

# Leveraging Parameter-Efficient Transfer Learning for Multi-Lingual Text-to-Speech Adaptation

Anonymous ACL submission

## Abstract

Different languages have distinct phonetic systems and vary in their prosodic features making it challenging to develop a Text-to-Speech (TTS) model that can effectively synthesise speech in multilingual settings. Furthermore, TTS architecture needs to be both efficient enough to capture nuances in multiple languages and efficient enough to be practical for deployment. The standard approach is to build transformer based model such as SpeechT5 and train it on large multilingual dataset. As the size of these models grow the conventional fine-tuning for adapting these model becomes impractical due to heavy computational cost. In this paper, we propose to integrate parameter-efficient transfer learning (PETL) methods such as adapters and hypernetwork with TTS architecture for multilingual speech synthesis. Notably, in our experiments PETL methods able to achieve comparable or even better performance compared to full fine-tuning with only  $\sim 2.5\%$  tunable parameters<sup>1</sup>.

## 1 Introduction

Multilingual speech synthesis, generating speech in multiple languages from text input, represents a major advancement in speech processing with wide-reaching implications for global communication (Tan et al., 2021; Mehrish et al., 2023b). Unlike single-language systems, multilingual architectures break linguistic barriers, transforming education, entertainment, healthcare, and customer service by facilitating seamless communication across languages (Marais et al., 2020; Seong et al., 2021; Le et al., 2024; Panda et al., 2020).

Current multilingual TTS architectures face challenges (Nuthakki et al., 2023; Kaur and Singh, 2023), including the complexity of modeling diverse linguistic structures, phonetic variations, and

prosodic features across languages. Resource constraints, such as the availability of multilingual corpora and linguistic expertise, can impede model development, particularly for low-resource languages or underrepresented dialects (Tan et al., 2021; Mehrish et al., 2023b). Addressing these challenges requires concerted efforts in data collection, model development, and evaluation.

The advancement of architectural designs, coupled with pre-training models such as SpeechT5 (Ao et al., 2021), reflects challenges similar to those encountered in NLP. Achieving optimal performance through fine-tuning these models for diverse downstream tasks or domain adaptations requires substantial task-specific datasets. Moreover, fine-tuning all model parameters necessitates significant memory resources allocated to each task. With limited data available for various underrepresented languages, full fine-tuning can further lead to poor generalization. Researchers have sought solutions to these challenges through the exploration of PETL methods (Li et al., 2023; Oh et al., 2023; Chen et al., 2023; Sathyendra et al., 2022; Vanderreydt et al., 2023; Le et al., 2021). However, their investigation remains limited for TTS adaptation.

In this paper, we extend PETL approaches to the multilingual TTS, focusing on adapter (Houlsby et al., 2019) and Hyper-Network (Üstün et al., 2022). We pioneer the hyper-networks for multilingual TTS adaptation and introduce the *Multi-Conditioned HyperGenerator* for multilingual TTS. Our major contributions include: (1) Regular & Dynamic Adapters: We embed language-specific parameters into SpeechT5 using regular adapters and explore a hyper-network to generate these parameters, referred to as HyperGenerator. (2) Parameter Efficiency: We achieve comparable or superior performance to full fine-tuning using only about 2.44% of the parameters. (3) Improved Zero-shot Performance: HyperGenerator outperforms full finetuning and regular adapters on an unseen

<sup>1</sup>The code and samples are available at: <https://anonymous.4open.science/r/multilingualTTS-BA4C>

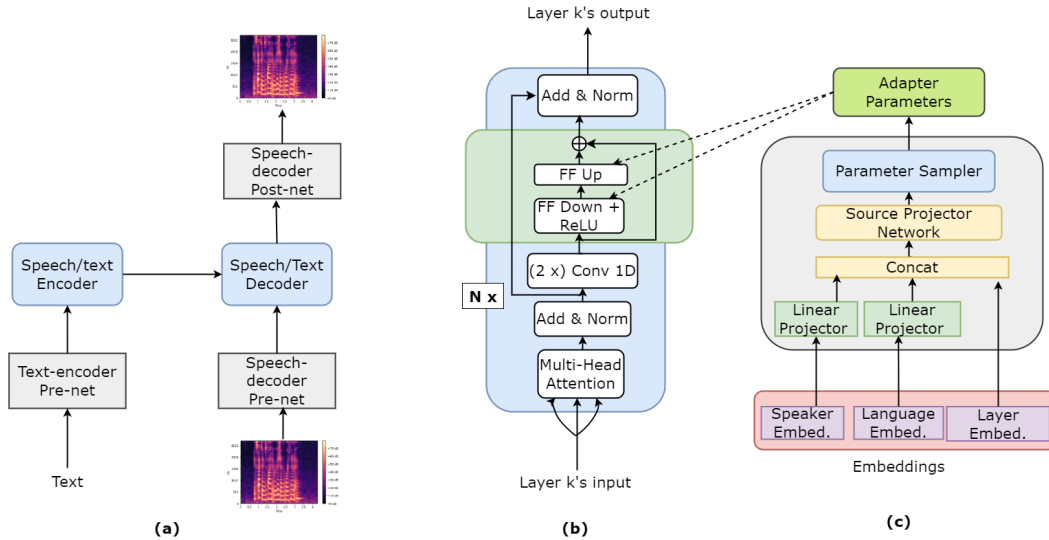


Figure 1: (a) SpeechT5 TTS architecture. (b) Encoder/decoder architecture with adapter. (c) HyperGenerator architecture.

language with the same parameter count.

## 2 Related work

Research on PETL methods for large multilingual pretrained models like XLSR (Vanderreydt et al., 2023) has gained significant attention, notably in Automatic Speech Recognition (ASR) (Fu et al., 2023; Zhang et al., 2023; Yu et al., 2023; Yang et al., 2023; Shi and Kawahara, 2024). Li et al. (Li et al., 2023) propose a benchmark utilizing XLSR-53 (Conneau et al., 2020), employing PETL such as regular adapters (Pfeiffer et al., 2020), prefix tuning (Li and Liang, 2021), and LoRA (Hu et al., 2021). Le et al. (Le et al., 2021) and Zhao et al. (Zhao et al., 2022) explore multilingual neural machine translation, focusing on lightweight adapter tuning. Morioka et al. (Morioka et al., 2022) advocate for integrating regular adapters with TTS models for few-shot speaker adaptation, while Mehrish et al. (Mehrish et al., 2023a) introduce a mixture of experts for low-resource speaker adaptation.

## 3 Methodology

### 3.1 Base Model Architecture: SpeechT5

SpeechT5 (Ao et al., 2021) merges NLP and speech synthesis techniques, extending the transformer-based T5 architecture (Raffel et al., 2020). It integrates self-attention mechanisms and CNNs to capture both temporal dependencies and spectral features in speech. By pre-training on large-scale speech corpora and fine-tuning on specific datasets, SpeechT5 excels in tasks such as speech recognition, TTS, and speech translation.

### 3.2 Adapter

In this work, we integrate language-specific parameters using adapter modules, commonly employed in the NLP for multilingual or multi-task scenarios (Ansell et al., 2021). Following the formulation of (Houlsby et al., 2019), we insert one adapter block after each convolutional block of every transformer module in the SpeechT5 model as shown in Figure 1. Each adapter module, with fewer parameters compared to the main network (SpeechT5), down-projects the input to a lower-dimensional space, applies a non-linearity, and then up-projects back to the original dimensions. A residual connection is added to produce the final output. During language adaptation, only the adapter parameters are updated while keeping the main network frozen.

### 3.3 HyperGenerator

HyperGenerator consists of a hyper network (Üstün et al., 2022) that generates the weights of all adapter modules. As depicted in Figure 1, a single hyper-network is employed to create adapters for multiple languages and layers, with conditioning on  $(s, l, p)$ , where  $s$  denotes speaker embeddings,  $l$  represents the target language, and  $p$  indicates the encoder or decoder layer ID. This method, unlike traditional adapters, promotes cross-language and cross-layer information sharing, enabling the hyper-network to efficiently distribute its capacity among them. By adapting parameters based on speaker characteristics and language specifics, the hyper-network augments the effectiveness of adapters. Further-

Model	de		fr		fi		hu		nl		avg		Params
	MCD	CER	MCD	CER	MCD	CER	MCD	CER	MCD	CER	MCD	CER	
Finetune Multilingual w/o Pretrain	4.88	7.19	4.91	14.40	4.76	15.16	4.93	16.29	5.46	10.56	4.99	12.72	144M(100%)
Finetune Multilingual w/ Pretrain	4.87	6.66	4.95	11.82	4.82	12.64	5.00	15.52	5.51	10.28	5.03	11.38	144M(100%)
Adapter Multilingual w/ Pretrain	4.75	<b>6.30</b>	4.93	11.81	4.75	9.58	4.92	16.11	5.40	10.37	4.95	10.83	3.56M(2.47%)
HyperAdapter Multilingual w/ Pretrain	4.78	6.52	5.04	15.50	4.67	<b>7.05</b>	<b>4.87</b>	<b>13.13</b>	<b>5.36</b>	10.96	4.94	<b>10.63</b>	3.52M(2.44%)
Finetune Monolingual w/ Pretrain	4.86	6.44	4.85	<b>10.80</b>	4.67	15.09	4.90	15.54	5.53	<b>8.47</b>	4.96	11.27	144M(100%)
Adapter Monolingual w/ Pretrain	4.77	6.82	4.82	14.32	<b>4.63</b>	9.41	4.94	14.53	5.36	14.22	<b>4.90</b>	11.86	3.56M(2.47%)
HyperAdapter Monolingual w/ Pretrain	<b>4.71</b>	7.94	<b>4.82</b>	14.66	4.77	13.47	4.93	14.00	5.40	13.21	4.93	12.66	3.52M(2.44%)

Table 1: Evaluation results for *seen* languages along with the percentage of parameters updated during training.

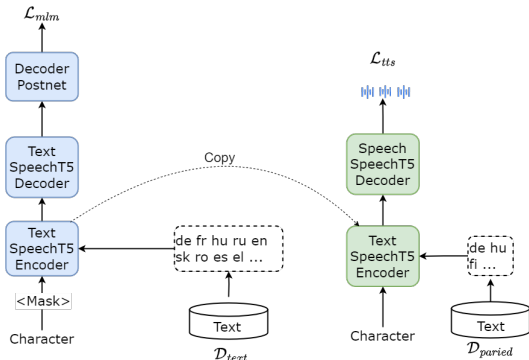


Figure 2: Multilingual Masked Text Pretraining. Where  $\mathcal{L}_{mlm}$  and  $\mathcal{L}_{tts}$  is mask language modeling and reconstruction loss respectively.

more, to ensure network efficiency, we utilize a shared hyper-network to generate adapter parameters across all layers within the TTS backbone, further conditioning it with the layer ID  $p$ .

## 4 Experimental Setup

### 4.1 Baseline and Dataset

We developed a baselines with the following 3 configurations for comparing the performance of the adapter and HyperGenerator with full fine-tuning. **Monolingual:** We finetune the SpeechT5 individually for each language, uniquely optimizing its parameters to enhance speech synthesis performance. **Multilingual:** We finetune the SpeechT5 with diverse speech data from multiple languages. This improves the model’s ability to understand and generate speech across various linguistic contexts, capturing cross-lingual patterns, phonetic variations, and language-specific features.

**Multilingual Masked Text Pretraining:** Multilingual models like multilingual BERT (Devlin et al., 2018) have demonstrated strong cross-lingual transfer capabilities in NLP tasks. Leveraging multilingual pre-training improves generalization to other languages without specific target data. In this settings, we extend MLM pre-training to SpeechT5 to

enhance pronunciation and prosody transfer. The left side of Figure 2 illustrates the unsupervised pre-training of SpeechT5’s text encoder and decoder using text-only data  $\mathcal{D}_{text}$  with MLM. The pre-trained text encoder is then integrated into the TTS pipeline, as shown on the right side of Figure 2, and trained on paired speech-text data  $\mathcal{D}_{paired}$ .

For fine-tuning using monolingual and multilingual configuration, as discussed in Section 4.1, we leverage German (de), French (fr), Finnish (fi), Hungarian (hu), and Dutch (nl)—as the five *seen* European languages from the CSS10 dataset (Park and Mulc, 2019). To evaluate zero-shot performance, we use Spanish (es) as an *unseen* language. For Multilingual Masked Text Pretraining, we utilize transcripts from VoxPopuli (Wang et al., 2021), M-AILABS (Bakhturina et al., 2021), and CSS10 (Park and Mulc, 2019) to pre-train the SpeechT5 text encoder-decoder for a character-based masked language modeling task.

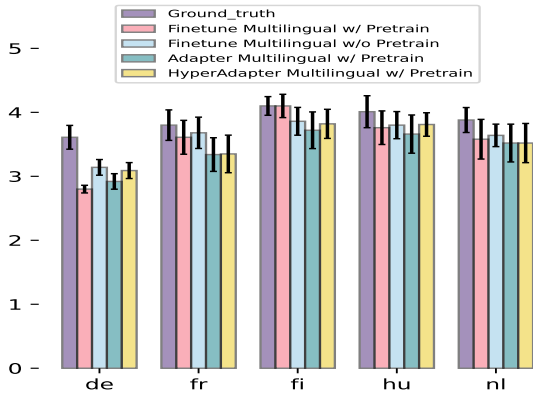
### 4.2 Training and Evaluation

We follow the data partition outlined in (Saeki et al., 2023). We use pretrained checkpoint<sup>2</sup> of SpeechT5 for all experiments. Speaker embeddings<sup>3</sup> are set at a dimension of 256, while language embeddings are initialized using pretrained weights from lang2vec (Littell et al., 2017). The layer embedding dimension is set at 64. The bottleneck dimension for adapters is 128, whereas for HyperGenerator, is 32 for ensuring the same number of parameters across both architectures. We employed MCD (Kominek et al., 2008) and assess intelligibility using Character Error Rates (CERs) computed with the multilingual ASR (Radford et al., 2023)<sup>4</sup> as objective metrics. Furthermore, to evaluate naturalness, we conducted listening tests to calculate the MOS of synthesized speech. We recruited five native speaker via Amazon Mechanical Turk (AMT)

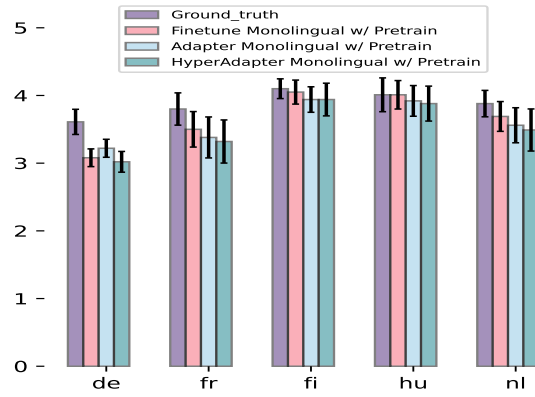
<sup>2</sup>[https://huggingface.co/microsoft/speecht5\\_tts](https://huggingface.co/microsoft/speecht5_tts)

<sup>3</sup>Pretrained speaker verification model (Wan et al., 2018).

<sup>4</sup><https://github.com/openai/whisper>



(a) Multilingual performance



(b) Monolingual performance

Figure 3: Subjective evaluation on naturalness: MOS score

for each of the languages.

Model	es	
	MCD	CER
Finetune Multilingual w/o Pretrain	5.75	39.32
Finetune Multilingual w/ Pretrain	5.87	34.80
Adapter Multilingual w/ Pretrain	5.39	45.94
HyperAdapter Multilingual w/ Pretrain	<b>5.28</b>	<b>18.79</b>

Table 2: Evaluation results for *unseen* language (es).

## 5 Results

### 5.1 Objective and Subjective Evaluation

Table 1 shows that *Finetune Multilingual with Pre-training* outperforms *Finetune Multilingual without Pretraining* due to multilingual masked text pretraining. Both *Adapter* and *HyperGenerator* achieve similar or better performance than full fine-tuning with significantly fewer parameters. Full fine-tuning updates 144M parameters, while *Adapter* and *HyperGenerator* use only 3.56M and 3.52M parameters, respectively, with *HyperGenerator* showing superior performance in multilingual settings with text pretraining.

While the performance gain for *HyperGenerator* with *seen* languages is modest, it shows promise for zero-shot multilingual speech synthesis. Table 2 shows that *HyperGenerator* achieved a CER of 18.79% for Spanish, significantly lower than the over 30% CER for Multilingual Fine-tuning and adapter-based approaches. This highlights *HyperGenerator*’s dynamic adaptability and potential for efficient and accurate zero-shot synthesis. Figure 4 further demonstrates this, as speech from the same language clusters together, indicating *HyperGenerator*’s ability to adjust parameters based on

language, unlike static adapters.

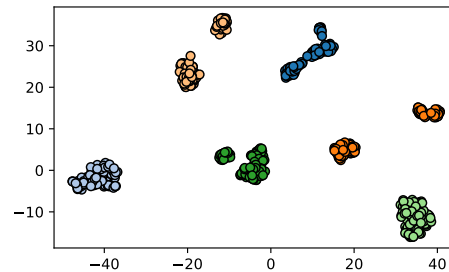


Figure 4: t-SNE plot of *HyperGenerator* parameters for 6 languages from the CSS10 test set, with same colors denoting speech samples from the same language.

MOS results (Figure 3a and Figure 3b) indicate that *Adapters* and *HyperGenerator* perform as well as or better than full fine-tuning in multilingual contexts. *HyperGenerator* consistently achieved the highest scores, with values of 3.09 for *de* and 3.81 for *hu*, demonstrating superior naturalness. Similar trends in monolingual scenarios highlight *HyperGenerator*’s effectiveness in generating high-quality speech across different languages.

## 6 Conclusion

In this paper, we advance multilingual speech synthesis using PETL methods like adapter fine-tuning, achieving SOTA performance with fewer parameters. Introducing regular and dynamic adapters with a hyper-network enhances efficiency and zero-shot performance. Future work could optimize adapters for specific languages, improve cross-lingual transfer learning, and reduce model complexity while maintaining high performance

## 252 **Limitations**

253 While our proposed approach shows promise in  
254 advancing multilingual TTS synthesis, there are  
255 several limitations that must be acknowledged. Ad-  
256 dressing these challenges will be crucial for enhanc-  
257 ing the robustness and applicability of our methods  
258 across a wider range of languages and use cases.  
259 The key limitations are as follows:

- 260 • The performance of hypernetworks and  
261 adapters can vary greatly depending on the  
262 hyperparameters used. Adjusting these set-  
263 tings for each language and task is often a  
264 complex and time-consuming process that re-  
265 quires significant computational resources.
- 266 • Languages such as Russian and Greek use  
267 scripts that differ from the Latin alphabet, like  
268 Cyrillic and Greek scripts, respectively. These  
269 scripts have unique rules for how letters and  
270 sounds are represented. The current PETL  
271 methods might not fully address these differ-  
272 ences, resulting in lower quality speech syn-  
273 thesis for these languages.
- 274 • Symbolic languages, such as Chinese and  
275 Japanese, have unique linguistic elements like  
276 Chinese logograms and Japanese kana, as  
277 well as complex grammatical structures in lan-  
278 guages like Russian and Greek. The proposed  
279 architecture in its current form can struggle  
280 to handle these diverse features effectively,  
281 which means modification to these adapta-  
282 tion techniques are needed to improve per-  
283 formance.

## 284 **Potential Risk**

285 While our research aims to advance multilingual  
286 TTS technology, it is crucial to acknowledge the  
287 potential risks associated with such systems. We  
288 will discuss some associated risks as follows :

- 289 • **Malicious Use and Disinformation:** The abil-  
290 ity of TTS systems to generate highly realistic  
291 speech could be used to create disinformation.  
292 This could lead to the spread of false informa-  
293 tion, manipulation of opinion, and erosion of  
294 trust in digital content.
- 295 • Our research utilizes the publicly available  
296 CSS10 dataset, however utilization of person-  
297 alized data to adapt these models can have  
298 the risk of privacy violations. Therefore it

is important to follow best data management  
practices that do not inadvertently compro-  
mise privacy.

- TTS systems are vulnerable to adversarial at-  
tacks where small perturbations to the input  
can lead to significant changes in the output.  
Although the proposed framework is robust  
to noise, the necessary security measures and  
continuous testing of the system against po-  
tential attacks can enhance resilience.

## **Ethical Considerations**

The TTS system could be used to produce mis-  
leading or harmful content. For instance, synthe-  
sized speech could be exploited to create fake audio  
recordings that mimic real individuals, potentially  
leading to misinformation or fraud. Additionally,  
the accessibility of TTS technology might raise  
concerns about the unauthorized use of voices, in-  
fringing on personal privacy and intellectual prop-  
erty rights.

## **References**

- Alan Ansell, Edoardo Maria Ponti, Jonas Pfeiffer, Se-  
bastian Ruder, Goran Glavaš, Ivan Vulić, and Anna  
Korhonen. 2021. **MAD-G: Multilingual adapter gen-  
eration for efficient cross-lingual transfer**. In *Find-  
ings of the Association for Computational Linguis-  
tics: EMNLP 2021*, pages 4762–4781, Punta Cana,  
Dominican Republic. Association for Computational  
Linguistics.
- Junyi Ao, Rui Wang, Long Zhou, Chengyi Wang,  
Shuo Ren, Yu Wu, Shujie Liu, Tom Ko, Qing Li,  
Yu Zhang, et al. 2021. **Speech5: Unified-modal  
encoder-decoder pre-training for spoken language  
processing**. *arXiv preprint arXiv:2110.07205*.
- Evelina Bakhturina, Vitaly Lavrukhin, Boris Ginsburg,  
and Yang Zhang. 2021. **Hi-fi multi-speaker english  
tts dataset**. *arXiv preprint arXiv:2104.01497*.
- Nanxin Chen, Izhak Shafran, Yu Zhang, Chung-Cheng  
Chiu, Hagen Soltau, James Qin, and Yonghui Wu.  
2023. **Efficient adapters for giant speech models**.  
*arXiv preprint arXiv:2306.08131*.
- Alexis Conneau, Alexei Baevski, Ronan Collobert,  
Abdelrahman Mohamed, and Michael Auli. 2020.  
**Unsupervised cross-lingual representation learn-  
ing for speech recognition**. *arXiv preprint  
arXiv:2006.13979*.
- Jacob Devlin, Ming-Wei Chang, Kenton Lee, and  
Kristina Toutanova. 2018. **BERT: pre-training of  
deep bidirectional transformers for language under-  
standing**. *CoRR*, abs/1810.04805.

349	Zihao Fu, Haoran Yang, Anthony Man-Cho So, Wai Lam, Lidong Bing, and Nigel Collier. 2023. On the effectiveness of parameter-efficient fine-tuning. In <i>Proceedings of the AAAI Conference on Artificial Intelligence</i> , volume 37, pages 12799–12807.	402
350		403
351		404
352		405
353		406
354	Neil Houlsby, Andrei Giurgiu, Stanislaw Jastrzebski, Bruna Morrone, Quentin De Laroussilhe, Andrea Gesmundo, Mona Attariyan, and Sylvain Gelly. 2019. Parameter-efficient transfer learning for nlp. In <i>International conference on machine learning</i> , pages 2790–2799. PMLR.	407
355		408
356		409
357		410
358		
359		
360	Edward J Hu, Phillip Wallis, Zeyuan Allen-Zhu, Yuanzhi Li, Shean Wang, Lu Wang, Weizhu Chen, et al. 2021. Lora: Low-rank adaptation of large language models. In <i>ICLR</i> .	411
361		412
362		413
363		414
364	Navdeep Kaur and Parminder Singh. 2023. Conventional and contemporary approaches used in text to speech synthesis: A review. <i>Artificial Intelligence Review</i> , 56(7):5837–5880.	415
365		416
366		417
367		418
368	John Kominek, Tanja Schultz, and Alan W Black. 2008. Synthesizer voice quality of new languages calibrated with mean mel cepstral distortion. In <i>SLTU</i> , pages 63–68.	419
369		420
370		421
371		
372	Hang Le, Juan Pino, Changhan Wang, Jiatao Gu, Didier Schwab, and Laurent Besacier. 2021. Lightweight adapter tuning for multilingual speech translation. <i>arXiv preprint arXiv:2106.01463</i> .	422
373		423
374		424
375		425
376	Matthew Le, Apoorv Vyas, Bowen Shi, Brian Karrer, Leda Sari, Rashel Moritz, Mary Williamson, Vimal Manohar, Yossi Adi, Jay Mahadeokar, et al. 2024. Voicebox: Text-guided multilingual universal speech generation at scale. <i>NeurIPS</i> , 36.	426
377		427
378		428
379		429
380		430
381	Xiang Lisa Li and Percy Liang. 2021. Prefix-tuning: Optimizing continuous prompts for generation. <i>arXiv preprint arXiv:2101.00190</i> .	431
382		432
383		433
384	Yingting Li, Ambuj Mehrish, Rishabh Bhardwaj, Navonil Majumder, Bo Cheng, Shuai Zhao, Amir Zadeh, Rada Mihalcea, and Soujanya Poria. 2023. Evaluating parameter-efficient transfer learning approaches on sure benchmark for speech understanding. In <i>ICASSP 2023</i> , pages 1–5. IEEE.	434
385		435
386		436
387		437
388		438
389		
390	Patrick Littell, David R Mortensen, Ke Lin, Katherine Kairis, Carlisle Turner, and Lori Levin. 2017. Uriel and lang2vec: Representing languages as typological, geographical, and phylogenetic vectors. In <i>EACL</i> , volume 2, pages 8–14.	439
391		440
392		441
393		442
394		
395	Laurette Marais, Johannes A Louw, Jaco Badenhorst, Karen Calteaux, Ilana Wilken, Nina Van Niekerk, and Glenn Stein. 2020. Awezamed: A multilingual, multimodal speech-to-speech translation application for maternal health care. In <i>2020 IEEE 23rd International Conference on Information Fusion (FUSION)</i> , pages 1–8. IEEE.	443
396		444
397		445
398		446
399		447
400		448
401		
	Ambuj Mehrish, Abhinav Ramesh Kashyap, Li Yingting, Navonil Majumder, and Soujanya Poria. 2023a. Adaptermix: Exploring the efficacy of mixture of adapters for low-resource tts adaptation. <i>arXiv preprint arXiv:2305.18028</i> .	449
		450
		451
		452
		453
		454
	Ambuj Mehrish, Navonil Majumder, Rishabh Bhardwaj, Rada Mihalcea, and Soujanya Poria. 2023b. A review of deep learning techniques for speech processing. <i>Information Fusion</i> , page 101869.	
	Nobuyuki Morioka, Heiga Zen, Nanxin Chen, Yu Zhang, and Yifan Ding. 2022. Residual adapters for few-shot text-to-speech speaker adaptation. <i>arXiv preprint arXiv:2210.15868</i> .	
	Praveena Nuthakki, Madhavi Katamaneni, Chandra Sekhar JN, Kumari Gubbala, Bullarao Domathoti, Venkata Rao Maddumala, and Kumar Raja Jeti. 2023. Deep learning based multilingual speech synthesis using multi feature fusion methods. <i>ACM Transactions on Asian and Low-Resource Language Information Processing</i> .	
	Changdae Oh, Hyeji Hwang, Hee-young Lee, YongTaek Lim, Geunyoung Jung, Jiyoung Jung, Hosik Choi, and Kyungwoo Song. 2023. Blackvip: Black-box visual prompting for robust transfer learning. In <i>CVPR</i> , pages 24224–24235.	
	Soumya Priyadarsini Panda, Ajit Kumar Nayak, and Satyananda Champati Rai. 2020. A survey on speech synthesis techniques in indian languages. <i>Multimedia Systems</i> , 26:453–478.	
	Kyubyong Park and Thomas Mulc. 2019. Css10: A collection of single speaker speech datasets for 10 languages. <i>arXiv preprint arXiv:1903.11269</i> .	
	Jonas Pfeiffer, Andreas Rücklé, Clifton Poth, Aishwarya Kamath, Ivan Vulić, Sebastian Ruder, Kyunghyun Cho, and Iryna Gurevych. 2020. Adapterhub: A framework for adapting transformers. <i>arXiv preprint arXiv:2007.07779</i> .	
	Alec Radford, Jong Wook Kim, Tao Xu, Greg Brockman, Christine McLeavey, and Ilya Sutskever. 2023. Robust speech recognition via large-scale weak supervision. In <i>ICML</i> , pages 28492–28518. PMLR.	
	Colin Raffel, Noam Shazeer, Adam Roberts, Katherine Lee, Sharan Narang, Michael Matena, Yanqi Zhou, Wei Li, and Peter J Liu. 2020. Exploring the limits of transfer learning with a unified text-to-text transformer. <i>Journal of machine learning research</i> , 21(140):1–67.	
	Takaaki Saeki, Soumi Maiti, Xinjian Li, Shinji Watanabe, Shinnosuke Takamichi, and Hiroshi Saruwatari. 2023. Learning to speak from text: Zero-shot multilingual text-to-speech with unsupervised text pretraining. In <i>International Joint Conference on Artificial Intelligence</i> .	

455	Kanthashree Mysore Sathyendra, Thejaswi Muniyappa, Feng-Ju Chang, Jing Liu, Jinru Su, Grant P. Strimel, Athanasios Mouchtaris, and Siegfried Kunzmann. 2022. Contextual adapters for personalized speech recognition in neural transducers. In <i>ICASSP 2022</i> , pages 8537–8541.	Jinming Zhao, Hao Yang, Ehsan Shareghi, and Ghulamreza Haffari. 2022. M-adapter: Modality adaptation for end-to-end speech-to-text translation. <i>arXiv preprint arXiv:2207.00952</i> .	510 511 512 513
461	Jiwon Seong, WooKey Lee, and Suan Lee. 2021. Multilingual speech synthesis for voice cloning. In <i>2021 IEEE International Conference on Big Data and Smart Computing</i> , pages 313–316. IEEE.		
465	Hao Shi and Tatsuya Kawahara. 2024. Exploration of adapter for noise robust automatic speech recognition. <i>arXiv preprint arXiv:2402.18275</i> .		
468	Xu Tan, Tao Qin, Frank Soong, and Tie-Yan Liu. 2021. A survey on neural speech synthesis. <i>arXiv preprint arXiv:2106.15561</i> .		
471	Ahmet Üstün, Arianna Bisazza, Gosse Bouma, Gertjan van Noord, and Sebastian Ruder. 2022. Hyper-x: A unified hypernetwork for multi-task multilingual transfer. In <i>Proceedings of the 2022 Conference on Empirical Methods in Natural Language Processing</i> , pages 7934–7949.		
477	Geoffroy Vanderreydt, Amrutha Prasad, Driss Khalil, Srikanth Madikeri, Kris Demuynck, and Petr Motlicek. 2023. Parameter-efficient tuning with adaptive bottlenecks for automatic speech recognition. In <i>ASRU-2023</i> , pages 1–7. IEEE.		
482	Li Wan, Quan Wang, Alan Papir, and Ignacio Lopez Moreno. 2018. Generalized end-to-end loss for speaker verification. In <i>ICASSP-2018</i> , pages 4879–4883. IEEE.		
486	Changhan Wang, Morgane Riviere, Ann Lee, Anne Wu, Chaitanya Talnikar, Daniel Haziza, Mary Williamson, Juan Pino, and Emmanuel Dupoux. 2021. Voxpopuli: A large-scale multilingual speech corpus for representation learning, semi-supervised learning and interpretation. <i>arXiv preprint arXiv:2101.00390</i> .		
492	Chao-Han Huck Yang, Bo Li, Yu Zhang, Nanxin Chen, Rohit Prabhavalkar, Tara N Sainath, and Trevor Strohman. 2023. From english to more languages: Parameter-efficient model reprogramming for cross-lingual speech recognition. In <i>ICASSP</i> , pages 1–5. IEEE.		
498	Yu Yu, Chao-Han Huck Yang, Jari Kolehmainen, Prashanth G Shivakumar, Yile Gu, Sungho Ryu, Roger Ren, Qi Luo, Aditya Gourav, I-Fan Chen, Yi-Chieh Liu, et al. 2023. Low-rank adaptation of large language model rescoring for parameter-efficient speech recognition. In <i>ASRU-2023</i> , pages 1–8. IEEE.		
505	Yu Zhang, Wei Han, James Qin, Yongqiang Wang, Ankur Bapna, Zhehuai Chen, Nanxin Chen, Bo Li, Vera Axelrod, Gary Wang, et al. 2023. Google usm: Scaling automatic speech recognition beyond 100 languages. <i>arXiv preprint arXiv:2303.01037</i> .		