78

Table: Average scores for each approach. Best results are shown in **bold**.

Per-frequency QSM (fQSM) is defined per-frequency transitions are stored.

 $fQSM_k = P(d_{t+1,k}|d_{t,k})$ 



#### **Conclusion and Future Work**

- Improvements in a variety of noises and SNR values prove that proposed quantized speech classification approach with an ASR-style language model successfully enhances the speech mixture and outperforms T-F masking-based approaches.
- It shows that quantized signal-approximation can be done successfully if the appropriate training target is considered.
- This approach, however, considers only bi-gram spectral models which are generated by considering only along-time transitions.
- In the future, we will explore higher-order N- gram models that consider both temporal and spectral transitions to enhance both magnitude and phase responses.

Audio, Speech and Information Retrieval (ASPIRE) research lab, <u>https://aspire.sice.indiana.edu/</u>