

DELAYGSE: A GENERATIVE SPEECH ENHANCEMENT FRAMEWORK WITH DELAYED TEXT-AWARE CONDITIONING

Xin Yuan¹, Junling Lv¹, Zezhou Xu², Xingjun Tan¹, Liangliang Li¹, Yanqiang Lei^{1*}

¹Guangzhou Shiyuan Electronic Technology Company Limited, Guangzhou, China

²Independent Researcher, Guangzhou, China

ABSTRACT

Generative speech enhancement methods often yield superior perceptual quality but are more prone than discriminative approaches to producing speech-like hallucinations under low signal-to-noise ratios (SNRs). We present DelayGSE, a multi-task generative framework that delays an automatic speech recognition (ASR) objective to suppress such artifacts, yielding a 15.8% relative WER reduction in ablations. To balance perceptual quality and semantic fidelity, we introduce an importance-aware codebook weighting for discrete-token multi-codebook modeling, where weights are computed from a joint metric of perceptual quality, intelligibility, and timbre similarity. This reduces inter-codebook training conflicts, speeds convergence, and improves both perceptual quality and intelligibility, achieving state-of-the-art results on public test sets. Optional conditioning on ground-truth text further restores speech at extremely low SNRs. Audio examples are available at <https://delaygse.github.io/>.

Index Terms— Generative Speech Enhancement, Delay-ASR Multi-Task Learning, Artifact Suppression, Importance-aware Codebook Weighting

1. INTRODUCTION

Speech enhancement improves the clarity and intelligibility of speech for real-time communication, entertainment, and human-computer interaction. Methods have advanced from masking/mapping—training networks to predict time-frequency masks or map noisy inputs to clean targets—to GAN-powered discriminative strategies that yield more natural outputs [1, 2].

Recent progress in generative modeling shifts the goal from suppressing noise to learning the distribution of clean speech, producing intelligible, high-quality audio that can approach studio-grade fidelity [3, 4, 5, 6, 7, 8, 9].

Generative enhancement falls into two families. (1) Continuous-feature models (e.g., diffusion and flow frameworks) learn reverse processes from noisy to clean speech. StoRM couples discriminative predictors with diffusion to retain fast convergence under extreme noise [3], while FlowSE

uses conditional flow matching to achieve strong results with 5 steps, enabling low latency [4]. However, increased substitution errors after Schrödinger-bridge enhancement suggest hallucinatory artifacts [5]. (2) Discrete-token approaches condition on noisy audio and autoregressively predict clean speech tokens for detokenization [6]. Low-Latency-SE aligns generation for low delay [7]; Genhancer compares AR/MaskGIT and pre-trained encoders (WavLM, Whisper) and observes cases where enhanced outputs worsen WER relative to the original noisy input [8].

Although generative models improve perceptual quality, they are more susceptible than discriminative approaches to speech-like hallucinations under low SNR or abrupt noise; motivated by discriminative multi-task remedies[10], we therefore introduce a delayed ASR objective within an LLM-style discrete-token system that employs a multi-codebook codec and applies delayed modeling to the multi-codebook LLM. The core contributions of this work are as follows:

- A delayed ASR objective that substantially reduces hallucinatory artifacts in generative enhancement.
- Importance-aware codebook weighting that derives per-codebook loss weights from each layer’s incremental impact on perceptual quality, intelligibility and timbre similarity, improving convergence and results.

2. RELATED WORK

2.1. Audio Codec

Audio LMs require a front-end codec that compresses continuous waveforms and quantizes them into discrete tokens. Neural audio codecs (NACs)—end-to-end convolutional autoencoders with residual vector quantization (RVQ)—achieve high-fidelity reconstruction at low bitrates while emitting discrete codebook indices suitable for LM modeling [11]. Recent surveys compare discrete audio tokenizers across speech, music, and general audio, relating token quality to downstream utility [12]. To ensure final audio quality and broad applicability, our generative speech enhancement system adopts DAC (44.1 kHz)¹ as the front-end audio codec.

*Corresponding author: leiyangqiang@cvte.com

¹<https://github.com/descriptinc/descript-audio-codec>

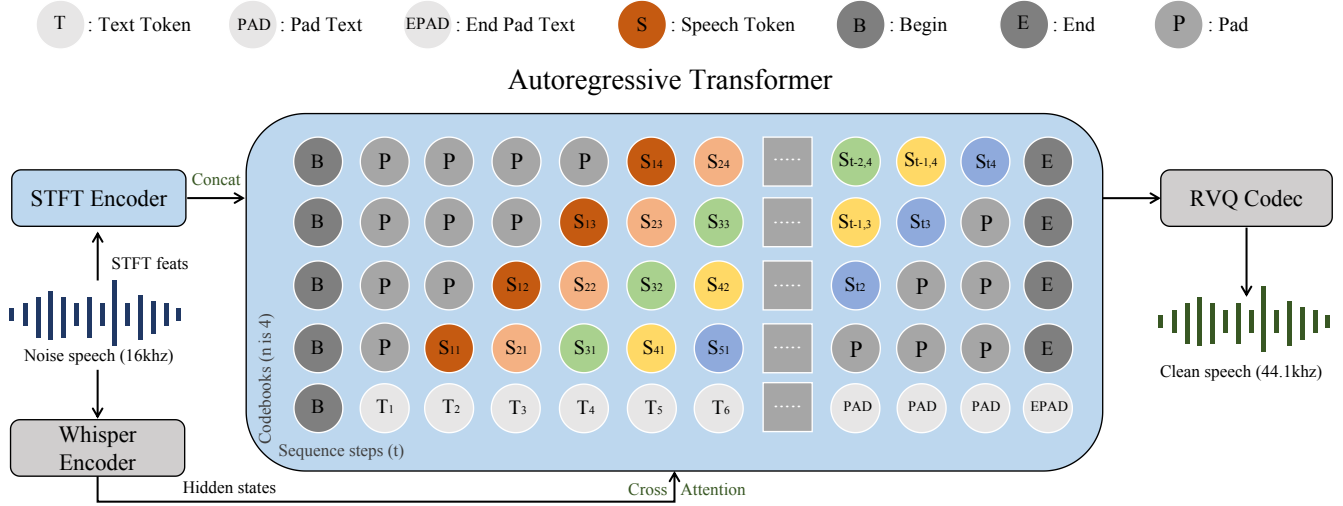


Fig. 1. DelayGSE framework. An STFT acoustic encoder and a pretrained Whisper semantic encoder condition an autoregressive Transformer that predicts RVQ tokens. We employ (i) codebook-level delay (stride-1 lower→higher conditioning) and (ii) text-first delayed ASR (predict text tokens, delay acoustic tokens by k) to suppress hallucinations. (Shown with 4 codebooks.)

2.2. Delay Pattern

Language modeling over discrete tokens derived from neural codecs typically employs multi-codebook (RVQ) parallel or hierarchical prediction. To model cross-codebook dependencies effectively within a single-stage autoregressive decoder, MusicGen introduces fixed-stride delays across codebooks, enabling higher-level codebooks to be predicted conditional on priors from lower-level ones[13]. This delay regime strikes a balance between inference parallelism and output quality and has subsequently been adopted by many speech generation models, such as Parler-TTS[14] and VoiceCraft[15].

A different form of “delay” appears in end-to-end spoken dialogue: Moshi casts dialogue as speech-to-speech generation but first predicts time-aligned text tokens as a prefix (the so-called inner monologue) before emitting acoustic tokens—i.e., within a single model, text serves as semantic planning, followed by delayed acoustic-token generation[16].

3. METHODS

3.1. Framework Overview

Our method builds upon an autoregressive Transformer framework that models the conditional distribution from degraded to clean speech. Given an input waveform x sampled at 16 kHz, the target clean speech y is first quantized into a sequence of discrete tokens $\mathbf{z} = (z_1, \dots, z_T)$ corresponding to the 44.1 kHz domain using a neural codec. The model is trained to estimate the conditional probability

$$p_{\theta}(\mathbf{z} | x) = \prod_{t=1}^T p_{\theta}(z_t | z_{<t}, \mathbf{A}, \mathbf{S}), \quad (1)$$

where $\mathbf{A} = E_{\text{STFT}}(x)$ denotes acoustic features extracted by an STFT encoder consisting of several Conformer layers, and $\mathbf{S} = E_{\text{Whisper}}(x)$ denotes semantic features obtained from a pretrained Whisper encoder².

The Transformer decoder generates the discrete tokens autoregressively. The acoustic condition \mathbf{A} is injected via concatenation with token embeddings, while the semantic condition \mathbf{S} is incorporated through cross-attention. After generation, the discrete sequence \mathbf{z} is converted back into a 44.1 kHz waveform \hat{y} by the codec decoder. The overall architecture is illustrated in Fig. 1.

Since our codec employs residual vector quantization (RVQ) with L codebooks, and $z_t^{(\ell)}$ represents the token from the ℓ -th codebook at step t , the conditional factorization is written as

$$p_{\theta}(\mathbf{z}_{1:L} | x) = \prod_{t=1}^T \prod_{\ell=1}^L p_{\theta}(z_t^{(\ell)} | \mathbf{z}_{<t, 1:L}, \mathbf{A}, \mathbf{S}). \quad (2)$$

During training, we adopt teacher forcing and minimize the cross-entropy loss over the discrete tokens:

$$\mathcal{L} = \sum_{\ell=1}^L w_{\ell} \left[- \sum_{t=1}^T \log p_{\theta}(z_t^{(\ell)*} | \mathbf{z}_{<t, 1:L}^*, \mathbf{A}, \mathbf{S}) \right], \quad (3)$$

where z^* denotes the ground-truth token and w_{ℓ} denotes the loss weight for the ℓ -th codebook.

²<https://huggingface.co/openai/whisper-large-v3>

3.2. Delay-based Modeling

To improve generation quality, we introduce two complementary delay strategies that operate at different levels of the model, i.e., *Codebook-level delay* and *Text-first multi-task delay*.

For the Codebook-level delay, we follow MusicGen, RVQ codebooks are predicted with a stride-1 offset:

$$p_{\theta}(\mathbf{z}_{1:L} | x) = \prod_{t=1}^T \prod_{\ell=1}^L p_{\theta}(z_t^{(\ell)} | \mathbf{z}_{<t,1:L}, \mathbf{z}_{t,1:\ell-1}, \mathbf{A}, \mathbf{S}). \quad (4)$$

For the Text-first multi-task delay, we are inspired by Moshi, text tokens \mathbf{c} are generated first, while speech tokens are delayed by k steps to enforce “semantic-before-acoustic”:

$$p_{\theta}(\mathbf{c}, \mathbf{z}_{1:L} | x) = p_{\theta}(\mathbf{c} | x) \cdot p_{\theta}(\mathbf{z}_{1:L} | x, \mathbf{c}), \quad (5)$$

$$p_{\theta}(\mathbf{z}_{1:L} | x, \mathbf{c}) = \prod_{t=1}^T \prod_{\ell=1}^L p_{\theta}(z_t^{(\ell)} | \mathbf{z}_{<t,1:L}, \mathbf{z}_{t,1:\ell-1}, \mathbf{c}_{\leq t+k}, \mathbf{A}, \mathbf{S}). \quad (6)$$

Finally, the overall training objective jointly optimizes both semantic and acoustic predictions through a weighted combination:

$$\mathcal{L}_{\text{total}} = \lambda_{\text{text}} \mathcal{L}_{\text{text}} + \sum_{\ell=1}^L w_{\ell} \mathcal{L}_{\text{speech}}^{(\ell)}. \quad (7)$$

3.3. Importance-aware Codebook Weighting

To quantify the unequal roles of RVQ codebook layers, we propose an *incremental contribution* method (Fig. 2). The validation audio is encoded into L layers, and a baseline score is obtained by decoding after randomizing all layers. Then, starting from the first layer, we progressively retain the true codebook while randomizing the others, and compute the reconstruction scores. The importance of each layer is defined as its incremental score gain, averaged and normalized over the validation set to serve as training weights.

4. EXPERIMENTS

4.1. Experiment Setup

The autoregressive model was initialized with Qwen 2.5 0.5B³, and the STFT encoder employed a 4-layer Conformer. Only cross-entropy loss was used for training, with the Whisper encoder frozen and the STFT encoder trained jointly with the autoregressive model. Optimization used Adam with an initial learning rate of 0.0005, 5,000-step warm-up, and cosine decay. Dynamic batching was applied, with each batch containing 40 seconds of audio, and training proceeded for 300,000 steps.

³<https://huggingface.co/Qwen/Qwen2.5-0.5B>

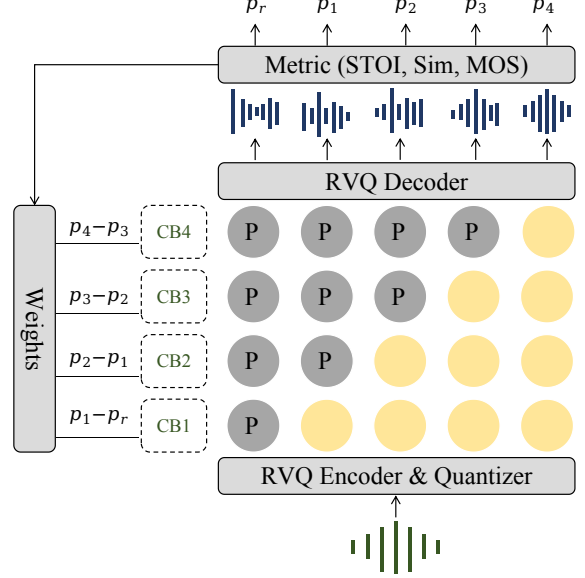


Fig. 2. Illustration of estimating the importance of each RVQ codebook layer using perceptual quality, intelligibility, and speaker similarity metrics, which are then used to weight the LLM training loss (CB denotes CodeBook).

Table 1. Statistics of Clean Datasets (30292 hours)

Data Source	Language	Duration (h)
In-house (purchased)	Chinese	165.4
HiFi-TTS[17]	English	291.6
LibriTTS-R[18]	English	585.0
In-house (cleaned)	Chinese	29250.0

Table 2. Statistics of Noise Datasets (589 hours)

Data Source	Environment	Duration (h)
WHAM[19]	(Various)	20.0
FSD50K[20]	(Various)	108.3
In-house recording	Conference	461.4

Following Section 3.3, perceptual quality, speaker similarity, and intelligibility were used to weight each codebook layer’s contribution to the loss: [0.30, 0.13, 0.12, 0.10, 0.08, 0.09, 0.09, 0.06, 0.03]. Lower-layer codebooks primarily capture semantic information, whereas higher layers encode acoustic features.

4.2. Datasets and Evaluation metrics

Our training data was acquired through simulation. As indicated in Tables 1 and 2, the clean data amount to more than 30,000 hours, while the noise data total approximately 600 hours. In addition, we produced 700,000 reverberated sam-

Table 3. Comparison of Objective Metrics (MOS, WER, SIM) Across Different Models and Datasets (NR = No Reverb, WR = With Reverb, RR = Real Recording)

	Internal			VCTK-DEMAND			DNS Challenge			URGENT 2025			
	MOS↑	WER↓	SIM↑	MOS↑	WER↓	SIM↑	MOS↑			MOS↑	WER↓		SIM↑
							NR	WR	RR		EN	ZH	
Noise	2.484	0.041	0.347	3.037	0.070	0.691	3.138	2.751	3.018	2.742	0.180	0.131	0.697
LLaSE-G1	3.817	0.411	0.250	3.526	0.121	0.620	4.093	3.951	3.725	3.451	0.353	0.507	0.596
Storm	3.353	0.724	0.131	3.592	0.074	0.632	4.001	3.142	3.494	3.229	0.371	0.572	0.559
FlowSE	3.255	0.147	0.244	3.564	0.079	<u>0.625</u>	3.960	3.298	3.509	3.167	0.334	0.305	0.598
DGSE-EW	4.007	0.120	0.298	3.728	0.086	0.571	<u>4.172</u>	4.087	<u>3.830</u>	3.650	0.242	0.191	0.562
DGSE-IW	4.043	0.099	0.344	3.738	0.072	0.607	4.202	4.146	3.752	3.746	0.218	0.156	0.661
DGSE-IW+A	<u>4.040</u>	<u>0.063</u>	<u>0.324</u>	3.726	<u>0.066</u>	0.573	4.166	4.117	3.840	3.772	<u>0.196</u>	<u>0.143</u>	0.634
DGSE-IW+AT	4.034	0.043	0.323	<u>3.732</u>	0.063	0.573	No Reference Text			<u>3.771</u>	0.153	0.104	<u>0.637</u>

ples by simulating environments of large, medium, and small meeting rooms.

The evaluation set comprised the public datasets DNS Challenge [21], VCTK-DEMAND [22], and URGENT 2025(a subset containing only Chinese and English) [23], together with an internal meeting-room dataset of 258 recordings. The internal recordings were captured in several large, medium, and small conference rooms at close-talk, 3 m, 5 m, and 8 m distances under quiet, steady-state, and transient noise conditions.

Perceptual quality was evaluated using DNSMOS P.808, while intelligibility was measured by the word error rate (WER) with the FireRedASR⁴. Speaker similarity (SIM) was measured by the cosine similarity between embeddings extracted from the enhanced audio and a registered clean reference sample⁵.

4.3. Results and Analysis

We selected three mainstream open-source generative models for comparison: **LLaSE-G1**[9], **Storm**[3], and **FlowSE**[4]. In addition, we implemented three ablation variants of our proposed framework, **DelayGSE**, as described below.

- **DGSE-EW**: DelayGSE with Equal-Weight training.
- **DGSE-IW**: DelayGSE with Importance-Weight training.
- **DGSE-IW+A**: DGSE-IW + Delayed ASR training.
- **DGSE-IW+AT**: DGSE-IW + Delayed ASR training + Inference on ground truth Text.

Table 3 reports comparisons across benchmarks and baselines. The importance-weighted variant (DGSE-IW) achieves the best or near-best perceptual quality and timbre similarity while maintaining competitive WER, outperforming continuous-feature approaches such as FlowSE and StoRM, which show weaker robustness under reverberation, and

LLaSE-G1, which produces high-quality audio but substantially higher WER. These results highlight a common trade-off in generative enhancement: models favoring waveform fidelity risk hallucinations that harm intelligibility, whereas our discrete-token, multi-codebook design better preserves phonetic and lexical cues without sacrificing perceptual quality across diverse noise and reverberation.

Ablation studies confirm that importance-aware weighting improves metrics compared to equal-weight training, reflecting the disproportionate semantic contribution of lower RVQ layers and reduced gradient conflicts across codebooks. The addition of delayed ASR supervision further lowers WER (15.8% relative reduction) and suppresses hallucinations, albeit with minor perceptual trade-offs. In contrast, ground-truth text conditioning yields an even greater WER reduction (33.1%) and restores heavily degraded speech. Stress tests with random text show that the model mainly relies on audio when signals are clear and resorts to text only under severe distortion, demonstrating selective and robust conditioning. In summary, DelayGSE attains a superior trade-off between perceptual quality and intelligibility, enhanced by complementary benefits of importance weighting and delayed ASR.

5. CONCLUSION

We introduced DelayGSE, a generative speech enhancement framework with delayed ASR supervision and importance-aware codebook weighting training. Experiments show that delayed ASR reduces hallucination-like artifacts and improves recognition accuracy, while codebook weighting accelerates convergence and enhances perceptual quality. The model also supports optional text conditioning for recovering severely degraded speech. Future work will also focus on reducing latency and extending the approach to broader multilingual and noisy scenarios.

⁴<https://github.com/FireRedTeam/FireRedASR>

⁵<https://github.com/modelscope/3D-Speaker/>

6. REFERENCES

- [1] Chengshi Zheng, Huiyong Zhang, Wenzhe Liu, Xiaoxue Luo, Andong Li, Xiaodong Li, and Brian C J Moore, “Sixty years of frequency-domain monaural speech enhancement: From traditional to deep learning methods,” *Trends in Hearing*, vol. 27, 2023.
- [2] Shengkui Zhao, Bin Ma, Karn N. Watcharasupat, and Woon-Seng Gan, “Frcrn: Boosting feature representation using frequency recurrence for monaural speech enhancement,” in *IEEE ICASSP*, 2022, pp. 9281–9285.
- [3] Jean-Marie Lemerrier, Julius Richter, Simon Welker, and Timo Gerkmann, “Storm: A diffusion-based stochastic regeneration model for speech enhancement and dereverberation,” *IEEE/ACM Trans. Audio, Speech and Lang. Proc.*, vol. 31, pp. 2724–2737, 2023.
- [4] Ziqian Wang, Zikai Liu, Xinfu Zhu, Yike Zhu, Mingshuai Liu, Jun Chen, Longshuai Xiao, Chao Weng, and Lei Xie, “FlowSE: Efficient and High-Quality Speech Enhancement via Flow Matching,” in *Interspeech*, 2025, pp. 4858–4862.
- [5] Rauf Nasretdinov, Roman Korostik, and Ante Jukić, “Robust speech recognition with schrödinger bridge-based speech enhancement,” in *IEEE ICASSP*, 2025, pp. 1–5.
- [6] Ziqian Wang, Xinfu Zhu, Zihan Zhang, YuanJun Lv, Ning Jiang, Guoqing Zhao, and Lei Xie, “Selm: Speech enhancement using discrete tokens and language models,” in *IEEE ICASSP*, 2024, pp. 11561–11565.
- [7] Huaying Xue, Xiulian Peng, and Yan Lu, “Low-latency speech enhancement via speech token generation,” in *IEEE ICASSP*, 2024, pp. 661–665.
- [8] Haici Yang, Jiaqi Su, Minje Kim, and Zeyu Jin, “Genhancer: High-fidelity speech enhancement via generative modeling on discrete codec tokens,” in *Interspeech*, 2024, pp. 1170–1174.
- [9] Boyi Kang, Xinfu Zhu, Zihan Zhang, Zhen Ye, Mingshuai Liu, Ziqian Wang, Yike Zhu, Guobin Ma, Jun Chen, Longshuai Xiao, Chao Weng, Wei Xue, and Lei Xie, “LLaSE-g1: Incentivizing generalization capability for LLaMA-based speech enhancement,” in *Proceedings of the 63rd ACL*, 2025, pp. 13292–13305.
- [10] You-Jin Li, Rong Chao, Borching Su, and Yu Tsao, “Speech enhancement with map-based training for robust asr,” in *IEEE ICASSP*, 2025, pp. 1–5.
- [11] Rithesh Kumar, Prem Seetharaman, Alejandro Luebs, Ishaan Kumar, and Kundan Kumar, “High-fidelity audio compression with improved rvqgan,” in *NeurIPS*, 2023, pp. 27980–27993.
- [12] Pooneh Mousavi, Gallil Maimon, Adel Moumen, Darius Petermann, Jiatong Shi, et al., “Discrete audio tokens: More than a survey!,” 2025, arXiv:2506.10274.
- [13] Jade Copet, Felix Kreuk, Itai Gat, Tal Remez, David Kant, et al., “Simple and controllable music generation,” in *NeurIPS*, 2023, pp. 47704–47720.
- [14] Yoach Lacombe, Vaibhav Srivastav, and Sanchit Gandhi, “Parler-tts,” <https://github.com/huggingface/parler-tts>, 2024.
- [15] Puyuan Peng, Po-Yao Huang, Shang-Wen Li, Abdelrahman Mohamed, and David Harwath, “VoiceCraft: Zero-shot speech editing and text-to-speech in the wild,” in *Proceedings of the 62nd ACL*, 2024, pp. 12442–12462.
- [16] Alexandre Défossez, Laurent Mazaré, Manu Orsini, Amélie Royer, Patrick Pérez, et al., “Moshi: a speech-text foundation model for real-time dialogue,” 2024, arXiv:2410.00037.
- [17] Evelina Bakhturina, Vitaly Lavrukhin, Boris Ginsburg, and Yang Zhang, “Hi-fi multi-speaker english tts dataset,” in *Interspeech*, 2021, pp. 2776–2780.
- [18] Yuma Koizumi, Heiga Zen, Shigeki Karita, Yifan Ding, Kohei Yatabe, et al., “Libritts-r: A restored multi-speaker text-to-speech corpus,” in *Interspeech*, 2023, pp. 5496–5500.
- [19] Gordon Wichern, Joe Antognini, Michael Flynn, Licheng Richard Zhu, Emmett McQuinn, et al., “Wham!: Extending speech separation to noisy environments,” in *Interspeech*, 2019, pp. 1368–1372.
- [20] Eduardo Fonseca, Xavier Favory, Jordi Pons, Frederic Font, and Xavier Serra, “Fsd50k: An open dataset of human-labeled sound events,” *IEEE/ACM Trans. Audio, Speech and Lang. Proc.*, vol. 30, pp. 829–852, 2021.
- [21] Maximilian Strake, Bruno Defraene, Kristoff Fluyt, Wouter Tirry, and Tim Fingscheidt, “Interspeech 2020 deep noise suppression challenge: A fully convolutional recurrent network (fcrn) for joint dereverberation and denoising,” in *Interspeech*, 2020, pp. 2467–2471.
- [22] Christophe Veaux, Junichi Yamagishi, and Simon King, “The voice bank corpus: Design, collection and data analysis of a large regional accent speech database,” in *International Conference Oriental COCOSA*, 2013, pp. 1–4.
- [23] Wangyou Zhang, Robin Scheibler, Kohei Saijo, Samuele Cornell, Chenda Li, et al., “Urgent challenge: Universality, robustness, and generalizability for speech enhancement,” in *Interspeech*, 2024, pp. 4868–4872.