
IMPROVING DIRECT PERSIAN–ENGLISH SPEECH-TO-SPEECH TRANSLATION WITH DISCRETE UNITS AND SYNTHETIC PARALLEL DATA

Sina Rashidi, Hossein Sameti
 Computer Engineering Department
 Sharif University of Technology
 Tehran, Iran
 {sina.rashidi, sameti}@sharif.edu

ABSTRACT

Direct speech-to-speech translation (S2ST), in which all components are trained jointly, is an attractive alternative to cascaded systems because it offers a simpler pipeline and lower inference latency. However, direct S2ST models require large amounts of parallel speech data in the source and target languages, which are rarely available for low-resource languages such as Persian. This paper presents a direct S2ST system for translating Persian speech into English speech, as well as a pipeline for synthetic parallel Persian–English speech generation.. The model comprises three components: (1) a conformer-based encoder, initialized from self-supervised pretraining, maps source speech to high-level acoustic representations; (2) a causal transformer decoder with relative position multi-head attention translates these representations into discrete target speech units; (3) a unit-based neural vocoder generates waveforms from the predicted discrete units. To mitigate the data scarcity problem, we construct a new Persian–English parallel speech corpus by translating Persian speech transcriptions into English using a large language model and then synthesizing the corresponding English speech with a state-of-the-art zero-shot text-to-speech system. The resulting corpus increases the amount of available parallel speech by roughly a factor of six. On the Persian–English portion of the CVSS corpus, the proposed model achieves improvement of 4.6 ASR BLEU with the synthetic data over direct baselines. These results indicate that combining self-supervised pretraining, discrete speech units, and synthetic parallel data is effective for improving direct S2ST in low-resource language pairs such as Persian–English.¹

Keywords Speech-to-Speech Translation · Direct Speech-to-Speech Translation · Discrete Speech Units · Audio Dubbing · Low-Resource Languages

1 Introduction

Speech-to-speech translation (S2ST) systems facilitate cross-lingual communication by converting speech in a source language directly into synthesized speech in a target language. Traditional S2ST pipelines decompose this task into three independent modules: automatic speech recognition (ASR), text-based machine translation (MT), and text-to-speech (TTS) synthesis [1]. While modular systems have been highly successful, they suffer from several drawbacks such as error propagation across modules, increased latency due to multiple processing stages, and multiplied training data requirements across subsystems.

Direct, or end-to-end, S2ST models (Figure 1) address these issues by learning a single model that maps source speech to target speech. By avoiding intermediate text representations, direct systems can reduce inference time and simplify deployment [2], which is particularly appealing for audio dubbing scenarios where many utterances must be processed

¹Code is publicly available at <https://github.com/sinarashidi/S2ST-Transformer>

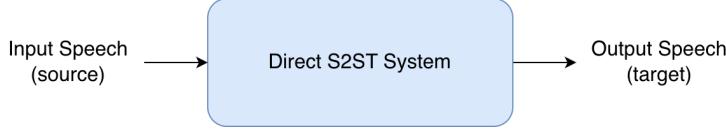


Figure 1: Demonstration of a direct speech-to-speech translation pipeline

with limited delay. However, direct S2ST models require large amounts of parallel speech data, and existing corpora are heavily skewed toward a small set of high-resource languages such as English and Spanish [2, 3, 4].

For low-resource languages like Persian, collecting large volumes of parallel speech is expensive and time-consuming. In addition, S2ST models for audio dubbing must not only translate content accurately but also generate natural-sounding speech in the target language. This combination of translation quality and speech naturalness makes data scarcity an even more pressing challenge.

In this work we focus on direct Persian–English S2ST for audio dubbing. Building on recent advances in discrete speech units, self-supervised speech representation learning, and unit-based neural vocoders, we propose a model that translates Persian speech into English speech without relying on intermediate text. Furthermore, we construct a new synthetic Persian–English parallel speech corpus using large language models (LLMs) and neural TTS, substantially expanding the amount of paired data available for this low-resource language pair.

Our contributions are as follows:

1. We present a direct S2ST architecture for Persian–English dubbing that combines a pretrained conformer-based encoder, a causal transformer decoder with relative positional attention, and a unit-based neural vocoder.
2. We design a data generation pipeline that creates a large synthetic parallel speech corpus by translating existing Persian speech transcriptions with a large language model and synthesizing English speech using a zero-shot TTS system. The resulting corpus increases the amount of available parallel speech data by about six times compared to existing resources.
3. We conduct a comprehensive experimental study on the CVSS Fa–En benchmark and the newly constructed corpus, comparing our system to strong direct baselines such as Translatotron and a speech-to-unit model.
4. We analyze the impact of self-supervised pretraining, discrete units, and synthetic data, showing that their combination yields consistent improvements in ASR BLEU for Persian–English S2ST.

2 Background

Direct S2ST has evolved rapidly over the past few years. Early works on speech translation focused on cascaded systems in which ASR, MT, and TTS were trained independently [1]. More recently, sequence-to-sequence architectures have enabled direct speech-to-text translation, and subsequently direct S2ST, by learning a unified mapping from source speech to target output.

Spectrogram-based direct S2ST models such as Translatotron [2] and Translatotron 2 [3] use an encoder–decoder architecture that predicts target mel-spectrograms, which are then converted to waveform by a neural vocoder. These models have demonstrated that direct S2ST can rival cascaded systems in high-resource settings and can preserve speaker characteristics, which is valuable for dubbing applications. However, operating entirely in the spectrogram domain can make training unstable and data-hungry, and spectrogram prediction models may struggle to disentangle linguistic content from speaker and prosodic information.

A complementary line of work replaces continuous spectrograms with discrete speech units learned by self-supervised speech models followed by vector quantization or clustering [4]. Discrete units serve as a compact, language-agnostic representation of speech, enabling modular combinations of unit-based ASR, MT, and TTS components. For S2ST, discrete units allow the translation network to focus on symbolic sequences, while a separate unit vocoder handles waveform synthesis. Recent systems such as direct S2ST with discrete units [4] and two-pass architectures like UnitY [5] demonstrate strong performance by leveraging self-supervised pretraining and large-scale unit discovery. Self-supervised representation learning for speech, exemplified by wav2vec 2.0 [6] and related models, has proven highly effective in low-resource scenarios. By pretraining on unlabeled speech and fine-tuning on task-specific data, these models substantially reduce the amount of labeled data needed to achieve reliable performance. Data augmentation techniques such as SpecAugment [7] further improve robustness by applying time warping and time/frequency masking to spectrograms during training.

Despite these advances, relatively little work has focused on low-resource language pairs such as Persian–English, and even fewer on direct S2ST without intermediate text. This paper aims to bridge that gap by combining self-supervised pretraining, discrete units, and synthetic parallel data in a single direct S2ST system tailored to Persian–English dubbing.

3 Datasets

Our experiments rely on three types of speech corpora and a synthetic Persian–English parallel corpus. We use the Persian portion of Common Voice [8], a large crowdsourced corpus of read speech containing thousands of speakers and substantial hours of audio, both for self-supervised pretraining and as the starting point for building the synthetic parallel corpus. As our main benchmark for evaluating translation quality, we adopt the Persian–English subset of CVSS [9], a multilingual S2ST corpus constructed by aligning speech-to-text translation data and synthesizing target speech with a neural TTS system. In addition, for training and adapting the neural unit vocoder used in our pipeline, we employ LJSpeech [10], an English TTS dataset consisting of high-quality recordings by a single speaker.

3.1 Synthetic Persian–English Parallel Speech Corpus

To alleviate the lack of Persian–English parallel speech, we construct a new synthetic corpus in three stages.

1. Collecting Persian speech and transcriptions. We start from the Persian portion of Common Voice, which provides utterances along with sentence-level transcriptions. After basic cleaning and filtering, we obtain a set of high-quality Persian speech segments with reliable text.
2. Translating Persian transcriptions to English with a large language model. Each Persian transcription is translated into English using a state-of-the-art large language model (GPT-4o). The model is prompted to produce fluent, semantically faithful translations suitable for spoken dialogue. This yields synthetic English text that is aligned at the utterance level with the original Persian speech.
3. Synthesizing English speech with neural TTS. The translated English sentences are converted into speech using a state-of-the-art zero-shot TTS model, VoiceCraft [11]. The result is an English speech utterance aligned with the original Persian speech at the utterance level. By repeating this process for all items, we obtain a large synthetic Persian–English parallel speech corpus.

Overall, the constructed corpus contains roughly six times more Persian–English parallel speech than the existing CVSS Fa-En subset. This substantial increase in data volume is crucial for training data-hungry direct S2ST models.

4 Model

The model follows a direct S2ST paradigm based on discrete speech units. It (Figure 2) consists of three main components: (1) a conformer-based speech encoder [12] initialized from self-supervised pretraining on Persian speech; (2) a causal transformer decoder with relative position multi-head attention, which maps encoder representations to discrete target speech units; (3) a neural vocoder that converts sequences of discrete units into English speech waveforms. The entire model is trained to maximize the likelihood of the target unit sequence given the source speech, without accessing intermediate source or target text. The unit vocoder is trained separately on English data and kept fixed during S2ST training.

4.1 Speech Encoder

The encoder is based on wav2vec 2.0, with its Transformer encoder layers replaced by Conformer layers. We first perform self-supervised pretraining on the Persian portion of Common Voice using a contrastive objective similar to wav2vec 2.0. During pretraining, random spans of the input are masked, and the model learns to distinguish the true latent representation of each masked region from a set of negative examples. This encourages the encoder to capture robust, high-level acoustic features that generalize well across tasks and domains.

After pretraining, the encoder is fine-tuned as part of the direct S2ST model. The combination of self-supervised pretraining and supervised fine-tuning improves data efficiency and robustness, especially in the low-resource Persian setting.

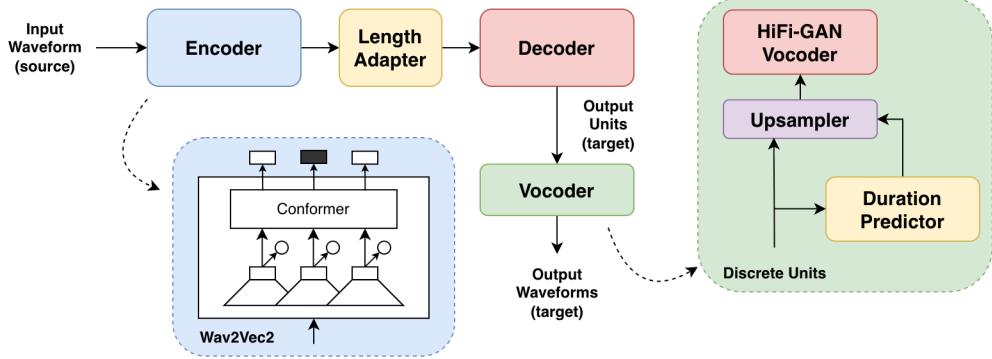


Figure 2: Architecture of the proposed model

4.2 Discrete Target Units and Decoder

On the target side, we represent English speech using discrete units obtained from a unit-based speech model trained on English data. Following prior work, we derive these units by (i) learning frame-level latent representations with HubERT [13], a self-supervised speech model and (ii) applying clustering to obtain a finite codebook of units. Each target utterance is thus mapped to a sequence of discrete indices.

The S2ST decoder is a causal transformer that autoregressively predicts the next unit given the previous units and the encoder outputs. We use relative position multi-head attention [14] to better model the temporal structure of speech and to allow the decoder to generalize to different utterance lengths. The decoder is trained with a cross-entropy objective over unit indices, optionally combined with label smoothing.

4.3 Length Adapter and Alignment

Due to differences in speaking rate and phonotactics between Persian and English, the lengths of source and target sequences may differ substantially. We incorporate a length adapter that bridges the temporal resolution between encoder representations and target units. The adapter uses simple convolutional and subsampling operations to reduce the time dimension while preserving relevant information for translation. This module is trained jointly with the encoder and decoder.

4.4 Unit Vocoder

The final component of the system is the neural unit vocoder that converts sequences of discrete units into English speech waveforms. The vocoder is trained on English speech and unit sequences derived from the LJSpeech corpus. In this work, we adopt HiFi-GAN [15] that has been shown to produce natural-sounding speech from discrete units. Once trained, the vocoder is frozen and reused for all S2ST experiments.

4.5 Data Augmentation

To improve robustness and reduce overfitting, we apply SpecAugment. Time shifting, frequency masking, and time masking are randomly applied to the encoder inputs. These perturbations encourage the model to focus on invariant acoustic cues and have been shown to improve performance in both ASR and S2ST tasks.

5 Experiments

5.1 Training Setup

All speech signals are resampled to 16 kHz. We use dynamic batch construction based on the total number of frames to efficiently utilize GPU memory. The main S2ST model is trained for 40 epochs with an initial learning rate of 2.5e-4 for the decoder and a smaller learning rate of 1e-5 for the pretrained encoder, using an optimizer with warmup and decay. The training loss is computed as the average cross-entropy over all target units in each mini-batch. We train two versions of the proposed model:

1. CVSS-only: trained on the CVSS Fa–En subset.

Table 1: ASR BLEU (Fa → En)

Model	CVSS-Only	CVSS+Synthetic
Translatotron	1.4	6.9
Translatotron 2 + Pre-training	2.4 3.8	- -
Speech-to-unit + Pre-training	1.6 2.8	11.8 13.2
Our Proposed model	4.1	17.8

2. CVSS+Synthetic: trained on the combination of CVSS and the newly constructed synthetic Persian–English corpus.

5.2 Baselines

We compare our approach to several direct S2ST baselines: (1) Translatotron: a spectrogram-based direct S2ST model that predicts target mel-spectrograms from source speech and uses a neural vocoder to synthesize waveform; (2) Translatotron 2 (+pretraining): an improved version with stronger pretraining and architectural refinements; (3) Speech-to-unit (+pretraining): a model that directly predicts discrete target units from source speech but uses a simpler encoder–decoder architecture than our proposed system. These baselines represent strong direct S2ST systems and provide a meaningful point of comparison for our contributions in encoder pretraining, unit modeling, and data augmentation.

We evaluate translation quality using ASR BLEU. In this metric, the synthesized English speech is fed to an English ASR model trained on LibriSpeech. The resulting transcripts are compared against the reference English transcripts using the BLEU metric. ASR BLEU correlates with translation quality while accounting for both translation and synthesis errors.

We also report qualitative observations on speech naturalness and alignment based on listening to model outputs, but the main quantitative comparisons are in terms of ASR BLEU.

5.3 Results

Table 1 summarizes the ASR BLEU scores on the CVSS Fa–En evaluation set for training with both CVSS-only and CVSS+Synthetic datasets. The proposed model achieves the highest ASR BLEU among all systems, outperforming the best baseline (Translatotron 2 with pretraining) by 0.3 BLEU. This gain is achieved despite using the same training data, indicating that the combination of a pretrained conformer encoder, discrete target units, and a tailored decoder architecture provides a more effective mapping from Persian speech to English speech units.

When we augment training with the synthetic Persian–English corpus, the proposed model benefits substantially from the increased data. Relative to direct baselines trained only on existing datasets, our system achieves gains of 0.3 ASR BLEU without synthetic data and 4.6 ASR BLEU with synthetic data. These improvements highlight the effectiveness of large-scale synthetic data for low-resource S2ST, particularly when combined with self-supervised pretraining and unit-based modeling.

Qualitatively, we observe that models trained with synthetic data produce more fluent and semantically complete translations, especially for longer utterances and less frequent phrases. The additional coverage of vocabulary and sentence patterns provided by the synthetic corpus appears to reduce omissions and mistranslations in the generated speech.

6 Discussion and Conclusion

In this work, we presented a direct Persian–English S2ST system designed for audio dubbing that integrates a self-supervised pretrained Conformer encoder, a discrete-unit Transformer decoder with relative positional attention, and a neural unit-based vocoder. To mitigate the scarcity of parallel data, we constructed a synthetic Persian–English corpus using large language model translation and neural TTS synthesis, increasing the amount of parallel speech by approximately six times. On the CVSS Fa–En benchmark, our model achieves up to a 4.6 BLEU improvement over strong direct baselines when trained with the synthetic corpus. These results demonstrate that the combination of

self-supervised pretraining, discrete units, and synthetic data is an effective strategy for improving S2ST in low-resource language pairs.

Our results highlight key factors that contribute to effective direct Persian–English speech-to-speech translation (S2ST) in low-resource conditions. First, self-supervised pretraining of the encoder significantly enhances model performance. The pretrained Conformer encoder captures robust acoustic and phonetic patterns from large amounts of unlabeled Persian speech, enabling the fine-tuned S2ST system to generalize more effectively than models trained from scratch.

Second, discrete acoustic units provide a strong intermediate representation for S2ST. By decoupling linguistic content modeling from waveform synthesis, the encoder–decoder can focus on cross-lingual mappings, while the unit vocoder specializes in producing natural speech. This division of responsibilities allows unit-based S2ST systems to match or surpass spectrogram-based approaches when combined with appropriate pretraining.

References

- [1] Wolfgang Wahlster. *Verbmobil: foundations of speech-to-speech translation*. Springer Science & Business Media, 2013.
- [2] Ye Jia, Ron J Weiss, Fadi Biadsy, Wolfgang Macherey, Melvin Johnson, Zhifeng Chen, and Yonghui Wu. Direct speech-to-speech translation with a sequence-to-sequence model. *arXiv preprint arXiv:1904.06037*, 2019.
- [3] Ye Jia, Michelle Tadmor Ramanovich, Tal Remez, and Roi Pomerantz. Translatotron 2: High-quality direct speech-to-speech translation with voice preservation. In *International Conference on Machine Learning*, pages 10120–10134. PMLR, 2022.
- [4] Ann Lee, Peng-Jen Chen, Changhan Wang, Jiatao Gu, Sravya Popuri, Xutai Ma, Adam Polyak, Yossi Adi, Qing He, Yun Tang, et al. Direct speech-to-speech translation with discrete units. *arXiv preprint arXiv:2107.05604*, 2021.
- [5] Hirofumi Inaguma, Sravya Popuri, Ilia Kulikov, Peng-Jen Chen, Changhan Wang, Yu-An Chung, Yun Tang, Ann Lee, Shinji Watanabe, and Juan Pino. Unity: Two-pass direct speech-to-speech translation with discrete units. *arXiv preprint arXiv:2212.08055*, 2022.
- [6] Alexei Baevski, Yuhao Zhou, Abdelrahman Mohamed, and Michael Auli. wav2vec 2.0: A framework for self-supervised learning of speech representations. *Advances in neural information processing systems*, 33:12449–12460, 2020.
- [7] Daniel S Park, William Chan, Yu Zhang, Chung-Cheng Chiu, Barret Zoph, Ekin D Cubuk, and Quoc V Le. SpecAugment: A simple data augmentation method for automatic speech recognition. *arXiv preprint arXiv:1904.08779*, 2019.
- [8] Rosana Ardila, Megan Branson, Kelly Davis, Michael Henretty, Michael Kohler, Josh Meyer, Reuben Morais, Lindsay Saunders, Francis M Tyers, and Gregor Weber. Common voice: A massively-multilingual speech corpus. *arXiv preprint arXiv:1912.06670*, 2019.
- [9] Ye Jia, Michelle Tadmor Ramanovich, Quan Wang, and Heiga Zen. Cvss corpus and massively multilingual speech-to-speech translation. *arXiv preprint arXiv:2201.03713*, 2022.
- [10] Keith Ito and Linda Johnson. The lj speech dataset. <https://keithito.com/LJ-Speech-Dataset/>, 2017.
- [11] Puyuan Peng, Po-Yao Huang, Daniel Li, Abdelrahman Mohamed, and David Harwath. Voicecraft: Zero-shot speech editing and text-to-speech in the wild. *arXiv preprint arXiv:2403.16973*, 2024.
- [12] Anmol Gulati, James Qin, Chung-Cheng Chiu, Niki Parmar, Yu Zhang, Jiahui Yu, Wei Han, Shibo Wang, Zhengdong Zhang, Yonghui Wu, et al. Conformer: Convolution-augmented transformer for speech recognition. *arXiv preprint arXiv:2005.08100*, 2020.
- [13] Wei-Ning Hsu, Benjamin Bolte, Yao-Hung Hubert Tsai, Kushal Lakhotia, Ruslan Salakhutdinov, and Abdelrahman Mohamed. Hubert: Self-supervised speech representation learning by masked prediction of hidden units. *IEEE/ACM transactions on audio, speech, and language processing*, 29:3451–3460, 2021.
- [14] Peter Shaw, Jakob Uszkoreit, and Ashish Vaswani. Self-attention with relative position representations. *arXiv preprint arXiv:1803.02155*, 2018.
- [15] Jungil Kong, Jaehyeon Kim, and Jaekyoung Bae. Hifi-gan: Generative adversarial networks for efficient and high fidelity speech synthesis. *Advances in neural information processing systems*, 33:17022–17033, 2020.