A Diffusion Model for High-quality **Adaptive** Text-to-Speech with **Untranscribed** Data



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In recent years, single-speaker TTS methods have been able to synthesize high-quality speech.

#### Issue

- The direct usage of Untranscribed Data remains a challenge.
- Requires sufficient amounts of Large-Scale
   Data for the Target Speaker.

### Adaptive TTS

Adaptive TTS models aim to generate high-quality speech for the target speaker given a **limited** amount of reference data.

by Zero-shot or Fine-tune

#### Issue

• When the amount of the reference speech is only around 10 seconds, the sample quality and speaker similarity are poor.

Recent diffusion models show impressive results in class-conditional and text-conditional image generation tasks via diffusion guidance methods.

• e.g., DALL·E 2, Midjourney, Stable Diffusion

#### Advantage

- High-Quality.
- Stable.
- Easy to Guided.

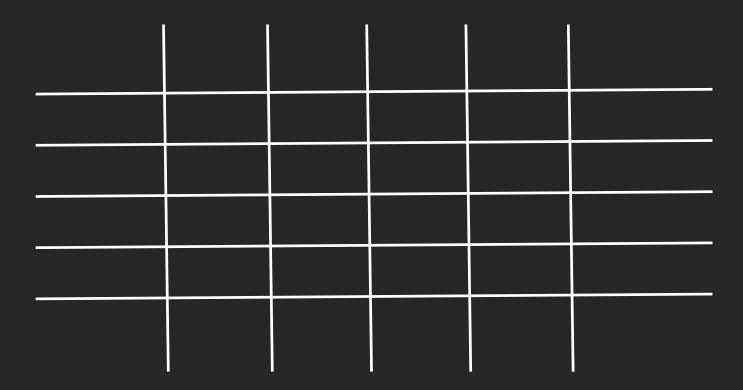


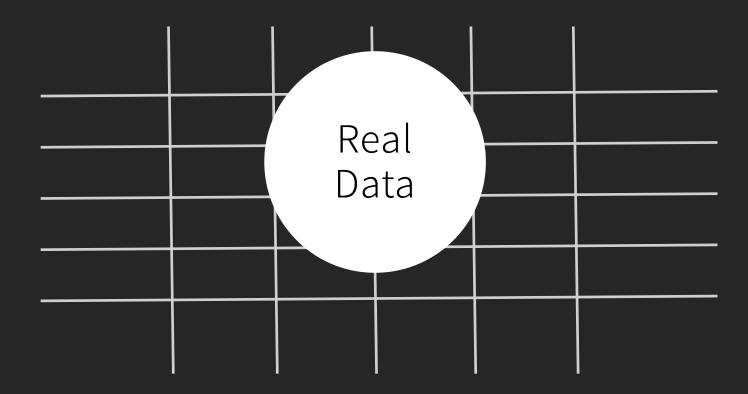
Taking advantage of the easy-to-guide nature of the Diffusion models, Guided TTS successfully constructed a single-speaker TTS that effectively utilizes untranscribed data.

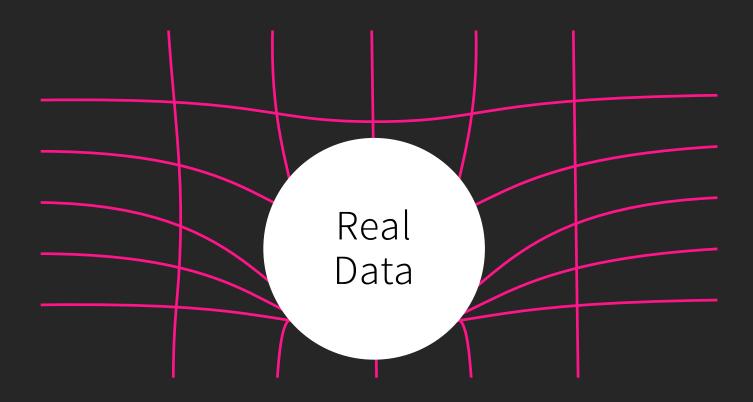
#### Guided TTS 2

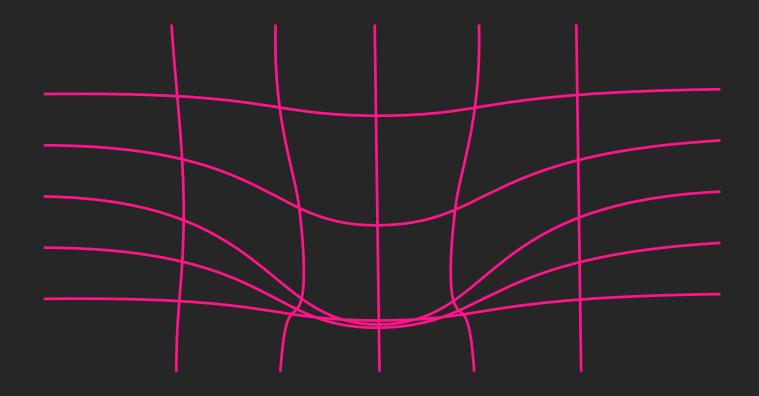
• Generate high-quality and high-similar speech with only 10 seconds of reference material.

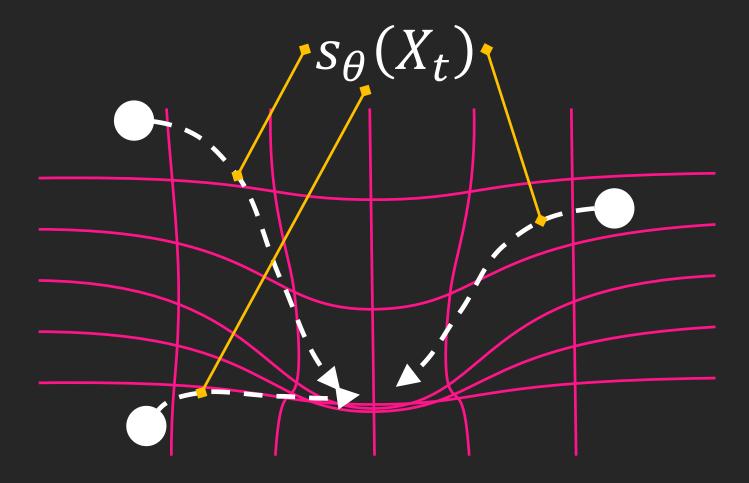




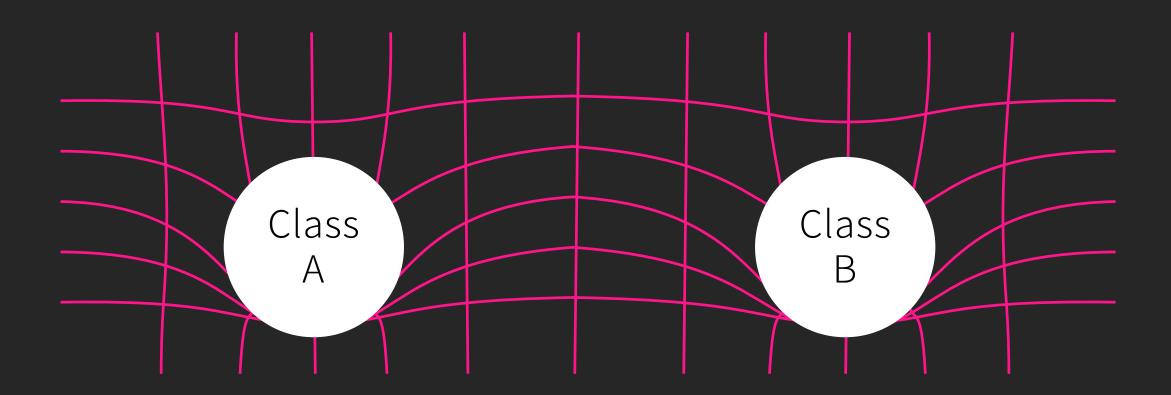




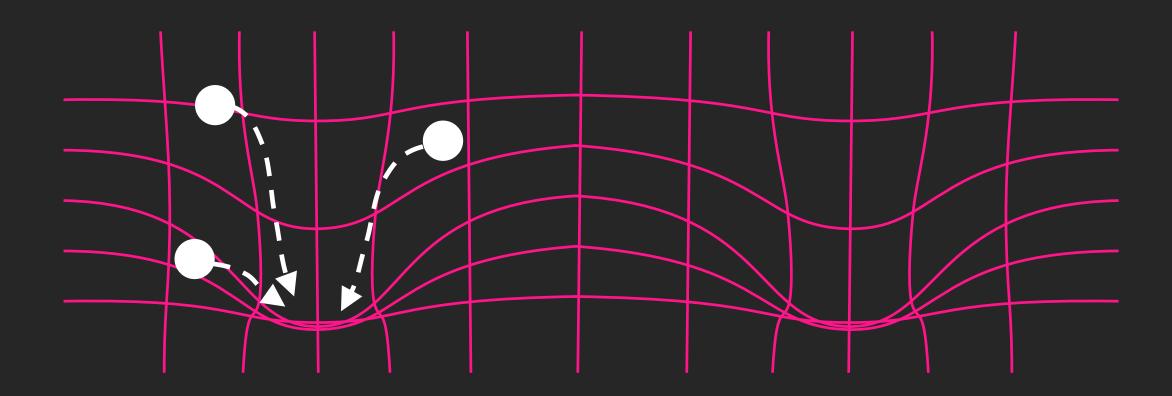




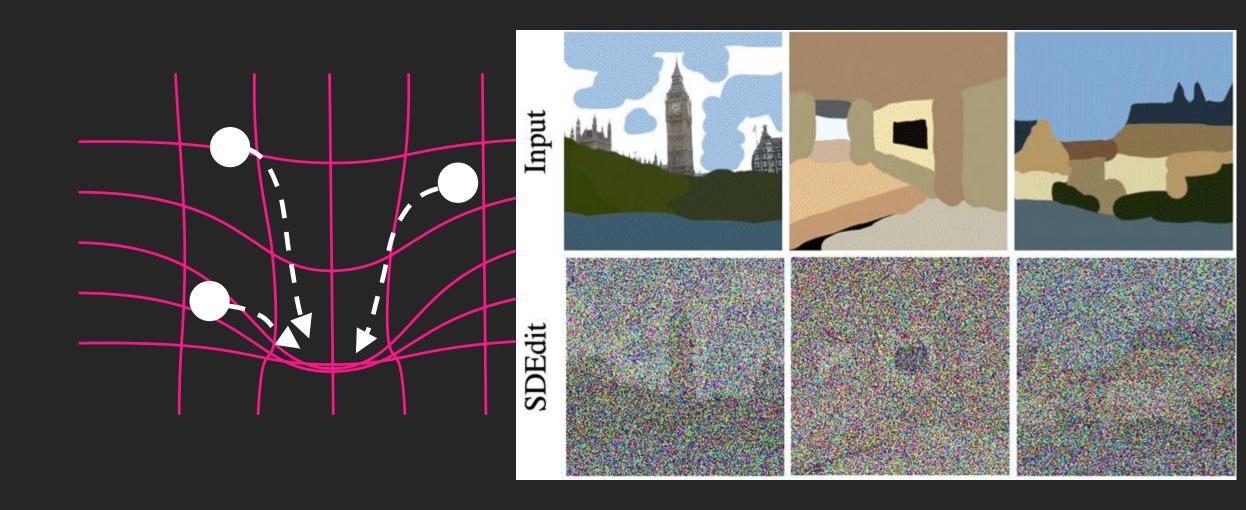
# ow to Guide?



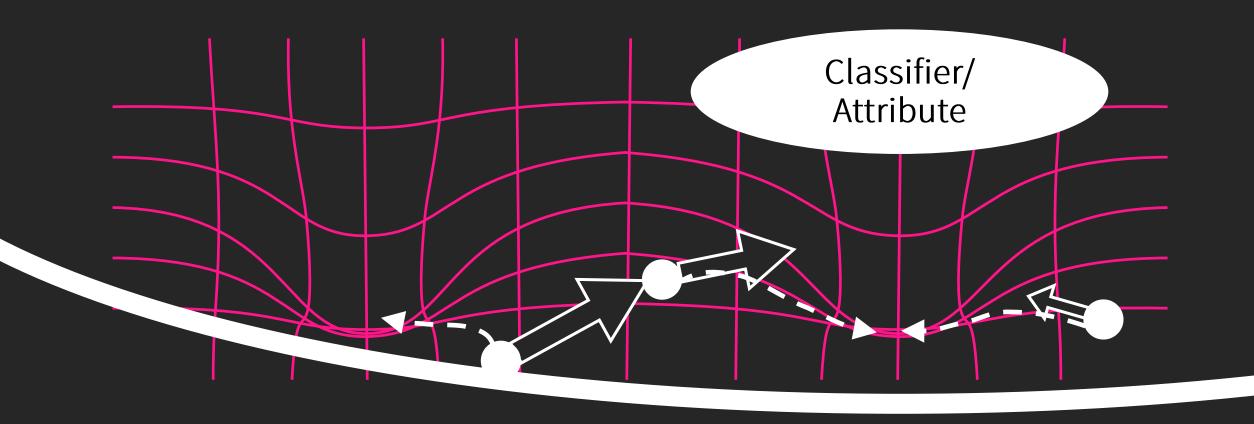
### jack the Initial State



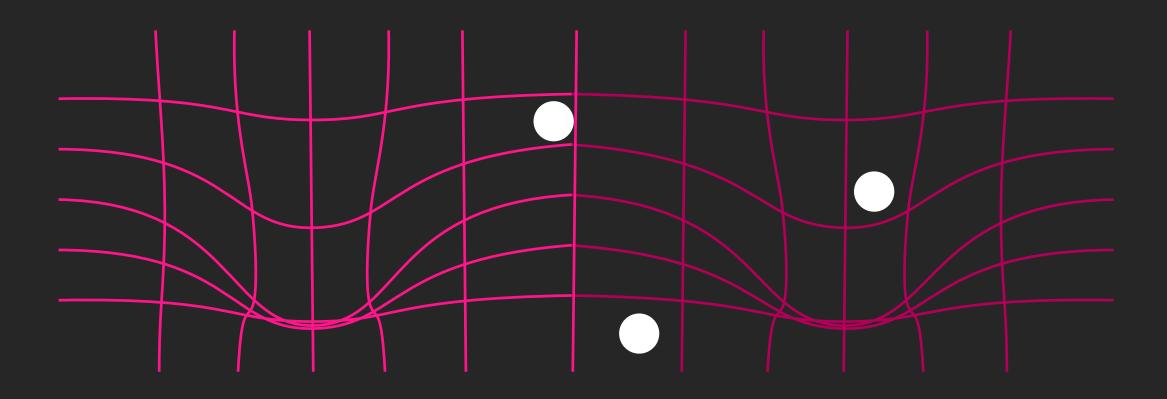
### jack the Initial State





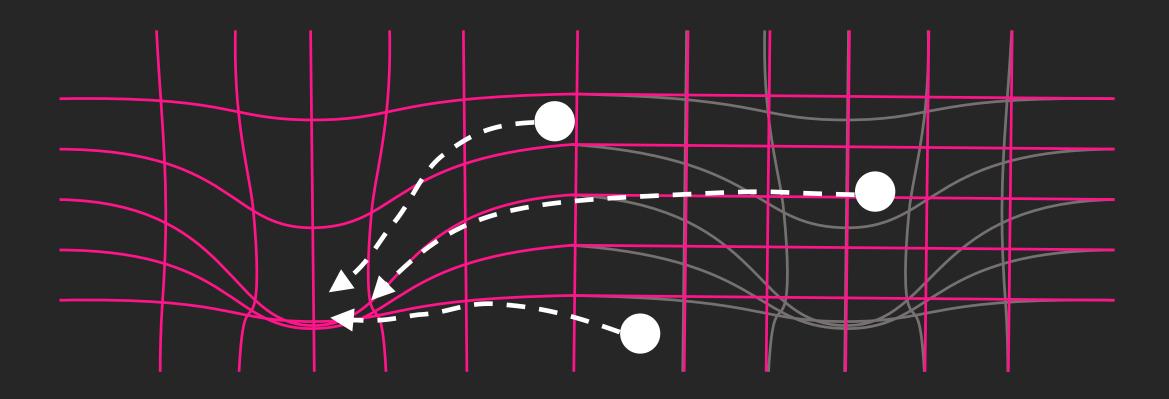


### Classifier-free Guidance



$$s_{\theta}(X_t)$$

#### Classifier-free Guidance



$$s_{\theta}(X_t|c=A)$$

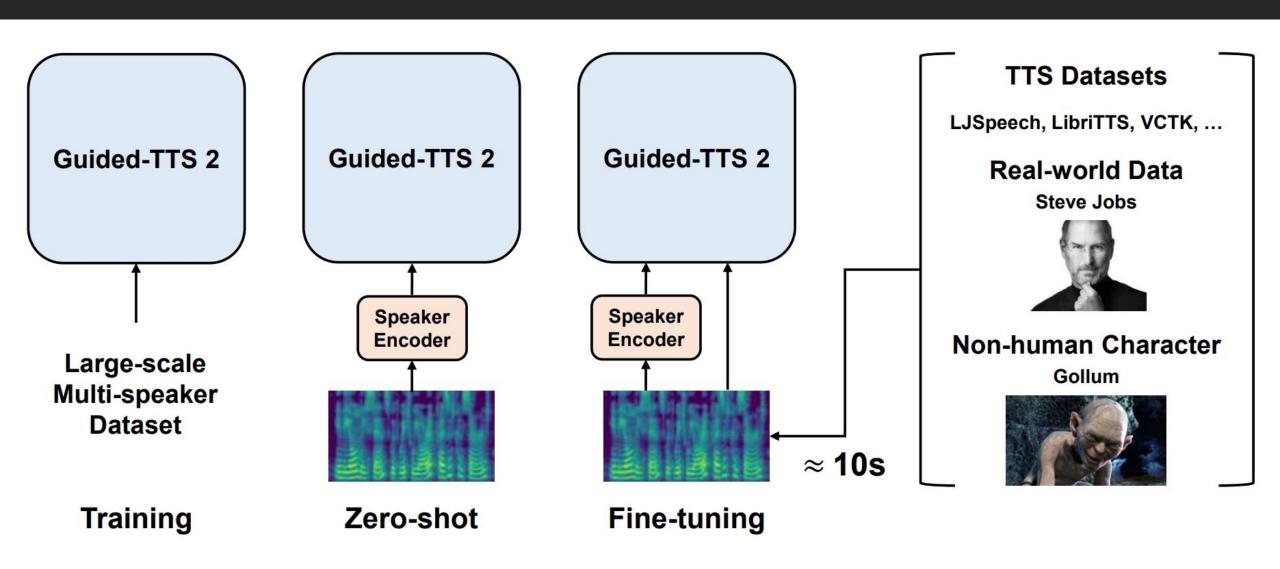
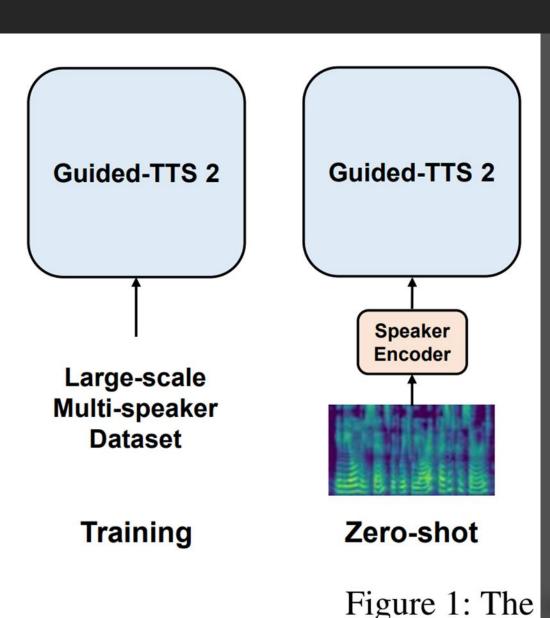


Figure 1: The overview of Guided-TTS 2.

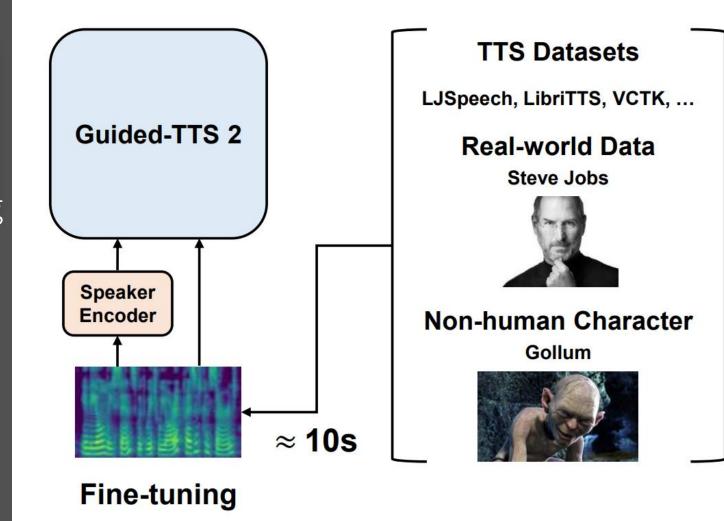


#### Zero-shot

- Speaker-conditional DDPM
- Guided by speaker embedding.
- Train a Speaker Encoder with GE2E loss.

#### Fine-tune

- Reference data: 10s
- Learning rate: 2e-5. lower than the pre-training learning rate 1e-4.
- Iterations: 500.
- Training Time: 40s on a NVIDIA RTX 8000 GPU.



overview of Guided-TTS 2.

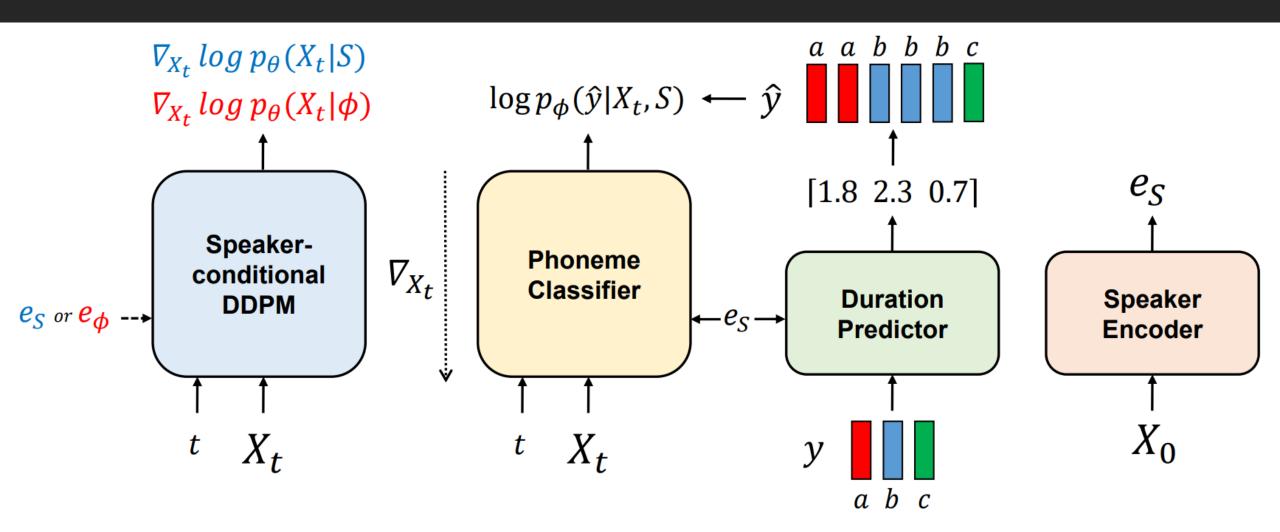


Figure 2: The overall components of Guided-TTS 2.

#### Guided TTS 2: Training

#### **Training Algorithm**

#### repeat until converged:

$$X_0 \sim \mathbf{X}$$

$$t \sim \mathcal{U}\left(\frac{1}{N}, 1\right)$$

$$\epsilon \sim \mathcal{N}(0, I)$$

$$X_t \leftarrow \sqrt{I - \lambda(t)} X_0 + \sqrt{\lambda(t)} \epsilon$$

#### Hyperparameter

• 
$$N = 50$$

• 
$$\lambda(t) = I - e^{-\int_0^t \beta_S \, ds}$$

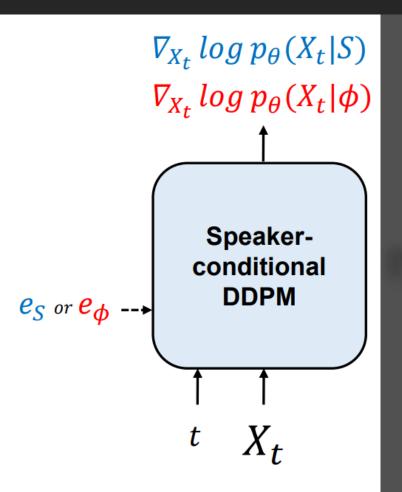
• 
$$\beta_t = \beta_0 + (\beta_T - \beta_0) \cdot t$$

• 
$$\beta_0 = 0.05, \beta_T = 20$$

Dropout rate of condition: 50%

$$\nabla_{\theta} \mathbb{E}_{t,X_{0},\epsilon_{t}} \left[ \left\| s_{\theta}(X_{t}|S = e_{S} \text{ or } e_{\emptyset}) + \frac{\epsilon}{\sqrt{\lambda(t)}} \right\|_{2}^{2} \right]$$

#### Speaker-conditional Guidance

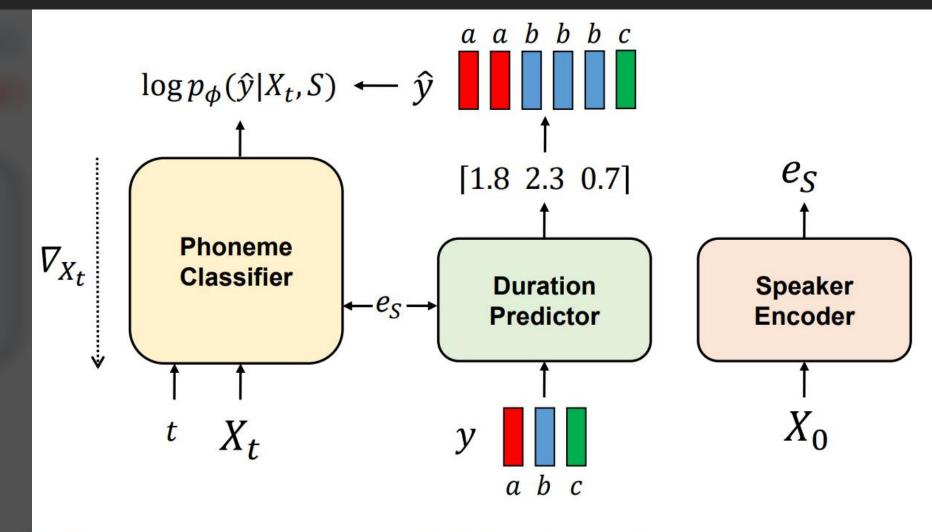


$$\hat{s}_{\theta}(X_{t}|\hat{S}) = s_{\theta}(X_{t}|\hat{S}) + \gamma_{S} \cdot \left(s_{\theta}(X_{t}|\hat{S}) - s_{\theta}(X_{t}|\emptyset)\right)$$

Proposed by Classifier-Free Diffusion Guidance

$$\hat{s}_{\theta}(X_t | \hat{y}, \hat{S}) = \hat{s}_{\theta}(X_t | \hat{S}) + \gamma_T \cdot \nabla_{X_t} \log p_{\varphi}(\hat{y} | X_t, \hat{S})$$

Framewise phoneme Classifier Guidance



2: The overall components of Guided-TTS 2.

#### Framewise Phoneme Classifier Guidance

This is about 70 times larger than the norm of the classifier gradient near  $X_0$ 

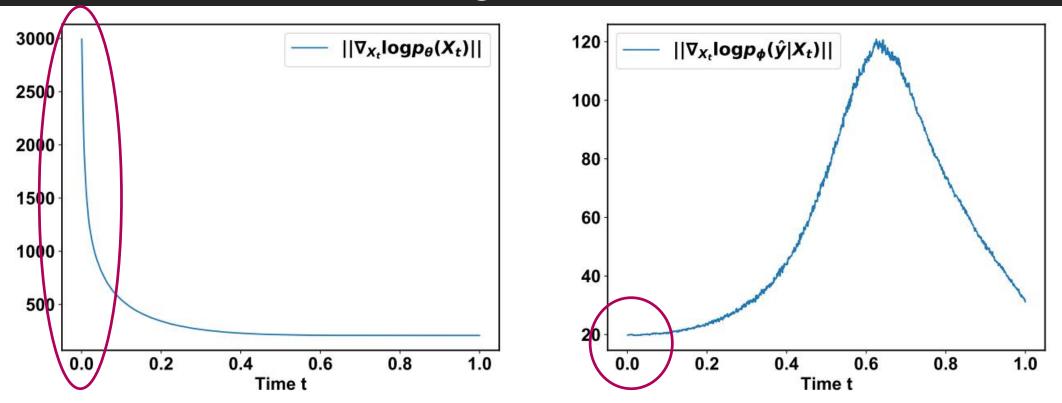
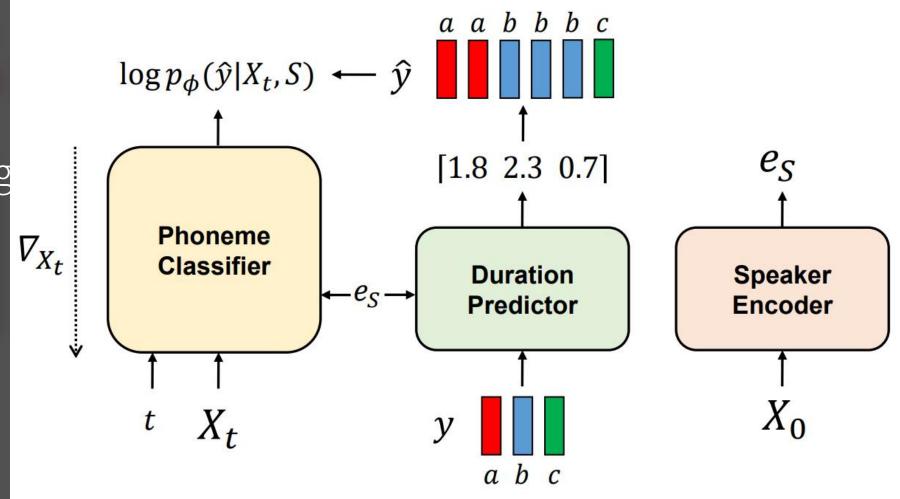


Figure 4: The norm of the unconditional score and the classifier gradient for each timestep t. (Left) The norm of the unconditional score (Right) The norm of the classifier gradient.

$$\hat{s}_{\theta}(X_{t}|\hat{y},\hat{S}) = \hat{s}_{\theta}(X_{t}|\hat{S}) + \gamma_{T} \cdot \frac{\|\hat{s}_{\theta}(X_{t}|\hat{S})\|}{\|\nabla_{X_{t}}\log p_{\varphi}(\hat{y}|X_{t},\hat{S})\|} \cdot \nabla_{X_{t}}\log p_{\varphi}(\hat{y}|X_{t},\hat{S})$$

#### Norm-based Guidance

The amount of scaling is proportional to the norm of the score.



2: The overall components of Guided-TTS 2.

#### Guided TTS 2: Sampling

#### **Sampling Algorithm**

- $\hat{y}$ : framewise phoneme label,  $\tau$ : temperature
- $\hat{S}$ : target speaker condition,  $\theta$ : parameter of DDPM

$$X_1 \sim \mathcal{N}(0, \tau^{-1}I)$$

for t in 
$$\{1, \dots, \frac{2}{N}, \frac{1}{N}\}$$
: Hyperparameter

$$z_t \sim \mathcal{N}(0, \tau^{-1}I)$$

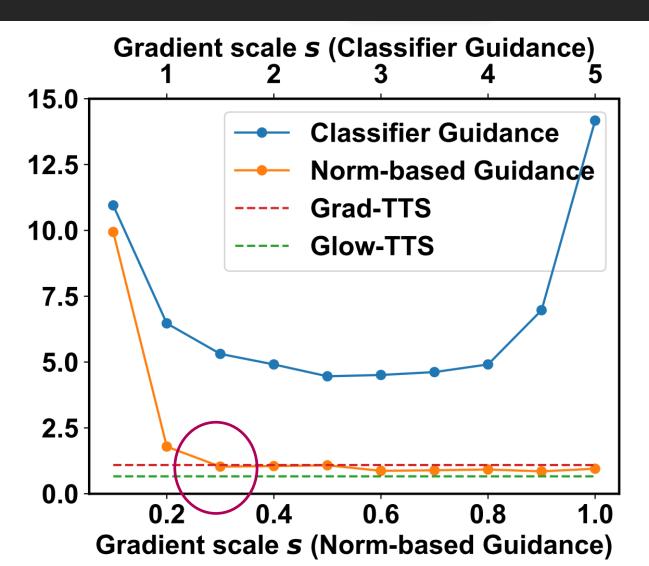
• 
$$\tau = 1.5$$

$$X_{t-\frac{1}{N}} \leftarrow X_t + \frac{\beta_t}{N} \left( \frac{1}{2} X_t + \hat{s}_{\theta} \left( X_t | \hat{y}, \hat{S} \right) \right) + \sqrt{\frac{\beta_t}{N}} z_t$$

return  $X_0$ 



#### CER of Norm-based Guidance (Guided-TTS 1)



- Classifier Guidance has more mispronunciations.
  - $\hat{s}_{\theta}(X_t|\hat{y},\hat{S}) = \hat{s}_{\theta}(X_t|\hat{S}) + \gamma_T \cdot \nabla_{X_t} \log p_{\varphi}(\hat{y}|X_t,\hat{S})$
- Norm-based Guidance has higher quality.
- $\gamma_T = 0.3$

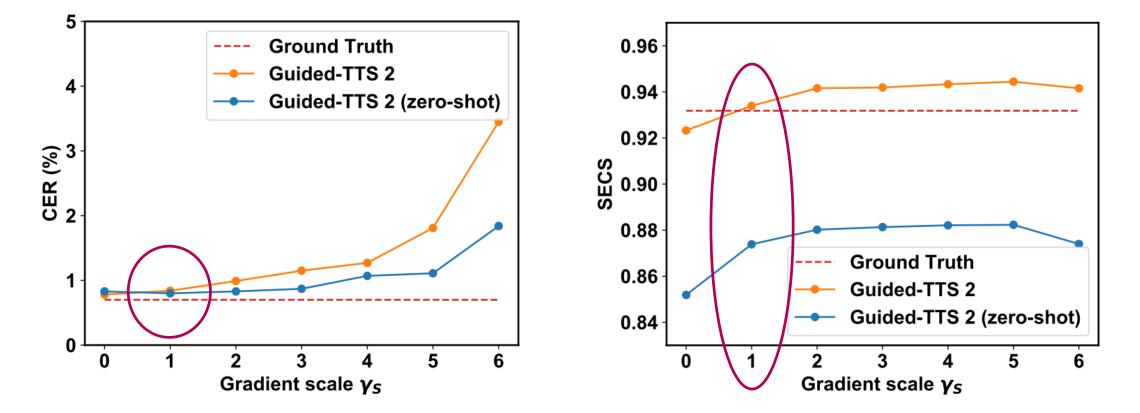


Figure 3: CER and SECS of Guided-TTS 2 according to speaker gradient scales. We refer to the fine-tuning setting of our model as Guided-TTS 2 and the zero-shot setting as Guided-TTS 2 (zero-shot).

$$\hat{s}_{\theta}(X_t|\hat{S}) = s_{\theta}(X_t|\hat{S}) + \gamma_{S} \cdot \left(s_{\theta}(X_t|\hat{S}) - s_{\theta}(X_t|\emptyset)\right)$$



• 
$$\gamma_S = 1$$

### Experiments

- Fine-tune has a higher Similarity.
- Zero-shot has a higher Quality.

Comparison with single-speaker TTS methods trained with LJSpeech.

Method	5-scale MOS	CER(%)	5-scale SMOS
Ground Truth	$4.45 \pm 0.05$	0.64	$3.85 \pm 0.08$
Mel + HiFi-GAN (Kong et al. (2020))	$4.24 \pm 0.08$	0.86	$3.80 \pm 0.08$
Grad-TTS (Popov et al. (2021))	$4.22 \pm 0.08$	0.98	$3.67 \pm 0.09$
Guided-TTS (Kim et al. (2021a))	$4.17 \pm 0.09$	1.23	$3.63 \pm 0.09$
Guided-TTS 2 (LT+LL fine-tune)	$4.21 \pm 0.09$	1.12	$3.69 \pm 0.09$
Guided-TTS 2 (LT+LL zero-shot)	$4.23 \pm 0.09$	0.89	$3.51 \pm 0.08$
Guided-TTS 2 (LT fine-tune)	$4.22 \pm 0.09$	1.16	$3.74 \pm 0.09$
Guided-TTS 2 (LT zero-shot)	$4.16 \pm 0.09$	1.03	$3.47 \pm 0.09$



Comparison with adaptive TTS methods.

Dataset	Method	5-scale MOS	CER(%)	5-scale SMOS
LibriTTS Dataset	Ground Truth	4.52 ± 0.05	0.70	$3.91 \pm 0.07$
	Mel + HiFi-GAN (Kong et al. (2020))	$4.28 \pm 0.08$	0.75	$3.86 \pm 0.08$
	Guided-TTS 2 (fine-tune)	$4.20 \pm 0.08$	0.84	$3.70 \pm 0.09$
	Guided-TTS 2 (zero-shot)	$4.25 \pm 0.09$	0.80	$3.51 \pm 0.10$
	YourTTS (Casanova et al. (2021))	$4.02 \pm 0.10$	2.38	$3.30 \pm 0.10$
	Meta-StyleSpeech (Min et al. (2021))	$3.98 \pm 0.11$	1.52	$3.42 \pm 0.09$
VCTK Dataset	Ground Truth	$4.45 \pm 0.05$	2.40	$3.71 \pm 0.07$
	Mel + HiFi-GAN (Kong et al. (2020))	$4.21 \pm 0.08$	2.81	$3.72 \pm 0.07$
	Guided-TTS 2 (fine-tune)	$4.11 \pm 0.09$	1.49	$3.57 \pm 0.10$
	Guided-TTS 2 (zero-shot)	$4.23 \pm 0.09$	0.81	$3.39 \pm 0.09$
	YourTTS (Casanova et al. (2021))	$3.94 \pm 0.10$	2.36	$3.19 \pm 0.09$
	Meta-StyleSpeech (Min et al. (2021))	$3.65 \pm 0.13$	1.84	$3.26 \pm 0.10$

### Guided TTS 2: Sampling

Method	Length(seconds)	<b>CER</b> (%)	SECS
	3	2.44	0.925
	5	1.67	0.930
Guided-TTS 2	10	1.12	0.929
	30	0.98	0.932
	60	1.14	0.931

Method	Iterations	Optimizer	CER(%)	SECS
Guided-TTS 2	0	-	0.80	0.873
	50	Initialize	0.82	0.908
	200	Initialize	0.88	0.929
	500	Initialize	0.84	0.937
	2000	Initialize	1.49	0.945
	500	Load	1.39	0.925

### Conclusion

This study proposes a novel Diffusion-based adaptive TTS method that generates speech quality comparable to the results of single-speaker TTS with only a few references.

- Adaptive TTS method.
- Hight-quality results.
- Only a few references are required.
- Effectively utilizes untranscribed data.

#### Demo: Steve Jobs

This audio was generated by a text-to-speech model for Steve Jobs. We use ten second untranscribed speech from Steve Jobs' Stanford Commencement Address.

- reference ⊲(€
- fine-tune
- zero-shot

#### Demo: Gollum

This audio was generated by a text-to-speech model for Gollum (魔戒 咕嚕), which can adapt to non-human characters using untranscribed data.

- reference
- fine-tune
- zero-shot

## Appendix: Trainset

Speaker Encoder	Speaker dependent phoneme classifier & duration predictor	Speaker conditional DDPM
<ul><li>VoxCeleb2</li><li>6,112 speaker</li><li>Over 1M utterance</li></ul>	<ul><li>LibriSpeech</li><li>2,484 speaker</li><li>Approximately 1,000 hours</li></ul>	LibriTTS • 2,456 speaker • 585 hours
		Libri-Light (unlab-600 + unlab-6k) • 2,231 speaker • Approximately 6,300 hours