

Parallel waveform synthesis

Eunwoo Song / Naver Clova

Who am I?

Education

- B.S., E.E., Yonsei Univ., Seoul, Korea (Aug 2010)
- Combined M.S. and Ph.D., EE., Yonsei Univ., Seoul, Korea (Feb 2019)

Work experience

- NAVER Corp., Seongnam, Korea
 - Senior Research Scientist (Mar 2017 - present)
 - DNN TTS Team Lead, Clova Voice
- Seoul National Univ., Seoul, Korea
 - Adjunct professor, Artificial Intelligence Institute (Aug 2022 - present)



Who am I?

Education

- B.S., E.E., Yonsei Univ., Seoul, Korea (Aug 2010)
- Combined M.S. and Ph.D., EE., Yonsei Univ., Seoul, Korea (Feb 2019)

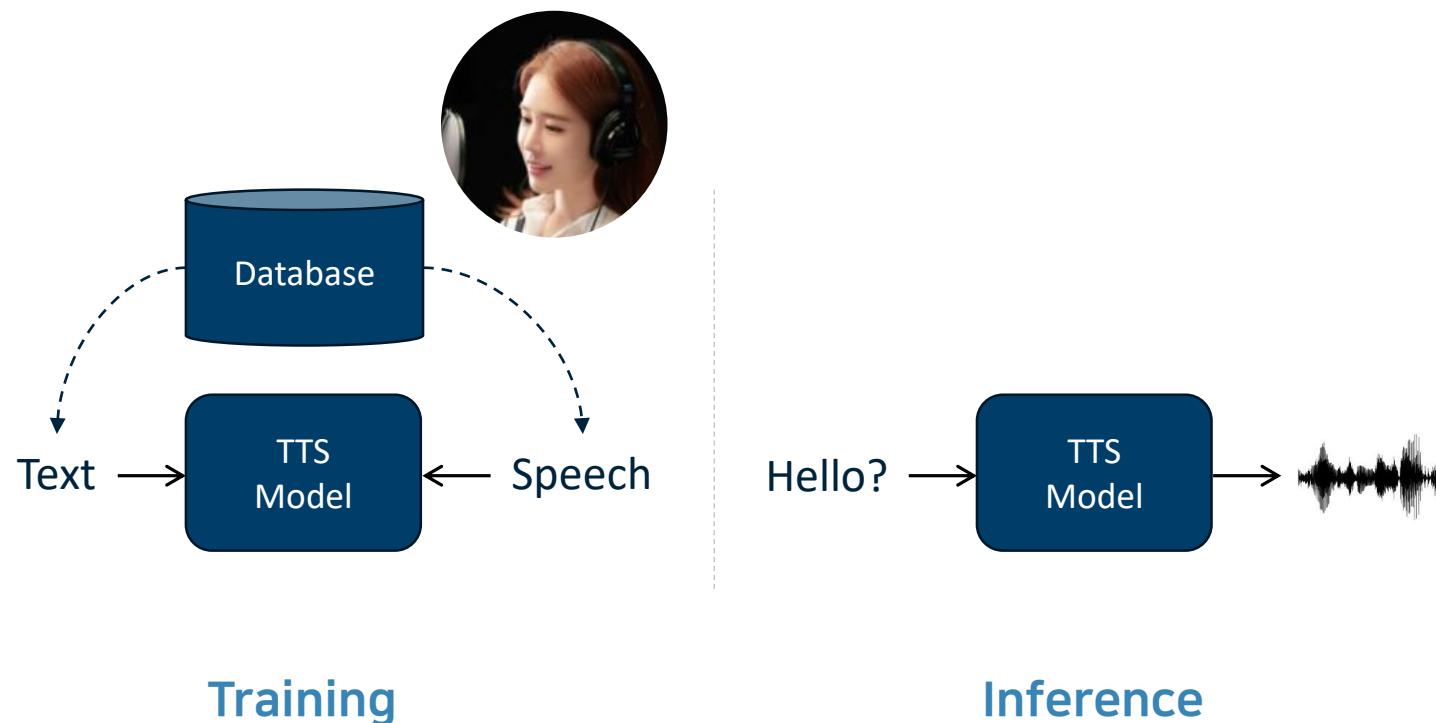
Work experience

- NAVER Corp., Seongnam, Korea
 - Senior Research Scientist (Mar 2017 - present)
 - DNN TTS Team Lead, Clova Voice
- Seoul National Univ., Seoul, Korea
 - Adjunct professor, Artificial Intelligence Institute (Aug 2022 - present)



Introduction

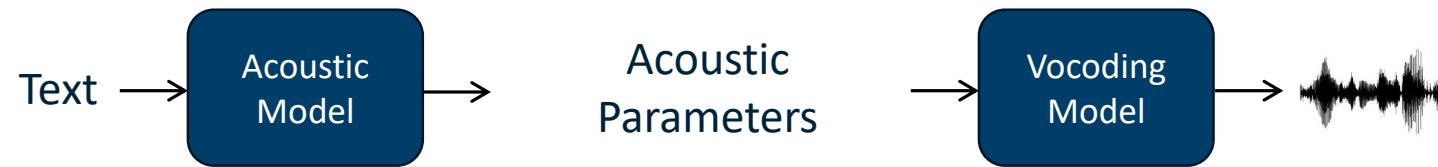
Deep learning-based TTS system



Human-like voice quality 😊

Introduction

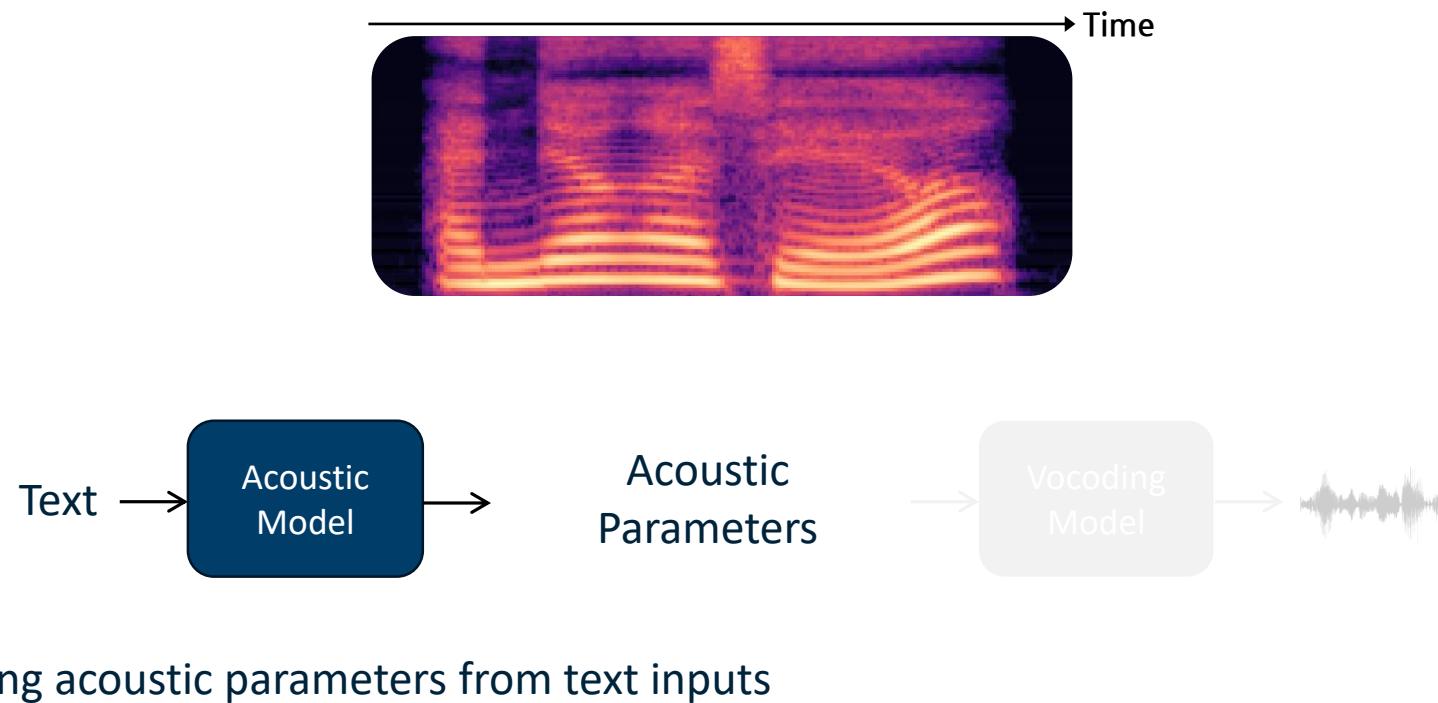
Deep learning-based TTS system



DNN TTS = Acoustic model + Vocoding model

Introduction

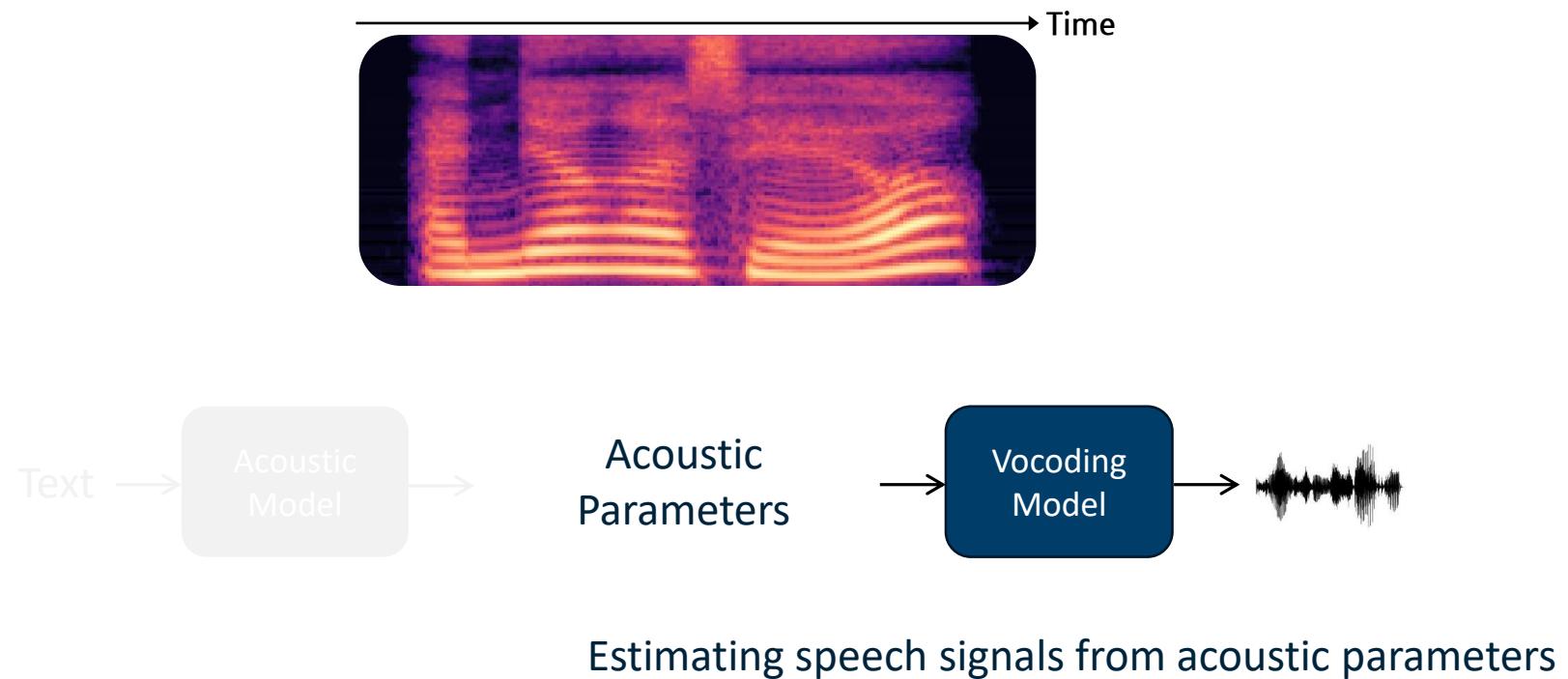
Deep learning-based TTS system



DNN TTS = Acoustic model + Vocoding model

Introduction

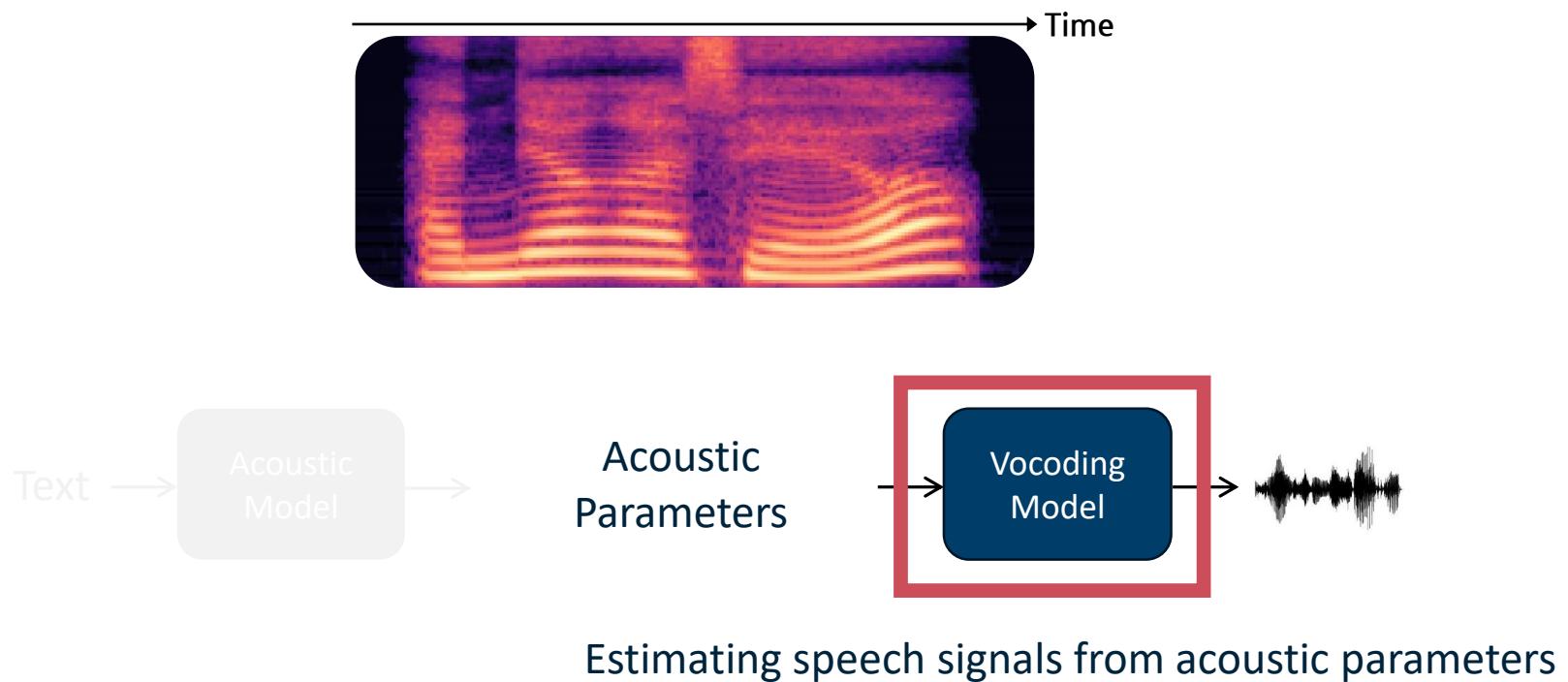
Deep learning-based TTS system



DNN TTS = Acoustic model + Vocoding model

Introduction

Deep learning-based TTS system



DNN TTS = Acoustic model + Vocoding model

Introduction

PARALLEL WAVEGAN: A FAST WAVEFORM GENERATION MODEL BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH MULTI-RESOLUTION SPECTROGRAM

Ryuichi Yamamoto¹, Eunwoo Song² and Jae-Min Kim²

¹LINE Corp., Tokyo, Japan.

²NAVER Corp., Seongnam, Korea

ABSTRACT

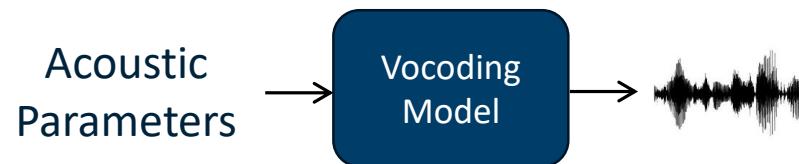
We propose Parallel WaveGAN, a distillation-free, fast, and small-footprint waveform generation method using a generative adversarial network. In the proposed method, a non-autoregressive WaveNet is trained by jointly optimizing multi-resolution spectrogram and adversarial loss functions, which can effectively capture the time-frequency distribution of the realistic speech waveform. As our method does not require density distillation used in the conventional teacher-student framework, the entire model can be easily trained. Furthermore, our model is able to generate high-fidelity speech even with its compact architecture. In particular, the proposed Parallel WaveGAN has only 1.44 M parameters and can generate 24 kHz speech waveform 28.68 times faster than real-time on a single GPU environment. Perceptual listening test results verify that our proposed method achieves 4.16 mean opinion score within a Transformer-based text-to-speech framework, which is comparative to the best distillation-based Parallel WaveNet system.

Parallel waveform synthesis

Vocoding models: Overview

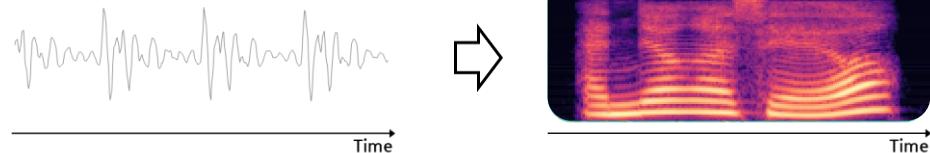
Vocoding models: Overview

Estimating speech signals from acoustic parameters



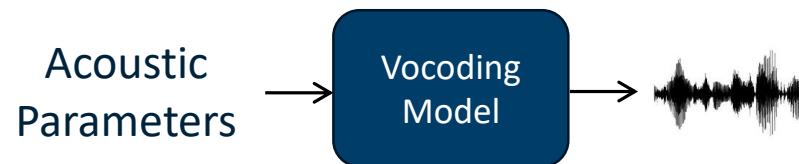
Acoustic parameters..?

Representing speech characteristics
such as F0, spectrum, v/uv ...



Vocoding models: Overview

Estimating speech signals from acoustic parameters



What is the main model?

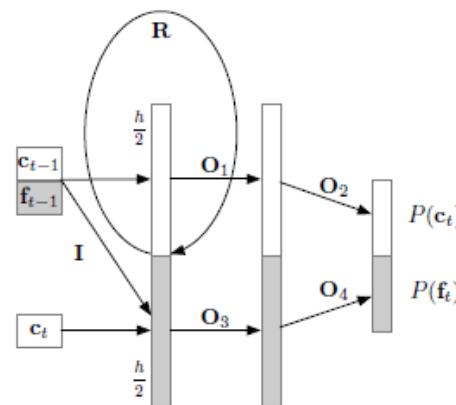
Vocoding models: Overview

Estimating speech signals from acoustic parameters



What is the main model?

WaveRNN based on the RNN model



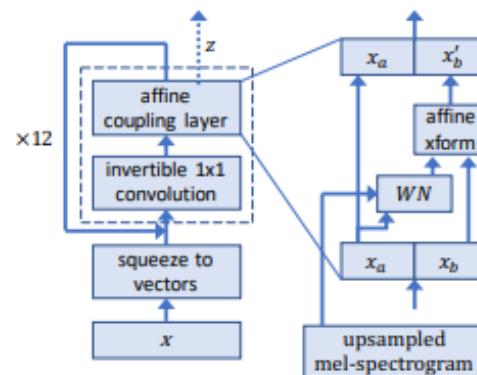
Vocoding models: Overview

Estimating speech signals from acoustic parameters



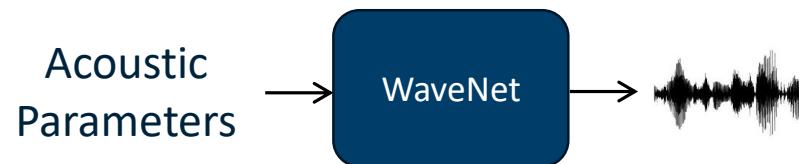
What is the main model?

WaveGlow based on the Flow model



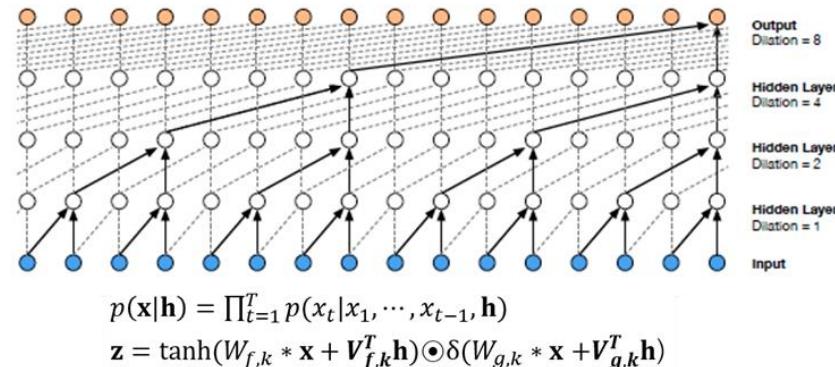
Vocoding models: Overview

Estimating speech signals from acoustic parameters



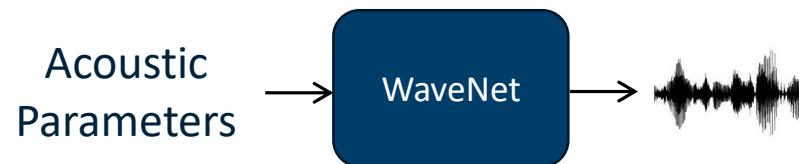
What is the main model?

WaveNet based on the CNN model



Vocoding models: Overview

Estimating speech signals from acoustic parameters



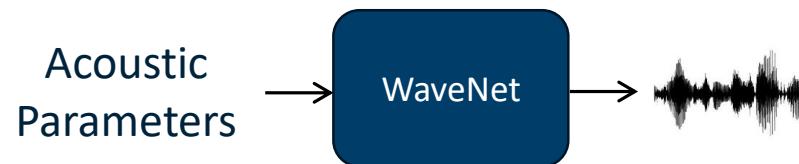
What is the main model?

WaveNet based on the CNN model

Estimating the current sample from the previous samples
We define this method as autoregressive vocoding model

Vocoding models: Overview

Estimating speech signals from acoustic parameters



What is the main model?

WaveNet based on the CNN model

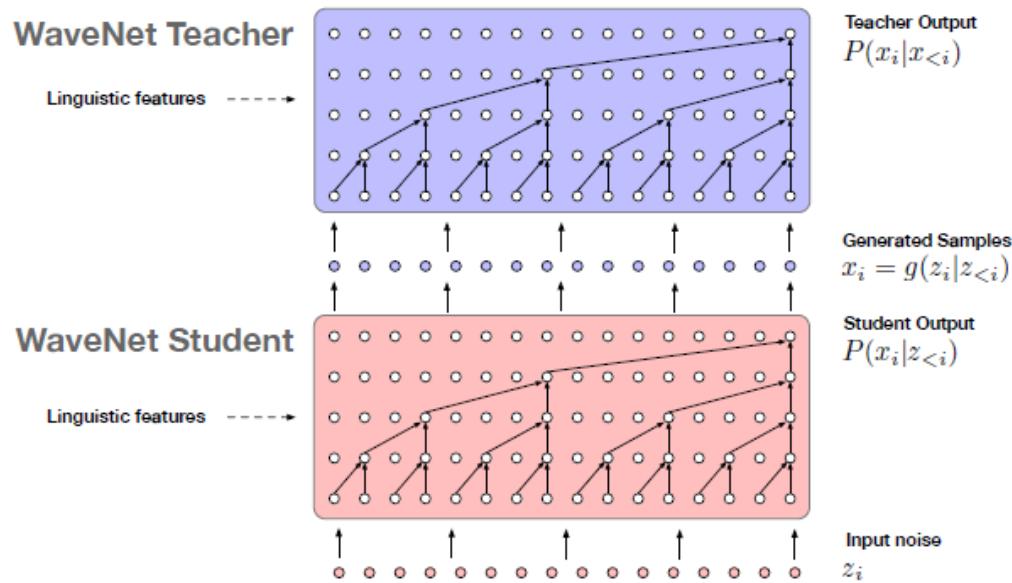
Estimating the current sample from the previous samples
We define this method as autoregressive vocoding model

WaveNet generates high-quality synthetic speech
However, it takes about 5 minutes to generate 1 sec audio



Vocoding models: Overview

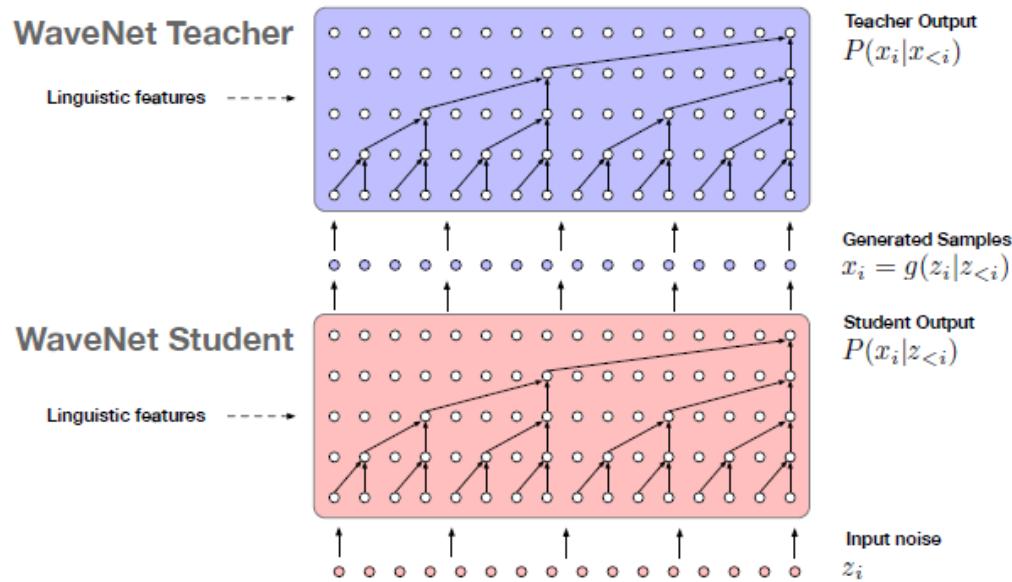
Estimating speech signals from acoustic parameters



One of the alternative method to address WaveNet's slow inference speed is the non-autoregressive **Parallel WaveNet**

Vocoding models: Overview

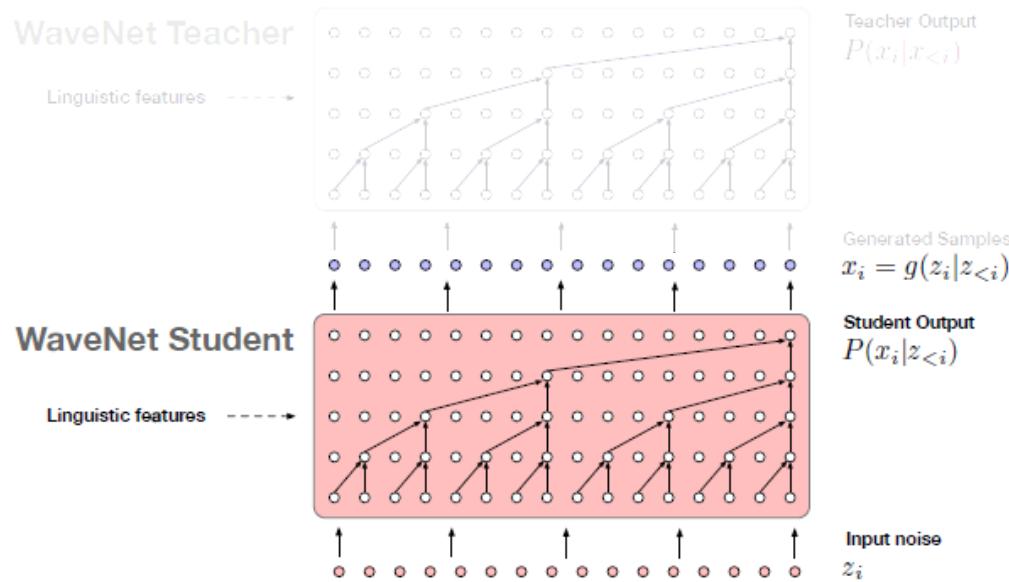
Estimating speech signals from acoustic parameters



Non-autoregressive Parallel WaveNet (=student) is trained to learn
the distribution of the autoregressive WaveNet (=teacher)

Vocoding models: Overview

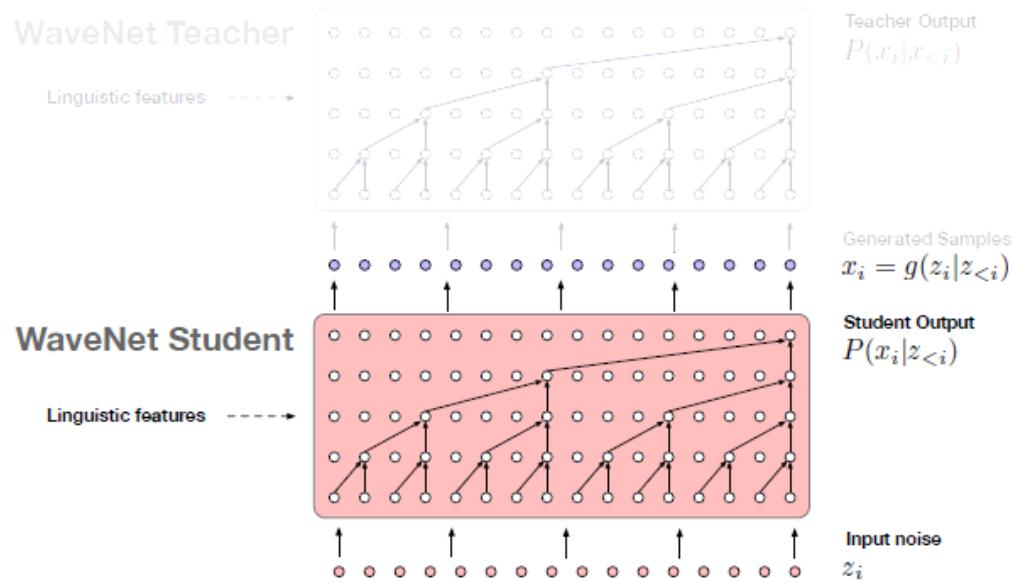
Estimating speech signals from acoustic parameters



Non-autoregressive Parallel WaveNet doesn't require the previous samples
Its inference speed is unlimited
(it takes about 0.02 sec to generate 1 sec audio)

Vocoding models: Overview

Estimating speech signals from acoustic parameters



There remain problems in the difficult training method...

Parallel waveform synthesis

Vocoding models: Parallel Wave**GAN**

Vocoding models: Parallel WaveGAN

PARALLEL WAVEGAN: A FAST WAVEFORM GENERATION MODEL BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH MULTI-RESOLUTION SPECTROGRAM

Ryuichi Yamamoto¹, Eunwoo Song² and Jae-Min Kim²

¹LINE Corp., Tokyo, Japan.

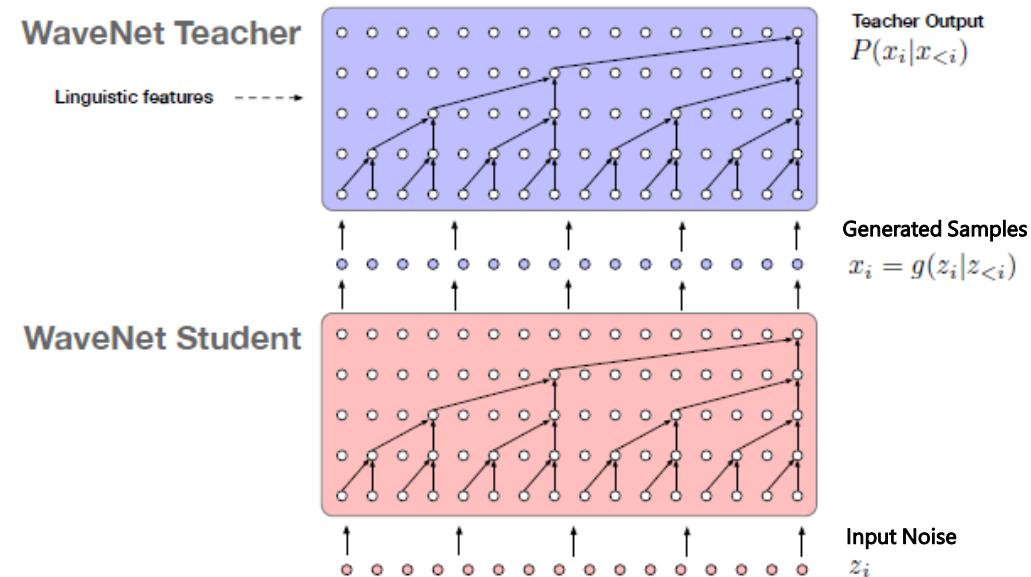
²NAVER Corp., Seongnam, Korea

ABSTRACT

We propose Parallel WaveGAN, a distillation-free, fast, and small-footprint waveform generation method using a generative adversarial network. In the proposed method, a non-autoregressive WaveNet is trained by jointly optimizing multi-resolution spectrogram and adversarial loss functions, which can effectively capture the time-frequency distribution of the realistic speech waveform. As our method does not require density distillation used in the conventional teacher-student framework, the entire model can be easily trained. Furthermore, our model is able to generate high-fidelity speech even with its compact architecture. In particular, the proposed Parallel WaveGAN has only 1.44 M parameters and can generate 24 kHz speech waveform 28.68 times faster than real-time on a single GPU environment. Perceptual listening test results verify that our proposed method achieves 4.16 mean opinion score within a Transformer-based text-to-speech framework, which is comparative to the best distillation-based Parallel WaveNet system.

Vocoding models: Parallel WaveGAN

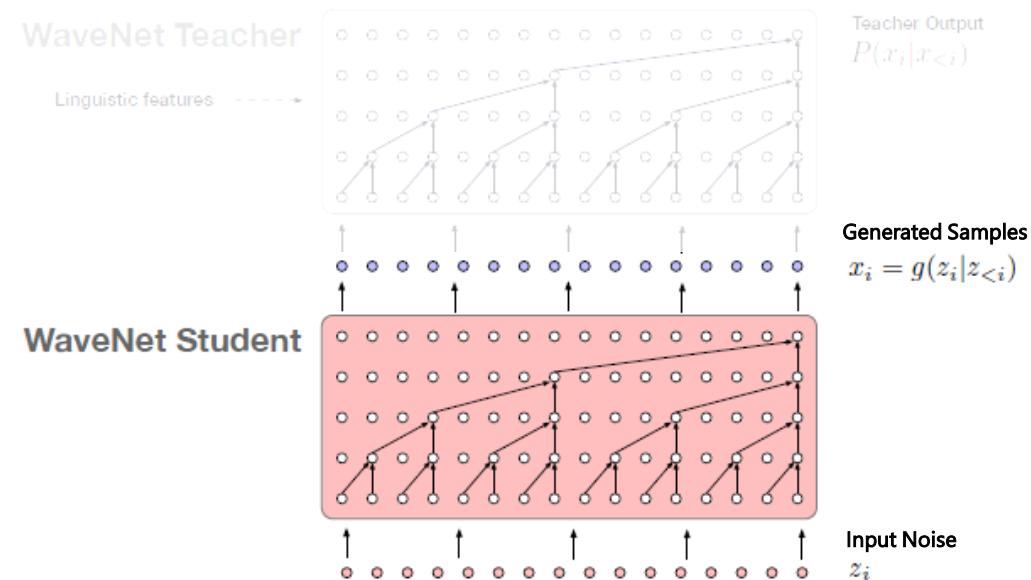
1. Removed the teacher-student distillation process



Vocoding models: Parallel WaveGAN

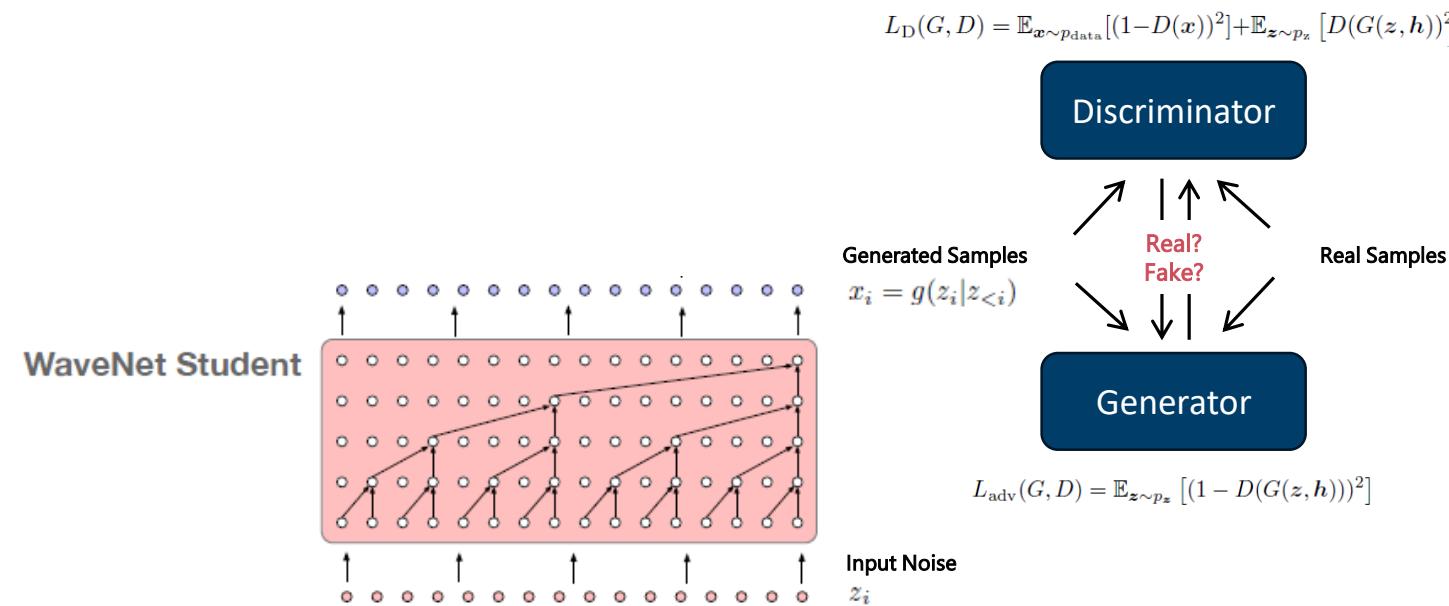
1. Removed the teacher-student distillation process

→ Entire model can be “easily” trained



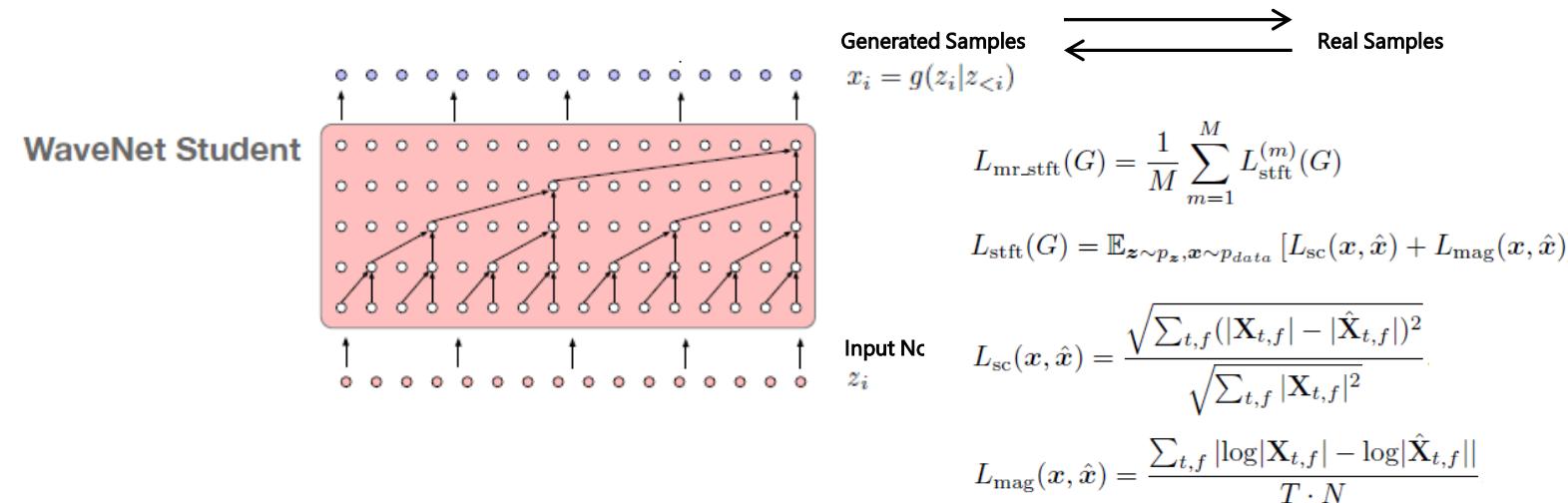
Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method



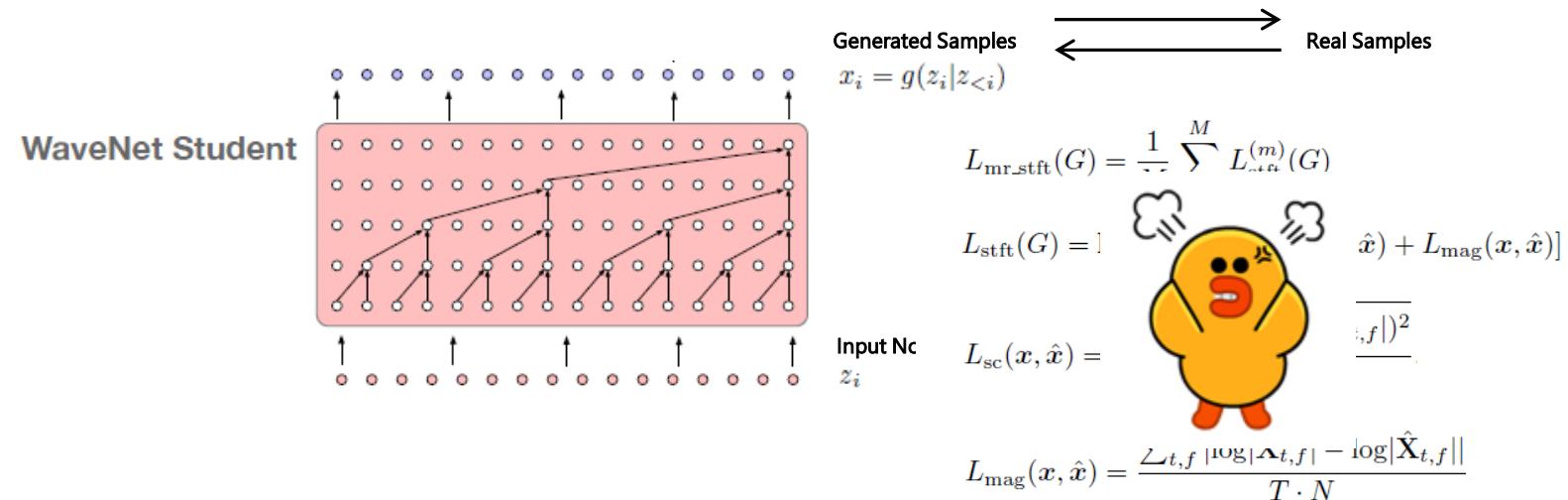
Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the multi-resolution STFT loss



Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the multi-resolution STFT loss



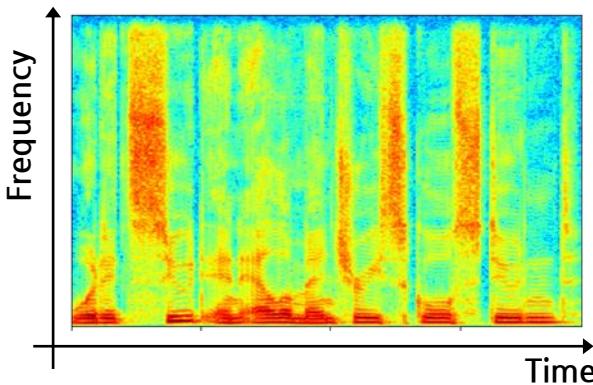
Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the multi-resolution **STFT** loss



STFT (short-time Fourier transform)?

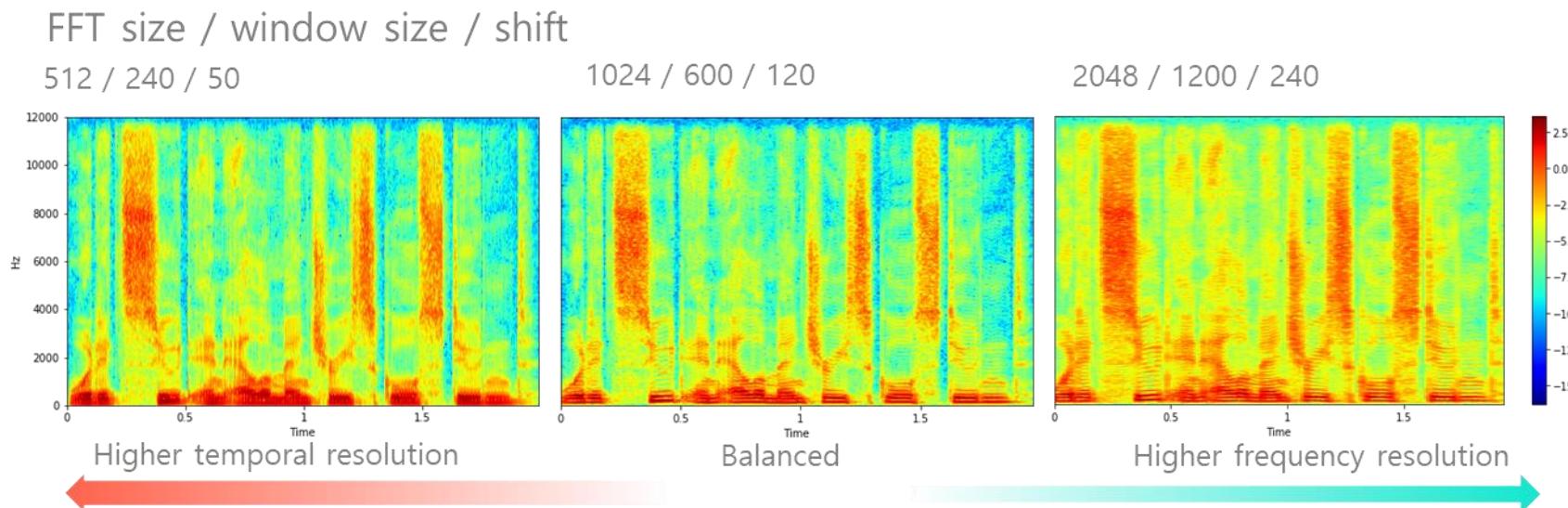
Time-frequency representation of speech signal



Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

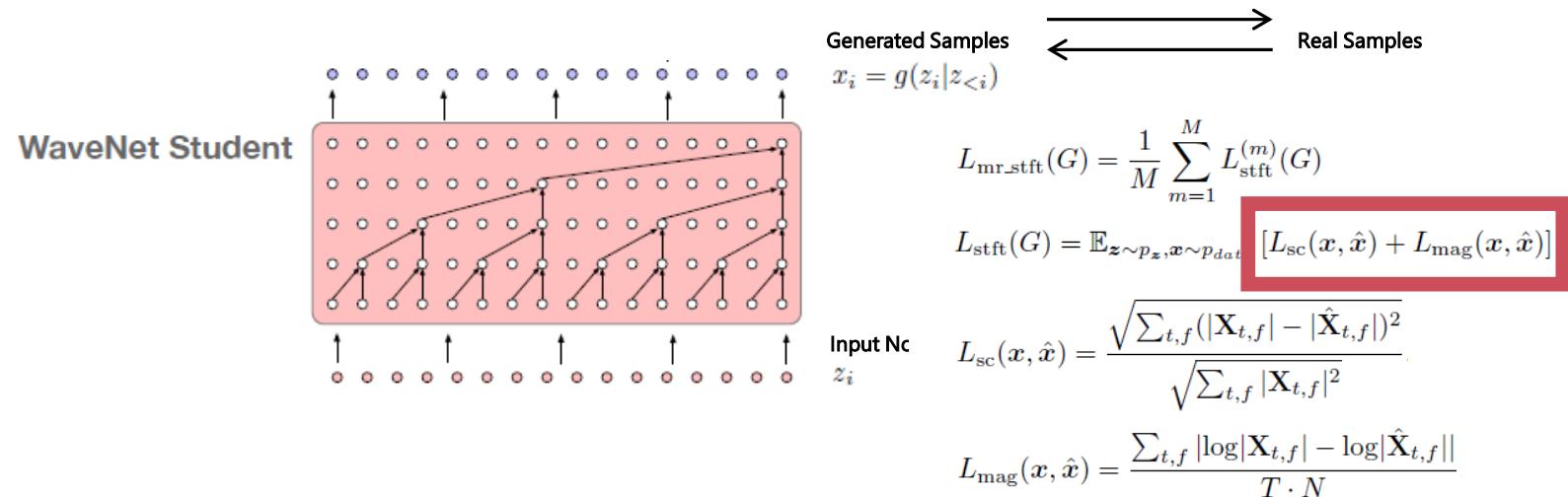


Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

There are **two** loss functions



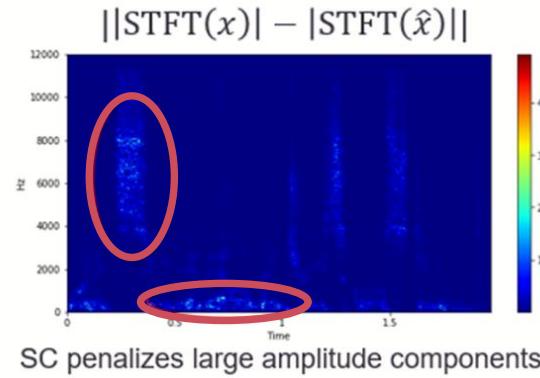
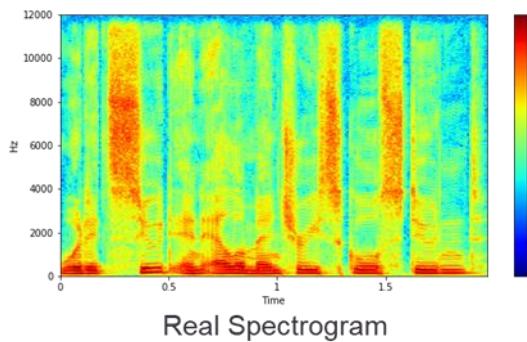
Vocoding models: Parallel WaveGAN

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

There are two loss functions

One penalizes **large energy components**



$$L_{\text{sc}}(x, \hat{x}) = \frac{\sqrt{\sum_{t,f} (|\mathbf{X}_{t,f}| - |\hat{\mathbf{X}}_{t,f}|)^2}}{\sqrt{\sum_{t,f} |\mathbf{X}_{t,f}|^2}}$$

Vocoding models: Parallel WaveGAN

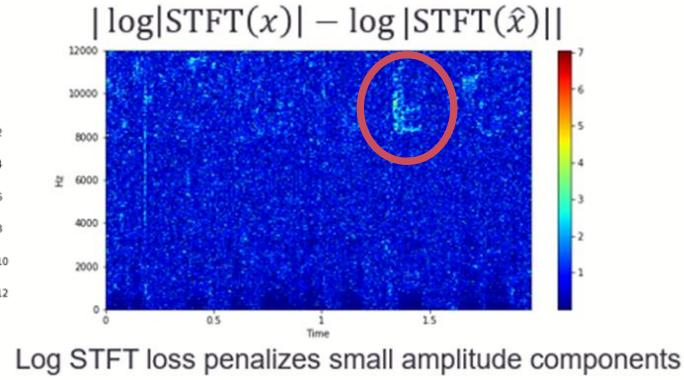
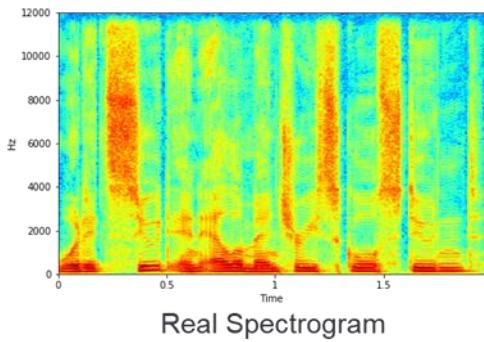
1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

There are two loss functions

One penalizes **large energy** components

The other penalizes **small energy** components



$$L_{\text{mag}}(x, \hat{x}) = \frac{\sum_{t,f} |\log|X_{t,f}| - \log|\hat{X}_{t,f}||}{T \cdot N}$$

Vocoding models: Parallel WaveGAN

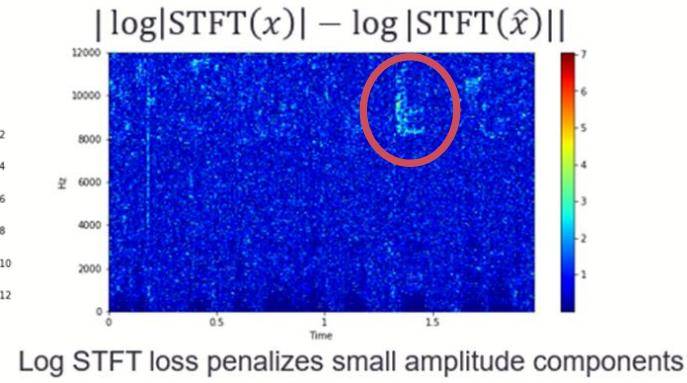
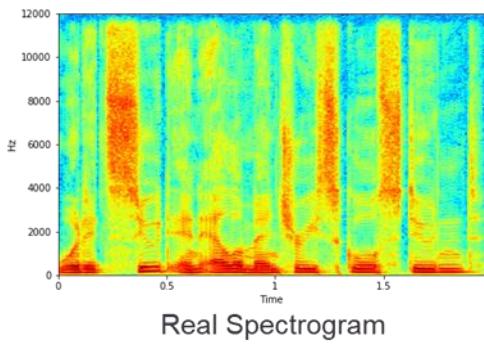
1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

There are two loss functions

One penalizes **large energy** components

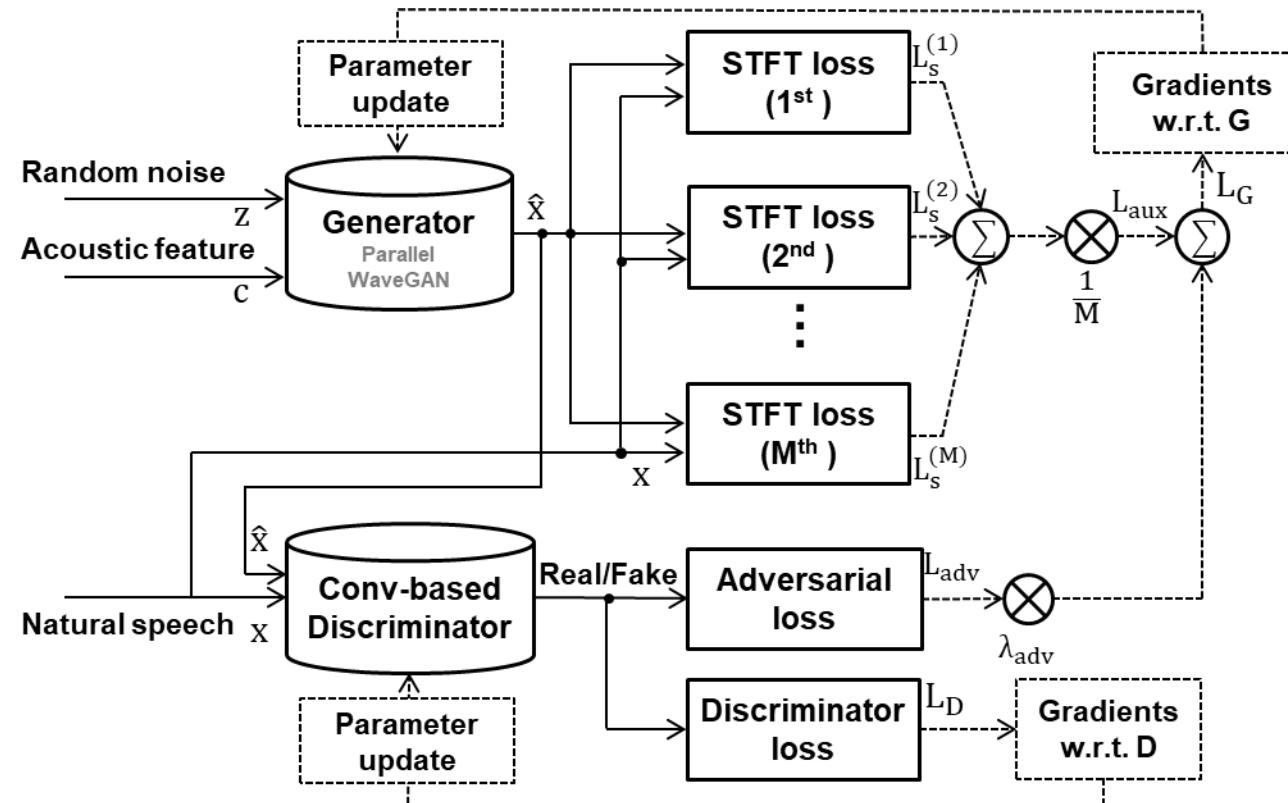
The other penalizes **small energy** components



$$L_{\text{mr_stft}}(G) = \frac{1}{M} \sum_{m=1}^M L_{\text{stft}}^{(m)}(G)$$
$$L_{\text{stft}}(G) = \mathbb{E}_{\mathbf{z} \sim p_{\mathbf{z}}, \mathbf{x} \sim p_{\text{data}}} [L_{\text{sc}}(\mathbf{x}, \hat{\mathbf{x}}) + L_{\text{mag}}(\mathbf{x}, \hat{\mathbf{x}})]$$
$$L_{\text{sc}}(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sqrt{\sum_{t,f} (|\mathbf{X}_{t,f}| - |\hat{\mathbf{X}}_{t,f}|)^2}}{\sqrt{\sum_{t,f} |\mathbf{X}_{t,f}|^2}}$$
$$L_{\text{mag}}(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sum_{t,f} |\log |\mathbf{X}_{t,f}| - \log |\hat{\mathbf{X}}_{t,f}||}{T \cdot N}$$

Vocoding models: Parallel WaveGAN

Training method



Vocoding models: Parallel WaveGAN

Training method

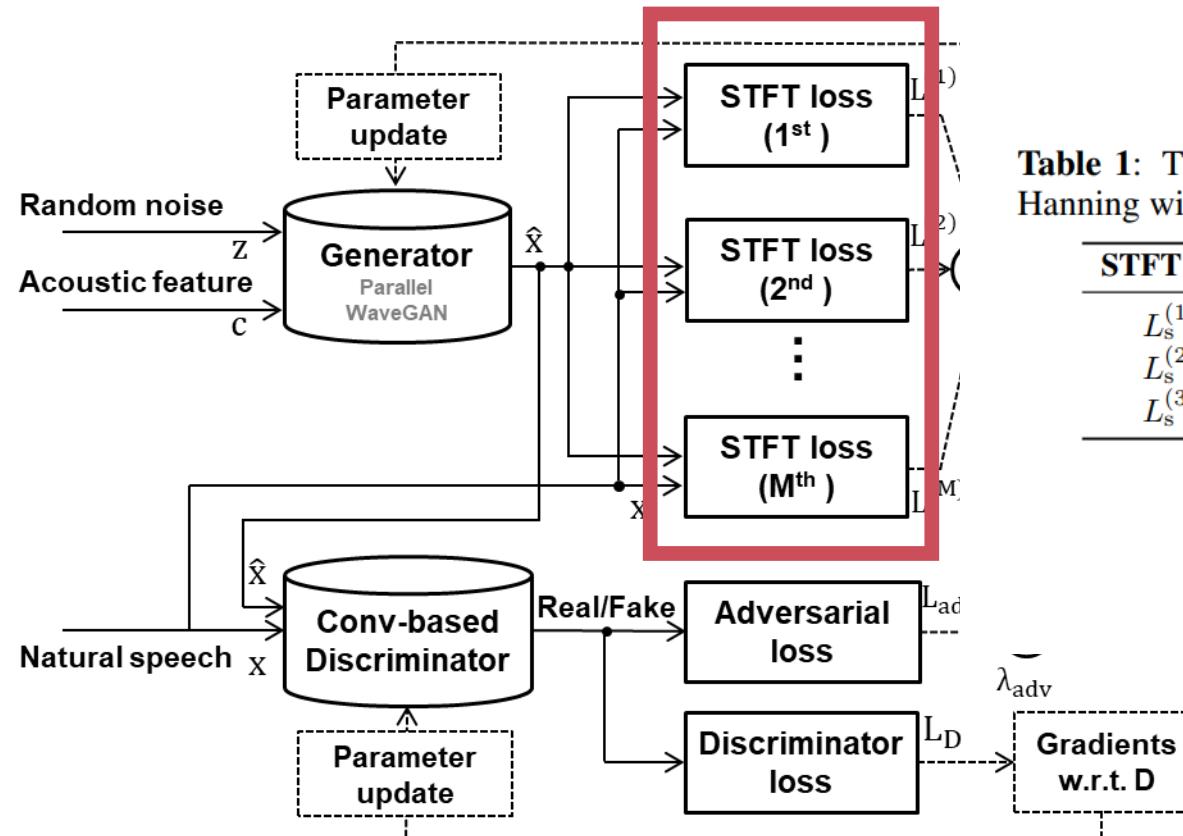


Table 1: The details of the multi-resolution STFT loss. A Hanning window was applied before the FFT process.

STFT loss	FFT size	Window size	Frame shift
$L_s^{(1)}$	1024	600 (25 ms)	120 (5 ms)
$L_s^{(2)}$	2048	1200 (50 ms)	240 (10 ms)
$L_s^{(3)}$	512	240 (10 ms)	50 (≈ 2 ms)

Vocoding models: Parallel WaveGAN

PARALLEL WAVEGAN: A FAST WAVEFORM GENERATION MODEL BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH MULTI-RESOLUTION SPECTROGRAM

Ryuichi Yamamoto¹, Eunwoo Song² and Jae-Min Kim²

¹LINE Corp., Tokyo, Japan.

²NAVER Corp., Seongnam, Korea

ABSTRACT

We propose Parallel WaveGAN, a distillation-free, fast, and small-footprint waveform generation method using a generative adversarial network. In the proposed method, a non-autoregressive WaveNet is trained by jointly optimizing multi-resolution spectrogram and adversarial loss functions, which can effectively capture the time-frequency distribution of the realistic speech waveform. As our method does not require density distillation used in the conventional teacher-student framework, the entire model can be easily trained. Furthermore, our model is able to generate high-fidelity speech even with its compact architecture. In particular, the proposed Parallel WaveGAN has only 1.44 M parameters and can generate 24 kHz speech waveform 28.68 times faster than real-time on a single GPU environment. Perceptual listening test results verify that our proposed method achieves 4.16 mean opinion score within a Transformer-based text-to-speech framework, which is comparative to the best distillation-based Parallel WaveNet system.

Vocoding models: Parallel WaveGAN

Evaluation results

Table 2: The inference speed and the MOS results with 95% confidence intervals: Acoustic features extracted from the recorded speech signal were used to compose the input auxiliary features. The evaluation was conducted on a server with a single NVIDIA Tesla V100 GPU. Note that the inference speed k means that the system was able to generate waveforms k times faster than real-time.

System index	Model	KLD-based distillation	STFT loss	Adversarial loss	Number of layers	Model size	Inference speed	MOS
System 1	WaveNet	-	-	-	24	3.81 M	0.32×10^{-2}	3.61 ± 0.12
System 2	ClariNet	Yes	$L_s^{(1)}$	-	60	2.78 M	14.62	3.88 ± 0.11
System 3	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	-	60	2.78 M	14.62	4.21 ± 0.09
System 4	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	60	2.78 M	14.62	4.21 ± 0.09
System 5	Parallel WaveGAN	-	$L_s^{(1)}$	Yes	30	1.44 M	28.68	1.36 ± 0.07
System 6	Parallel WaveGAN	-	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	30	1.44 M	28.68	4.06 ± 0.10
System 7	Recording	-	-	-	-	-	-	4.46 ± 0.08

Table 3: Training time comparison: All the experiments were conducted on a server with two NVIDIA Tesla V100 GPUs. Each vocoder model corresponds to System 1, 3, 4, and 6 described in Table 2, respectively. Note that the times for ClariNets include the training time for the teacher WaveNet.

Model	Training time (days)
WaveNet	7.4
ClariNet	12.7
ClariNet-GAN	13.5
Parallel WaveGAN (ours)	2.8

Table 4: MOS results with 95% confidence intervals: Acoustic features generated from the Transformer TTS model were used to compose the input auxiliary features.

Model	MOS
Transformer + WaveNet	3.33 ± 0.11
Transformer + ClariNet	4.00 ± 0.10
Transformer + ClariNet-GAN	4.14 ± 0.10
Transformer + Parallel WaveGAN (ours)	4.16 ± 0.09
Recording	4.46 ± 0.08

Vocoding models: Parallel WaveGAN

Evaluation results

Table 2: The inference speed and the MOS results with 95% confidence intervals: Acoustic features extracted from the recorded speech signal were used to compose the input auxiliary features. The evaluation was conducted on a server with a single NVIDIA Tesla V100 GPU. Note that the inference speed k means that the system was able to generate waveforms k times faster than real-time.

System index	Model	KLD-based distillation	STFT loss	Adversarial loss	Number of layers	Model size	Inference speed	MOS
System 1	WaveNet	-	-	-	24	3.81 M	0.32×10^{-2}	3.61 ± 0.12
System 2	ClariNet	Yes	$L_s^{(1)}$	-	60	2.78 M	14.62	3.88 ± 0.11
System 3	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	-	60	2.78 M	14.62	4.21 ± 0.09
System 4	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	60	2.78 M	14.62	4.21 ± 0.09
System 5	Parallel WaveGAN	-	$L_s^{(1)}$	Yes	30	1.44 M	28.68	3.36 ± 0.07
System 6	Parallel WaveGAN	-	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	30	1.44 M	28.68	4.06 ± 0.10
System 7	Recording	-	-	-	-			4.46 ± 0.08

Table 3: Training time comparison: All the experiments were conducted on a server with two NVIDIA Tesla V100 GPUs. Each vocoder model corresponds to System 1, 3, 4, and 6 described in Table 2, respectively. Note that the times for ClariNets include the training time for the teacher WaveNet.

Model	Training time (days)
WaveNet	7.4
ClariNet	12.7
ClariNet-GAN	13.5
Parallel WaveGAN (ours)	2.8

Table 4: MOS results with 95% confidence intervals: Acoustic features generated from the Transformer TTS model were used to compose the input auxiliary features.

Model	MOS
Transformer + WaveNet	3.33 ± 0.11
Transformer + ClariNet	4.00 ± 0.10
Transformer + ClariNet-GAN	4.14 ± 0.10
Transformer + Parallel WaveGAN (ours)	4.16 ± 0.09
Recording	4.46 ± 0.08

Vocoding models: Parallel WaveGAN

Evaluation results

Table 2: The inference speed and the MOS results with 95% confidence intervals: Acoustic features extracted from the recorded speech signal were used to compose the input auxiliary features. The evaluation was conducted on a server with a single NVIDIA Tesla V100 GPU. Note that the inference speed k means that the system was able to generate waveforms k times faster than real-time.

System index	Model	KLD-based distillation	STFT loss	Adversarial loss	Number of layers	Model size	Inference speed	MOS
System 1	WaveNet	-	-	-	24	3.81 M	0.32×10^{-2}	3.61 ± 0.12
System 2	ClariNet	Yes	$L_s^{(1)}$	-	60	2.78 M	14.62	3.88 ± 0.11
System 3	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	-	60	2.78 M	14.62	4.21 ± 0.09
System 4	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	60	2.78 M	14.62	4.21 ± 0.09
System 5	Parallel WaveGAN	-	$L_s^{(1)}$	Yes	30	1.44 M	28.68	1.36 ± 0.07
System 6	Parallel WaveGAN	-	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	30	1.44 M	28.68	4.06 ± 0.10
System 7	Recording	-	-	-	-	-	-	4.46 ± 0.08

Table 3: Training time comparison: All the experiments were conducted on a server with two NVIDIA Tesla V100 GPUs. Each vocoder model corresponds to System 1, 3, 4, and 6 described in Table 2, respectively. Note that the times for ClariNets include the training time for the teacher WaveNet.

Model	Training time (days)
WaveNet	7.4
ClariNet	12.7
ClariNet-GAN	13.5
Parallel WaveGAN (ours)	2.8

Table 4: MOS results with 95% confidence intervals: Acoustic features generated from the Transformer TTS model were used to compose the input auxiliary features.

Model	MOS
Transformer + WaveNet	3.33 ± 0.11
Transformer + ClariNet	4.00 ± 0.10
Transformer + ClariNet-GAN	4.14 ± 0.10
Transformer + Parallel WaveGAN (ours)	4.16 ± 0.09
Recording	4.46 ± 0.08

Vocoding models: Parallel WaveGAN

Evaluation results

Table 2: The inference speed and the MOS results with 95% confidence intervals: Acoustic features extracted from the recorded speech signal were used to compose the input auxiliary features. The evaluation was conducted on a server with a single NVIDIA Tesla V100 GPU. Note that the inference speed k means that the system was able to generate waveforms k times faster than real-time.

System index	Model	KLD-based distillation	STFT loss	Adversarial loss	Number of layers	Model size	Inference speed	MOS
System 1	WaveNet	-	-	-	24	3.81 M	0.32×10^{-2}	3.61 ± 0.12
System 2	ClariNet	Yes	$L_s^{(1)}$	-	60	2.78 M	14.62	3.88 ± 0.11
System 3	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	-	60	2.78 M	14.62	4.21 ± 0.09
System 4	ClariNet	Yes	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	60	2.78 M	14.62	4.21 ± 0.09
System 5	Parallel WaveGAN	-	$L_s^{(1)}$	Yes	30	1.44 M	28.68	1.36 ± 0.07
System 6	Parallel WaveGAN	-	$L_s^{(1)} + L_s^{(2)} + L_s^{(3)}$	Yes	30	1.44 M	28.68	4.06 ± 0.10
System 7	Recording	-	-	-	-	-	-	4.46 ± 0.08

Table 3: Training time comparison: All the experiments were conducted on a server with two NVIDIA Tesla V100 GPUs. Each vocoder model corresponds to System 1, 3, 4, and 6 described in Table 2, respectively. Note that the times for ClariNets include the training time for the teacher WaveNet.

Model	Training time (days)
WaveNet	7.4
ClariNet	12.7
ClariNet-GAN	13.5
Parallel WaveGAN (ours)	2.8

Table 4: MOS results with 95% confidence intervals: Acoustic features generated from the Transformer TTS model were used to compose the input auxiliary features.

Model	MOS
Transformer + WaveNet	3.33 ± 0.11
Transformer + ClariNet	4.00 ± 0.10
Transformer + ClariNet-GAN	4.14 ± 0.10
Transformer + Parallel WaveGAN (ours)	4.16 ± 0.09
Recording	4.46 ± 0.08

Vocoding models: Parallel WaveGAN



Demo samples



Open source
(implemented by Tomoki Hayashi, Nagoya Univ.)

Parallel waveform synthesis

Parallel WaveGAN: Toward **high-quality** synthesis

Toward high-quality synthesis

IMPROVED PARALLEL WAVEGAN VOCODER WITH PERCEPTUALLY WEIGHTED SPECTROGRAM LOSS

Eunwoo Song¹, Ryuichi Yamamoto², Min-Jae Hwang³, Jin-Seob Kim¹, Ohsung Kwon¹, Jae-Min Kim¹

¹NAVER Corp., Seongnam, Korea

²LINE Corp., Tokyo, Japan

³Search Solutions Inc., Seongnam, Korea

ABSTRACT

This paper proposes a spectral-domain perceptual weighting technique for Parallel WaveGAN-based text-to-speech (TTS) systems. The recently proposed Parallel WaveGAN vocoder successfully generates waveform sequences using a fast non-autoregressive WaveNet model. By employing multi-resolution short-time Fourier transform (MR-STFT) criteria with a generative adversarial network, the light-weight convolutional networks can be effectively trained without any distillation process. To further improve the vocoding performance, we propose the application of frequency-dependent weighting to the MR-STFT loss function. The proposed method penalizes perceptually-sensitive errors in the frequency domain; thus, the model is optimized toward reducing auditory noise in the synthesized speech. Subjective listening test results demonstrate that our proposed method achieves 4.21 and 4.26 TTS mean opinion scores for female and male Korean speakers, respectively.

“Weighted spectral Loss”

Toward high-quality synthesis

PARALLEL WAVEFORM SYNTHESIS BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH VOICING-AWARE CONDITIONAL DISCRIMINATORS

Ryuichi Yamamoto¹, Eunwoo Song², Min-Jae Hwang³ and Jae-Min Kim²

¹LINE Corp., Tokyo, Japan

²NAVER Corp., Seongnam, Korea

³Search Solutions Inc., Seongnam, Korea

ABSTRACT

This paper proposes voicing-aware conditional discriminators for Parallel WaveGAN-based waveform synthesis systems. In this framework, we adopt a projection-based conditioning method that can significantly improve the discriminator's performance. Furthermore, the conventional discriminator is separated into two waveform discriminators for modeling voiced and unvoiced speech. As each discriminator learns the distinctive characteristics of the harmonic and noise components, respectively, the adversarial training process becomes more efficient, allowing the generator to produce more realistic speech waveforms. Subjective test results demonstrate the superiority of the proposed method over the conventional Parallel WaveGAN and WaveNet systems. In particular, our speaker-independently trained model within a FastSpeech 2 based text-to-speech framework achieves the mean opinion scores of 4.20, 4.18, 4.21, and 4.31 for four Japanese speakers, respectively.

“Voicing-aware discriminators”

Toward high-quality synthesis

High-fidelity Parallel WaveGAN with Multi-band Harmonic-plus-Noise Model

Min-Jae Hwang^{1}, Ryuichi Yamamoto^{2*}, Eunwoo Song³ and Jae-Min Kim³*

¹Search Solutions Inc., Seongnam, Korea

²LINE Corp., Tokyo, Japan

³NAVER Corp., Seongnam, Korea

Abstract

This paper proposes a multi-band harmonic-plus-noise (HN) Parallel WaveGAN (PWG) vocoder. To generate a high-fidelity speech signal, it is important to well-reflect the harmonic-noise characteristics of the speech waveform in the time-frequency domain. However, it is difficult for the conventional PWG model to accurately match this condition, as its single generator inefficiently represents the complicated nature of harmonic-noise structures. In the proposed method, the HN WaveNet models are employed to overcome this limitation, which enable the separate generation of the harmonic and noise components of speech signals from the pitch-dependent sine wave and Gaussian noise sources, respectively. Then, the energy ratios between harmonic and noise components in multiple frequency bands (i.e., subband harmonics) are predicted by an additional harmonicity estimator. Weighted by the estimated harmonics, the gain of harmonic and noise components in each subband is adjusted, and finally mixed together to compose the full-band speech signal. Subjective evaluation results showed that the proposed method significantly improved the perceptual quality of the synthesized speech.

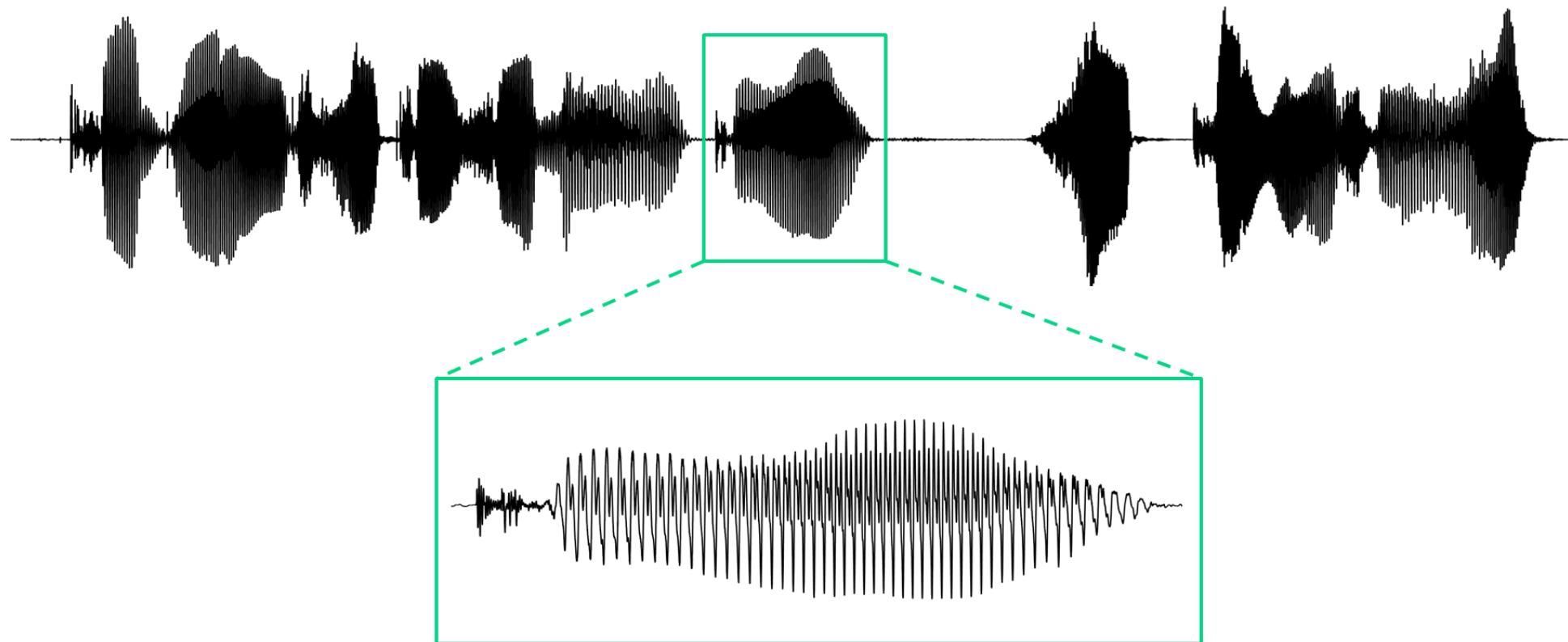
“Harmonic/noise generators”

Parallel waveform synthesis

Toward high-quality synthesis: Speech fundamentals

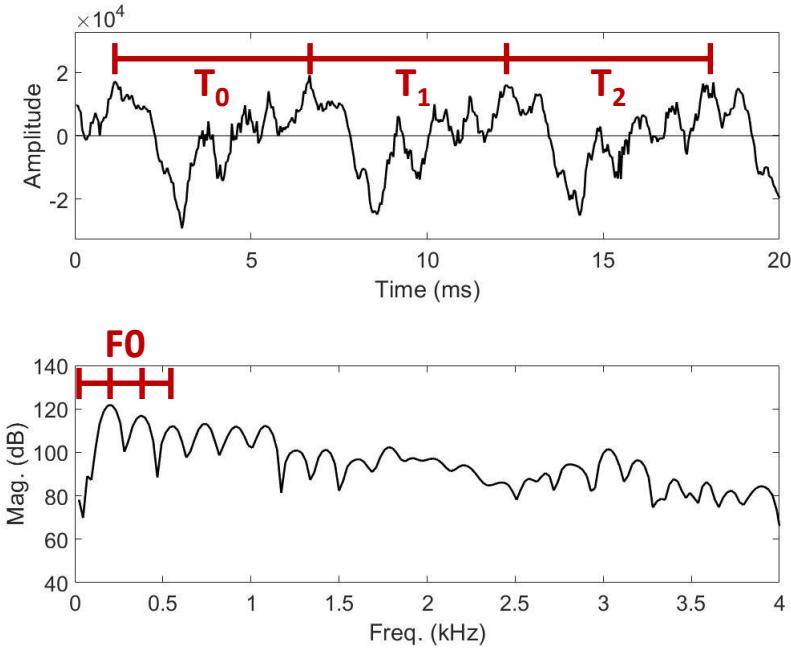
Speech fundamentals

Speech waveform



Speech fundamentals

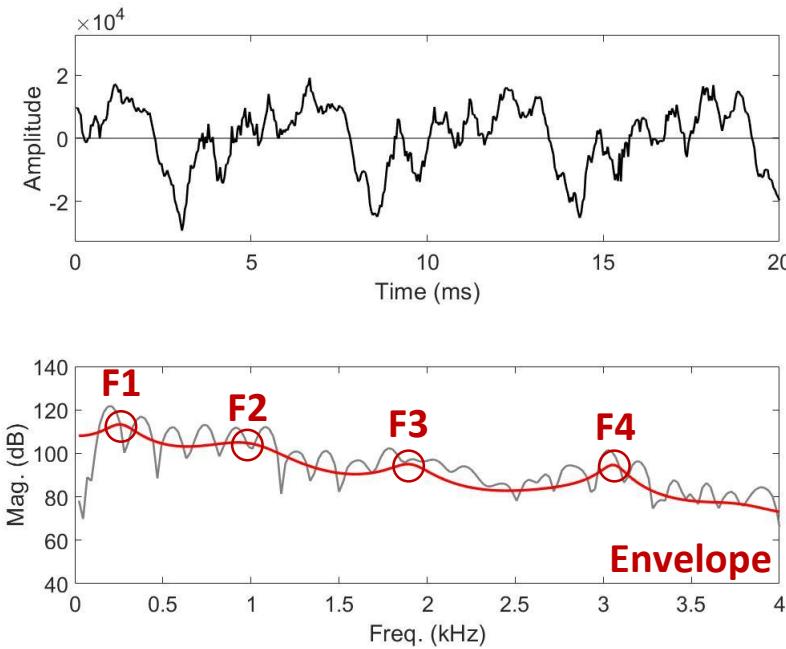
Pitch period



- Pitch period = $T_0 \approx T_1 \approx T_2$
 - Long-term period of speech (time-domain)
- Fundamental frequency (F0) = $1/T_0$
 - $1 / PP$ (frequency-domain)
 - Female voice : Ave. 200 Hz
 - Male voice : Ave. 100 Hz
- Harmonic spectrum
 - Multiple peaks of speech spectrum (interval=F0)
- Formant frequency (F1, F2, ...)
 - Vocal tract resonance

Speech fundamentals

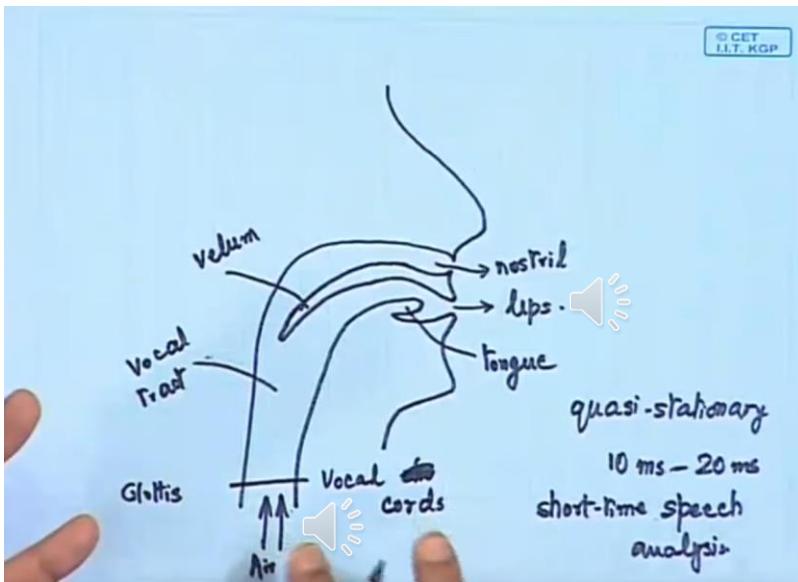
Formant frequency



- Pitch period = $T_0 \approx T_1 \approx T_2$
 - Long-term period of speech (time-domain)
- Fundamental frequency (F0) = $1/T_0$
 - 1 / PP (frequency-domain)
 - Female voice : Ave. 200 Hz
 - Male voice : Ave. 100 Hz
- Harmonic spectrum
 - Multiple peaks of speech spectrum (interval=F0)
- Formant frequency (F1, F2, ...)
 - Vocal tract resonance

How do we produce speech?

Speech production model



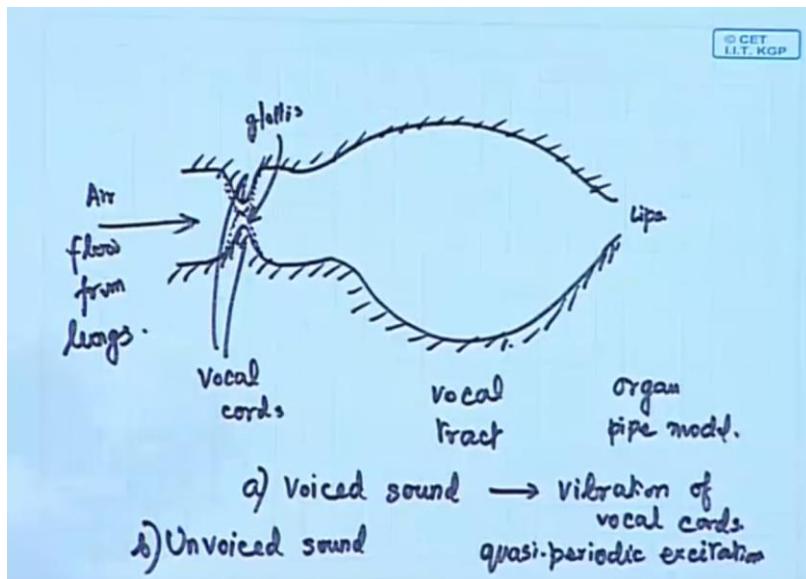
https://www.youtube.com/watch?v=X_JvfZiGEek

- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- Vocal tract filter
 - Shaping voice color



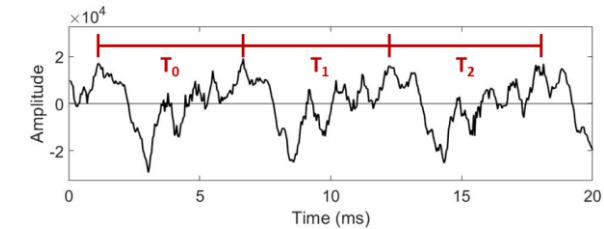
How do we produce speech?

Speech production model



https://www.youtube.com/watch?v=X_JvfZiGEek

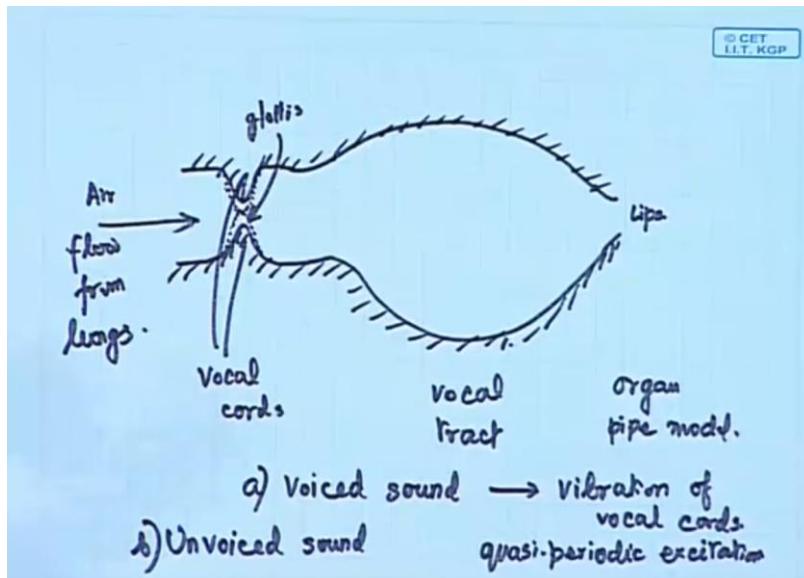
- Lung
 - Power supply
- Vocal source
 - Voiced sound : **quasi-periodic**
 - Unvoiced sound : noisy
- Vocal tract filter
 - Shaping voice color



Source → **Filter** → Speech

How do we produce speech?

Speech production model



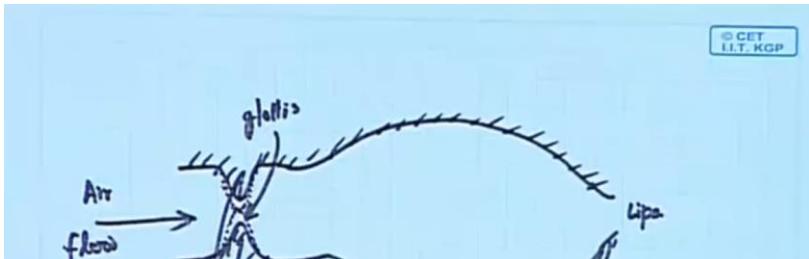
https://www.youtube.com/watch?v=X_JvfZiGEek

- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- **Vocal tract filter**
 - Shaping voice color



How do we produce speech?

Speech production model



- Linear prediction
 - Weighted sum. of previous samples.
 - $\hat{s}(n) = \sum_{k=1}^p a(k)s(n - k)$
- Prediction error
 - Time-domain
 - $e(n) = s(n) - \hat{s}(n) = s(n) - \sum_{k=1}^p a(k)s(n - k)$
 - Minimizing mean square error
 - $\underset{a_k}{\operatorname{argmin}} E \left\{ \|s(n) - \sum_{k=1}^p a(k)s(n - k)\|^2 \right\}$

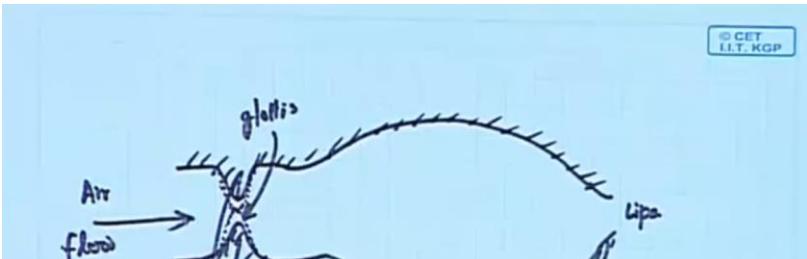


- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- **Vocal tract filter**
 - Shaping voice color



How do we produce speech?

Speech production model



- Linear prediction
 - Weighted sum. of previous samples.
 - $\hat{s}(n) = \sum_{k=1}^p a(k)s(n - k)$
- Prediction error
 - Frequency-domain
 - $S(z) = E(z)H(z) = E(z) \frac{1}{1 - \sum_{k=1}^p a_k z^{-k}}$

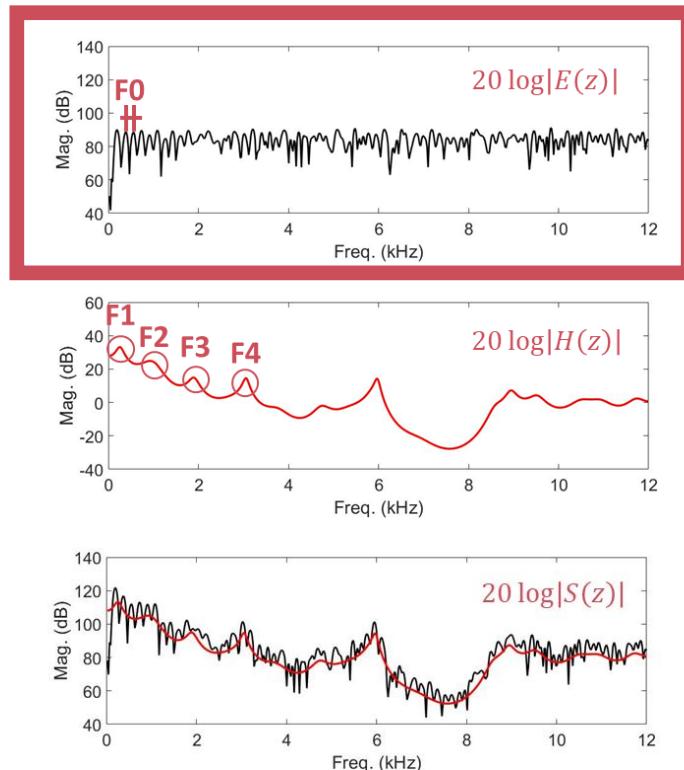
LPC filter

- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- **Vocal tract filter**
 - Shaping voice color



How do we produce speech?

Speech production model

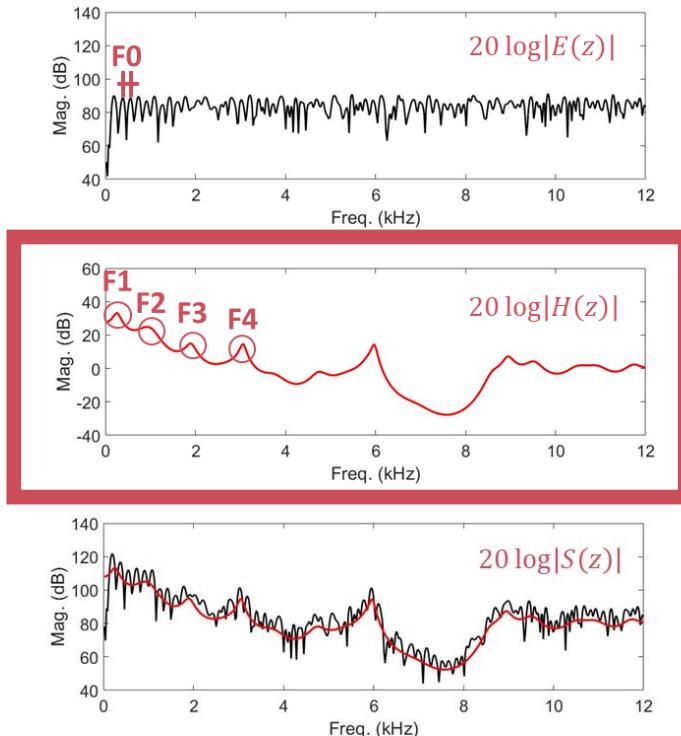


- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- Vocal tract filter
 - Shaping voice color



How do we produce speech?

Speech production model

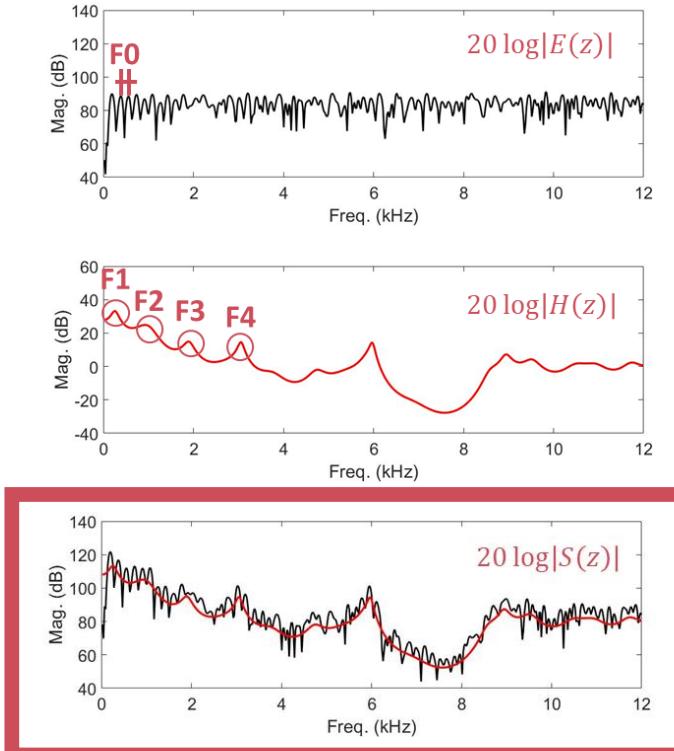


- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- Vocal tract filter
 - Shaping voice color



How do we produce speech?

Speech production model



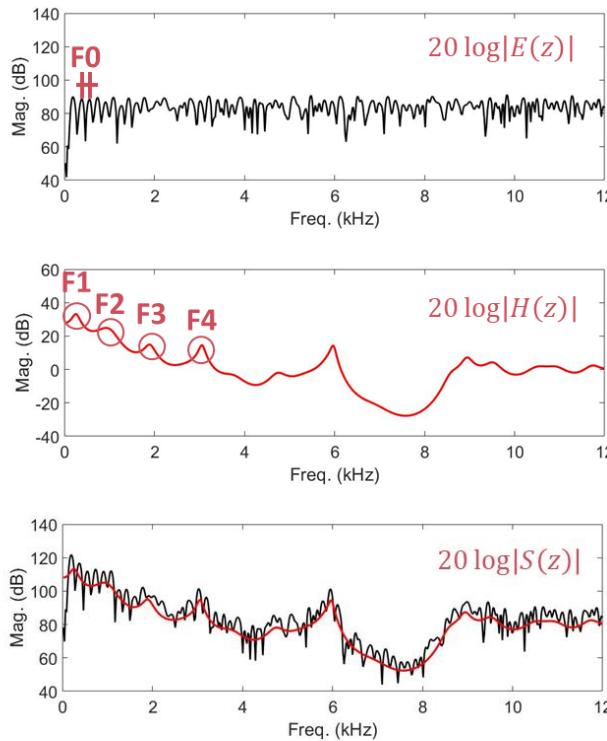
- Lung
 - Power supply
- Vocal source
 - Voiced sound : quasi-periodic
 - Unvoiced sound : noisy
- Vocal tract filter
 - Shaping voice color



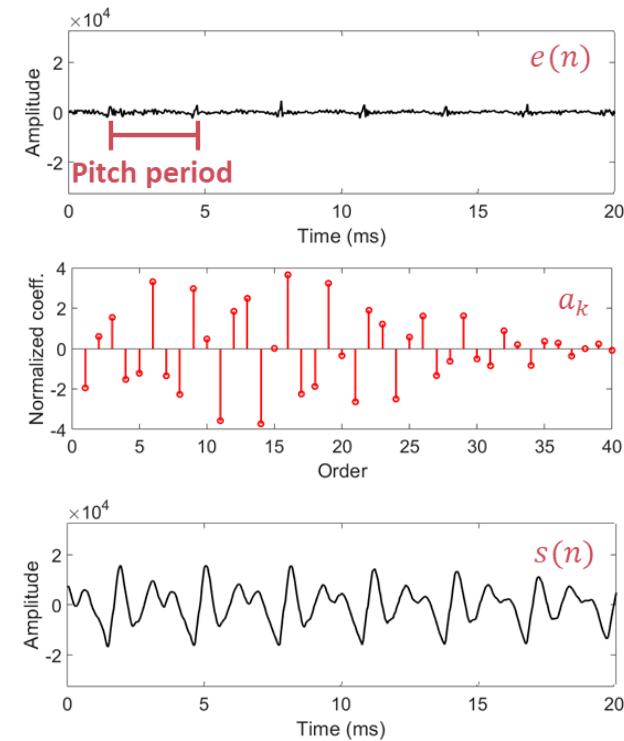
$$S(z) = E(z)H(z) = E(z) \times \frac{1}{1 - \sum_{k=1}^p a_k z^{-k}}$$

How do we produce speech?

Speech production model

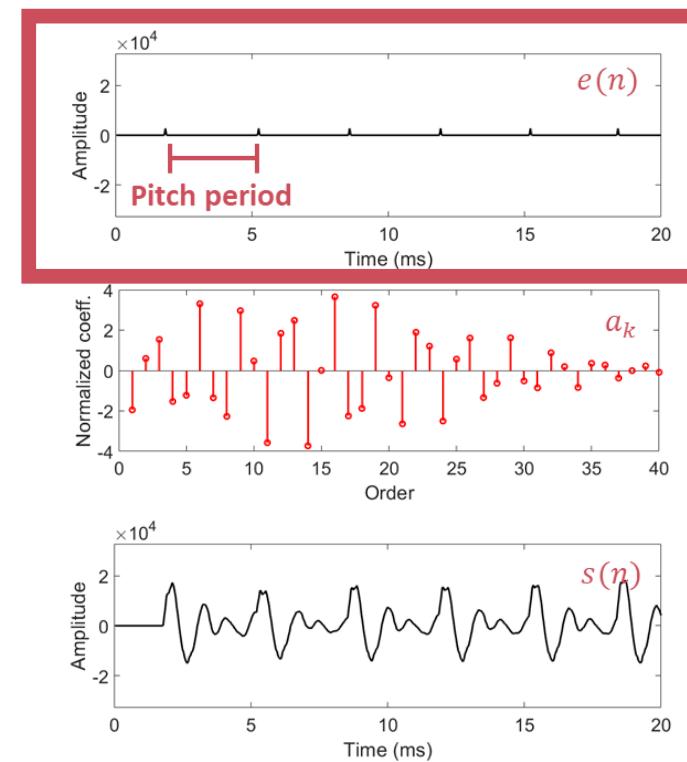
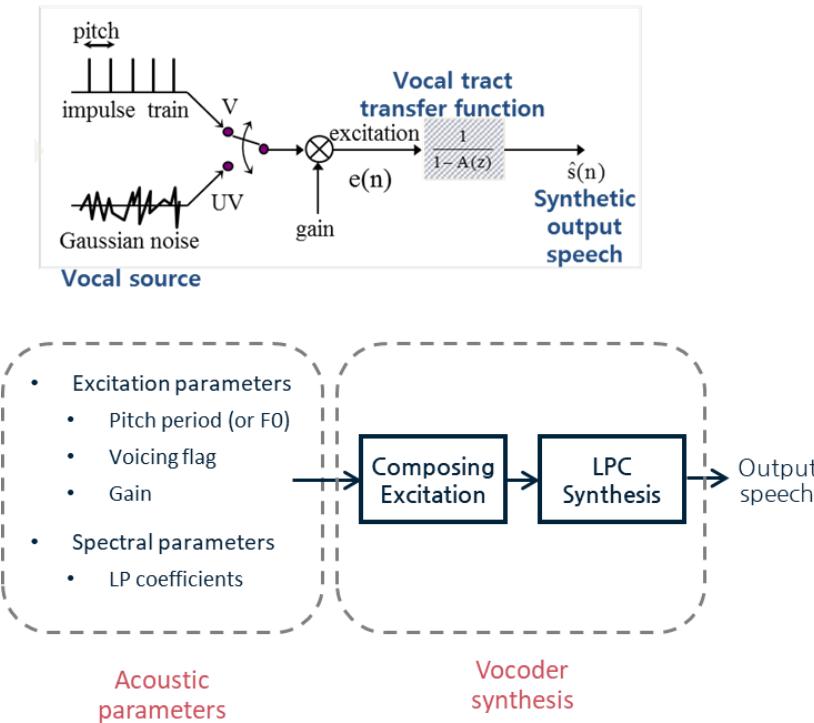


→ Time-domain



How do we produce speech?

Parametric LPC vocoder

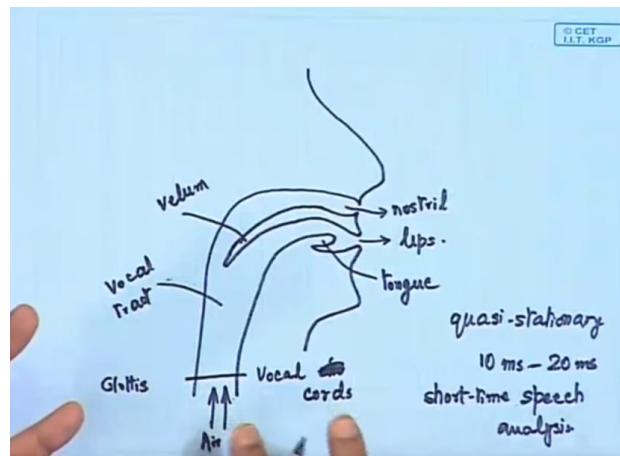


Parallel waveform synthesis

Toward high-quality synthesis: Perceptually **weighted** spectral loss

Perceptually weighted spectral loss

Combining **LPC synthesis filter** with neural **excitation vocoders**



https://www.youtube.com/watch?v=X_JvfZiGEek

Speech production model

Vocal source → Excitation

Voiced sound: quasi-periodic
Unvoiced sound: aperiodic

Vocal tract → **LPC synthesis**
Shaping voice color

Perceptually weighted spectral loss

Combining **LPC** synthesis filter with neural **excitation** vocoders



WaveNet + LPC filter = ExcitNet, LP-WaveNet, ...

WaveRNN + LPC filter = LPCNet

Perceptually weighted spectral loss

Combining **LPC** synthesis filter with neural **excitation** vocoders



WaveNet + LPC filter = ExcitNet, LP-WaveNet, ...

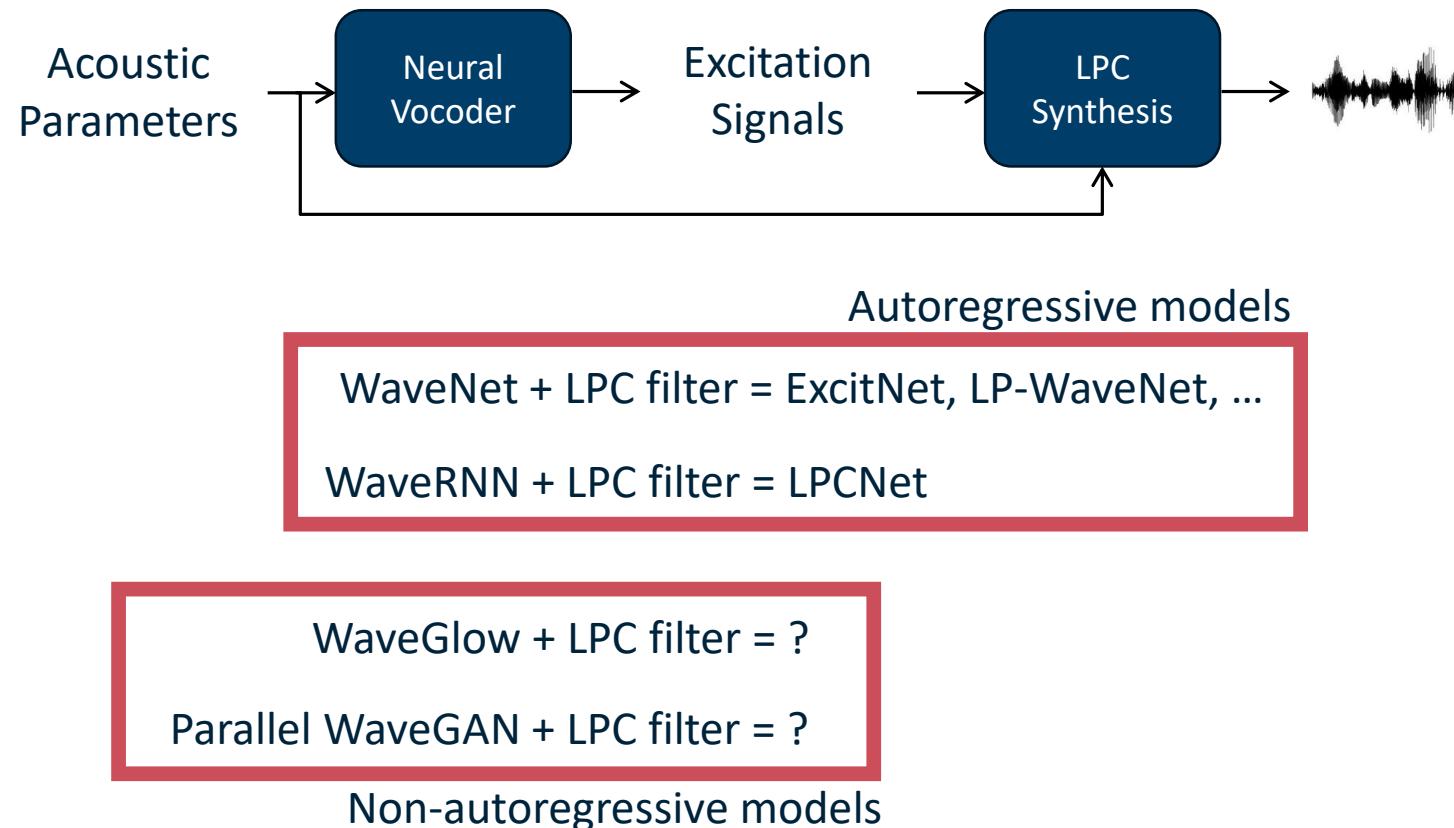
WaveRNN + LPC filter = LPCNet

WaveGlow + LPC filter = ?

Parallel WaveGAN + LPC filter = ?

Perceptually weighted spectral loss

Combining **LPC synthesis filter** with neural **excitation** vocoders



Perceptually weighted spectral loss

Combining **LPC synthesis filter** with neural **excitation vocoders**



Autoregressive models

WaveNet + LPC filter = ExcitNet, LP-WaveNet, ...

WaveRNN + LPC filter = LPCNet

WaveGlow + LPC filter = ?

Parallel WaveGAN + LPC filter = ?

Non-autoregressive models

→ Not suitable for estimating **excitation signals**



Recall: Parallel WaveGAN

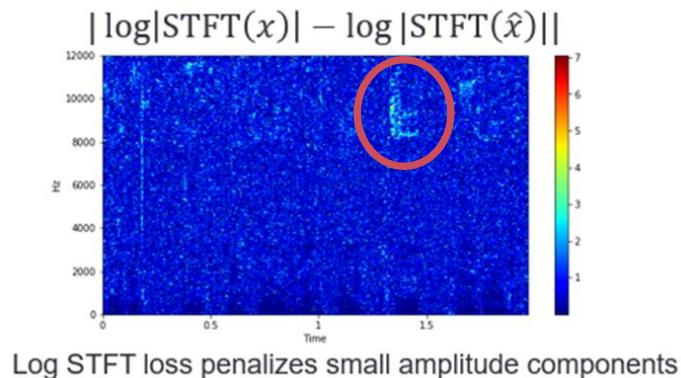
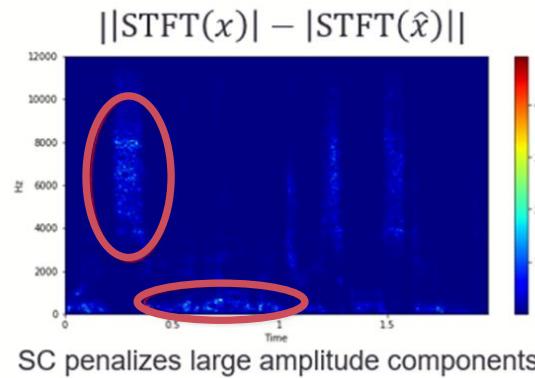
1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the **multi-resolution STFT loss**

STFT is calculated in different T/F resolutions

There are two loss functions

One penalizes **large energy** components

The other penalizes **small energy** components



$$L_{\text{mr.stft}}(G) = \frac{1}{M} \sum_{m=1}^M L_{\text{stft}}^{(m)}(G)$$

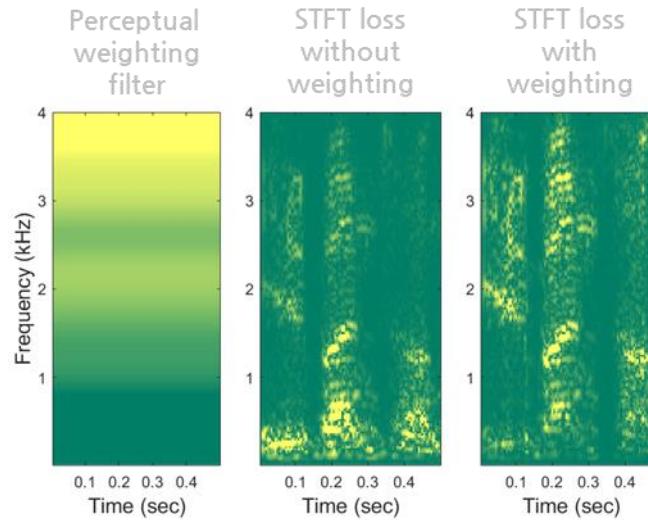
$$L_{\text{stft}}(G) = \mathbb{E}_{z \sim p_z, x \sim p_{\text{data}}} [L_{\text{sc}}(x, \hat{x}) + L_{\text{mag}}(x, \hat{x})]$$

$$L_{\text{sc}}(x, \hat{x}) = \frac{\sqrt{\sum_{t,f} (|\mathbf{X}_{t,f}| - |\hat{\mathbf{X}}_{t,f}|)^2}}{\sqrt{\sum_{t,f} |\mathbf{X}_{t,f}|^2}}$$

$$L_{\text{mag}}(x, \hat{x}) = \frac{\sum_{t,f} |\log|\mathbf{X}_{t,f}| - \log|\hat{\mathbf{X}}_{t,f}||}{T \cdot N}$$

Perceptually weighted spectral loss

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the multi-resolution STFT loss
+ Applying perceptual weighting filter



$$L_{\text{sc}}^w(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sqrt{\sum_{t,f} (\mathbf{W}_{t,f}(|\mathbf{X}_{t,f}|) - |\hat{\mathbf{X}}_{t,f}|))^2}}{\sqrt{\sum_{t,f} |\mathbf{X}_{t,f}|^2}}$$

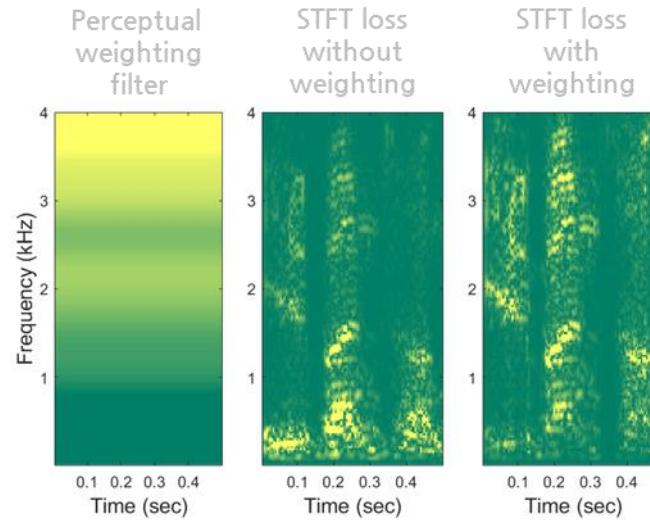
$$L_{\text{mag}}^w(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sum_{t,f} |\log \mathbf{W}_{t,f}(\log |\mathbf{X}_{t,f}|) - \log |\hat{\mathbf{X}}_{t,f}|)|}{T \cdot N}$$

$$\mathbf{W}(z) = 1 - \sum_{k=1}^p \tilde{\alpha}_k z^{-k}$$

Perceptually weighted spectral loss

1. Removed the teacher-student distillation process
2. Improved synthetic quality by using the adversarial training method
3. Further improved its quality by introducing the multi-resolution STFT loss
+ Applying perceptual weighting filter

This penalizes **perceptually-sensitive errors** in the freq. domain



$$L_{\text{sc}}^w(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sqrt{\sum_{t,f} \mathbf{W}_{t,f} (|\mathbf{X}_{t,f}| - |\hat{\mathbf{X}}_{t,f}|)^2}}{\sqrt{\sum_{t,f} |\mathbf{X}_{t,f}|^2}}$$

$$L_{\text{mag}}^w(\mathbf{x}, \hat{\mathbf{x}}) = \frac{\sum_{t,f} \log \mathbf{W}_{t,f} (\log |\mathbf{X}_{t,f}| - \log |\hat{\mathbf{X}}_{t,f}|)}{T \cdot N}$$

$$\mathbf{W}(z) = 1 - \sum_{k=1}^p \tilde{\alpha}_k z^{-k}$$

Perceptually weighted spectral loss

Evaluation results

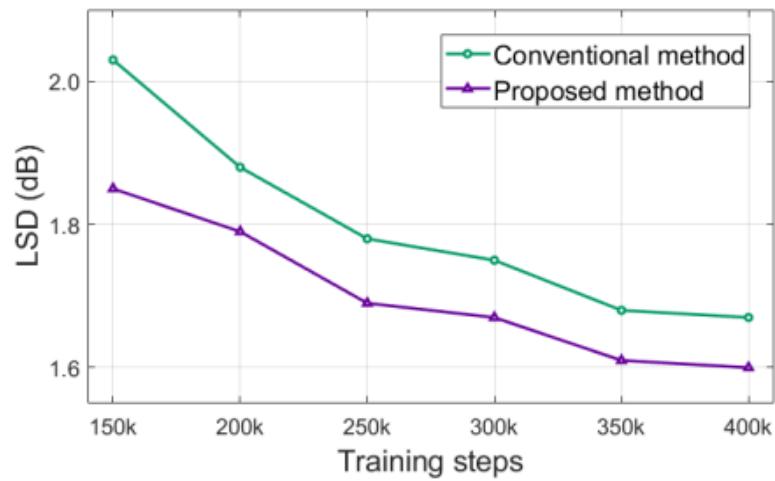


Fig. 2: Log-spectral distance (LSD; dB) between the original and generated speech signals

Table 4: Naturalness MOS test results with 95% confidence intervals for the TTS systems with respect to the different vocoding models: The MOS results for the proposed system are in bold font. The KRF and KRM denote Korean female and male speakers, respectively.

Index	Model	KRF	KRM
Test 1	WaveNet	3.64 ± 0.14	3.60 ± 0.13
Test 2	WaveNet + NS	4.36 ± 0.11	4.32 ± 0.10
Test 3	Parallel WaveGAN	4.02 ± 0.10	4.11 ± 0.11
Test 4	Parallel WaveGAN + NS	2.34 ± 0.10	1.72 ± 0.09
Test 5	Parallel WaveGAN + PW	4.26 ± 0.10	4.21 ± 0.10
Test 6	Raw	4.64 ± 0.07	4.59 ± 0.09

Acoustic model: Tacotron 2

NS: Noise-shaping (similar to LPC synthesis)

Perceptually weighted spectral loss

Evaluation results

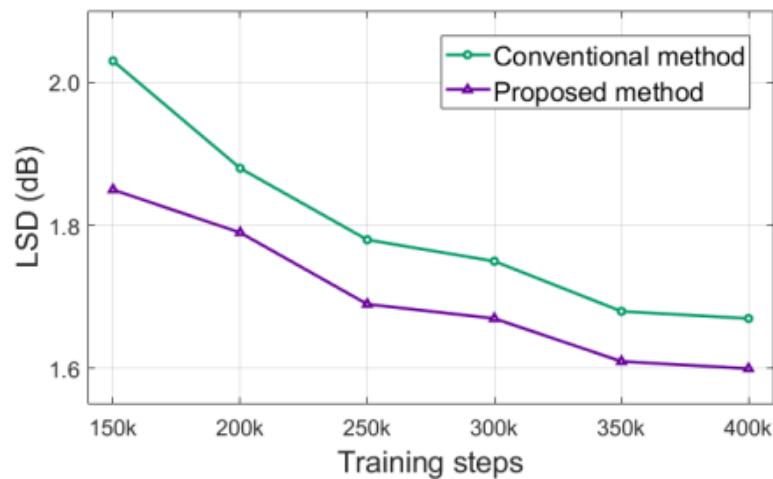


Fig. 2: Log-spectral distance (LSD; dB) between the original and generated speech signals

Table 4: Naturalness MOS test results with 95% confidence intervals for the TTS systems with respect to the different vocoding models: The MOS results for the proposed system are in bold font. The KRF and KRM denote Korean female and male speakers, respectively.

Index	Model	KRF	KRM
Test 1	WaveNet	3.64 ± 0.14	3.60 ± 0.13
Test 2	WaveNet + NS	4.36 ± 0.11	4.32 ± 0.10
Test 3	Parallel WaveGAN	4.02 ± 0.10	4.11 ± 0.11
Test 4	Parallel WaveGAN + NS	2.34 ± 0.10	1.72 ± 0.09
Test 5	Parallel WaveGAN + PW	4.26 ± 0.10	4.21 ± 0.10
Test 6	Raw	4.64 ± 0.07	4.59 ± 0.09

Acoustic model: Tacotron 2

NS: Noise-shaping (similar to LPC synthesis)

Perceptually weighted spectral loss

Evaluation results

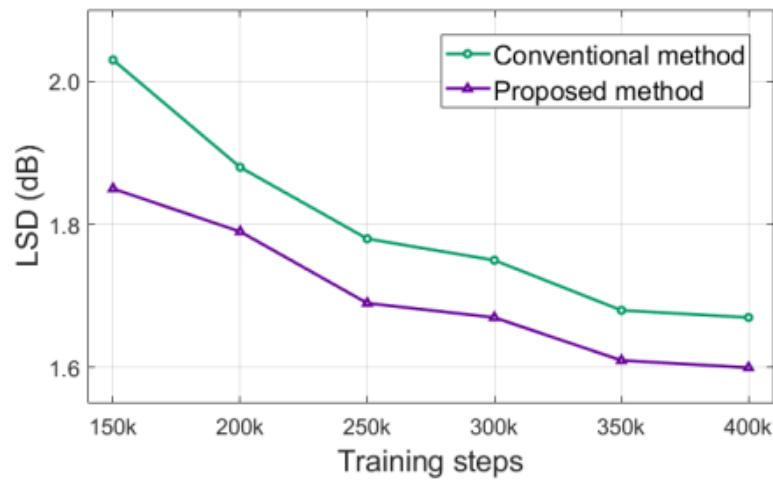


Fig. 2: Log-spectral distance (LSD; dB) between the original and generated speech signals

Table 4: Naturalness MOS test results with 95% confidence intervals for the TTS systems with respect to the different vocoding models: The MOS results for the proposed system are in bold font. The KRF and KRM denote Korean female and male speakers, respectively.

Index	Model	KRF	KRM
Test 1	WaveNet	3.64 ± 0.14	3.60 ± 0.13
Test 2	WaveNet + NS	4.36 ± 0.11	4.32 ± 0.10
Test 3	Parallel WaveGAN	4.02 ± 0.10	4.11 ± 0.11
Test 4	Parallel WaveGAN + NS	2.34 ± 0.10	1.72 ± 0.09
Test 5	Parallel WaveGAN + PW	4.26 ± 0.10	4.21 ± 0.10
Test 6	Raw	4.64 ± 0.07	4.59 ± 0.09

Acoustic model: Tacotron 2

NS: Noise-shaping (similar to LPC synthesis)

Perceptually weighted spectral loss

Evaluation results

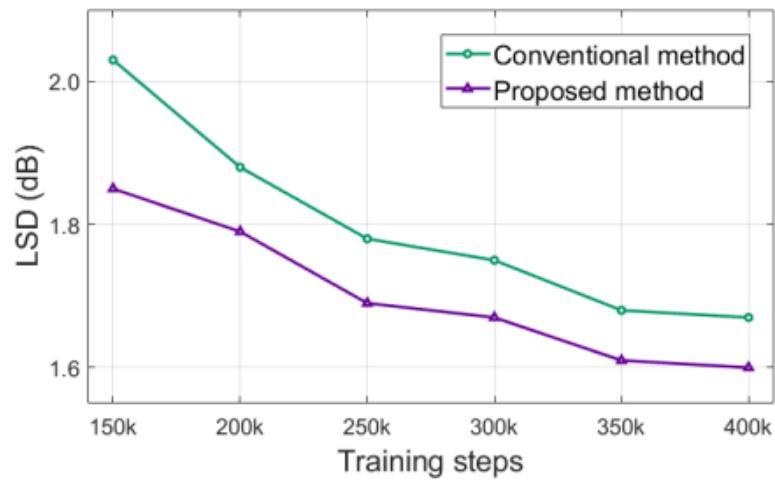


Fig. 2: Log-spectral distance (LSD; dB) between the original and generated speech signals

Table 4: Naturalness MOS test results with 95% confidence intervals for the TTS systems with respect to the different vocoding models: The MOS results for the proposed system are in bold font. The KRF and KRM denote Korean female and male speakers, respectively.

Index	Model	KRF	KRM
Test 1	WaveNet	3.64 ± 0.14	3.60 ± 0.13
Test 2	WaveNet + NS	4.36 ± 0.11	4.32 ± 0.10
Test 3	Parallel WaveGAN	4.02 ± 0.10	4.11 ± 0.11
Test 4	Parallel WaveGAN + NS	2.34 ± 0.10	1.72 ± 0.09
Test 5	Parallel WaveGAN + PW	4.26 ± 0.10	4.21 ± 0.10
Test 6	Raw	4.64 ± 0.07	4.59 ± 0.09

Acoustic model: Tacotron 2

NS: Noise-shaping (similar to LPC synthesis)

Perceptually weighted spectral loss



Demo samples

Parallel waveform synthesis

Toward high-quality synthesis: **Voicing-aware discriminators**

Voicing-aware discriminators

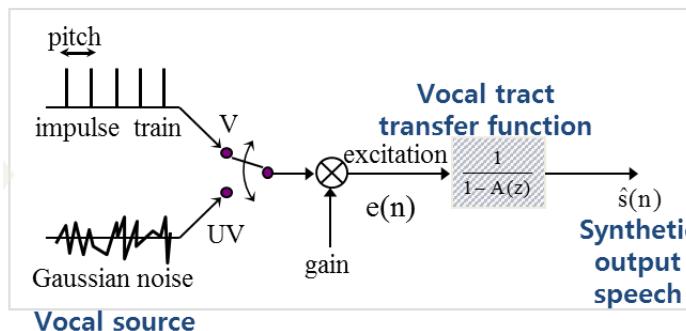
Voiced/unvoiced sounds



Voiced sound: Quasi-periodic



Unvoiced sound: aperiodic



Voicing-aware discriminators

Voiced/unvoiced sounds



Voiced sound: Quasi-periodic



Unvoiced sound: aperiodic

V: Characterized by **slowly evolving harmonic** components

Discriminator should cover **long-term variations** of voiced sound

Voicing-aware discriminators

Voiced/unvoiced sounds



Voiced sound: Quasi-periodic



Unvoiced sound: aperiodic

V: Characterized by slowly evolving harmonic components

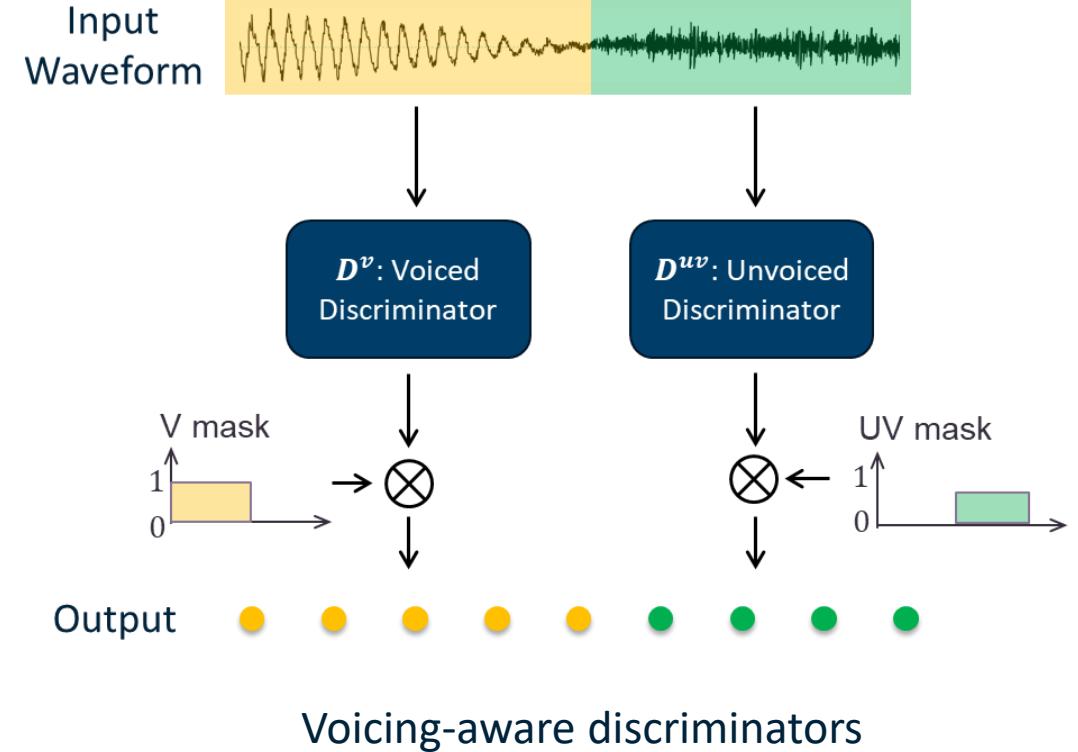
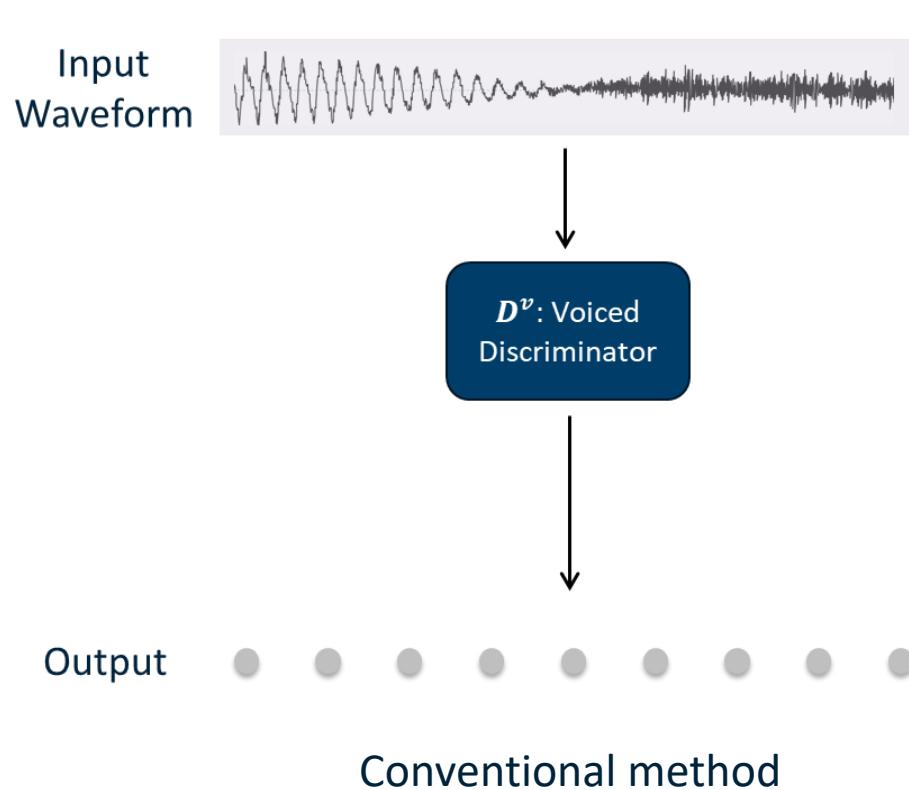
Discriminator should cover long-term variations of voiced sound

UV: Characterized by rapidly evolving noise components

Discriminator should catch short-term variations of unvoiced sound

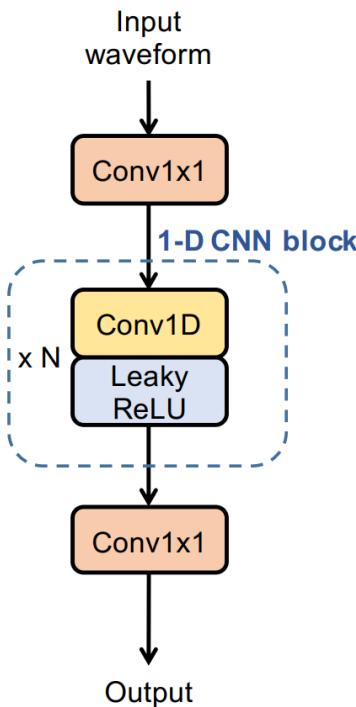
Voicing-aware discriminators

Voiced/unvoiced masking

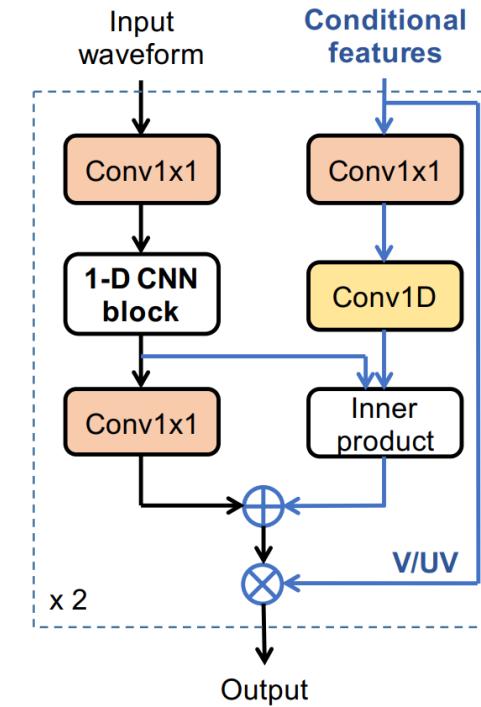


Voicing-aware discriminators

Voiced/unvoiced masking



Conventional method



Voicing-aware discriminators

T. Miyato, et al., “cGANs with projection discriminator,” Proc. ICLR, 2018.

Voicing-aware discriminators

Receptive field

Table 1. The dilation factors and receptive fields in the 1-D CNN blocks of the voicing-aware discriminators.

Discriminator	Dilation factors	Receptive field
D^v	[1, 2, 4, 8, 16, 32]	127
D^{uv}	[1, 1, 1, 1, 1, 1]	13

Voiced discriminator

Dilated convolution with **long** receptive field
Covering **long-term variations** of voiced sound

Unvoiced discriminator

Non-dilated convolution with **short** receptive field
Catching **short-term variations** of unvoiced sound

Voicing-aware discriminators

Receptive field

Table 1. The dilation f
blocks of the voicing-aw

Discriminator
D^v
D^{uv}

Voiced discriminator

Dilated convolutional
Covering long-term

Unvoiced discriminator

Non-dilated convolutional
Catching short-term

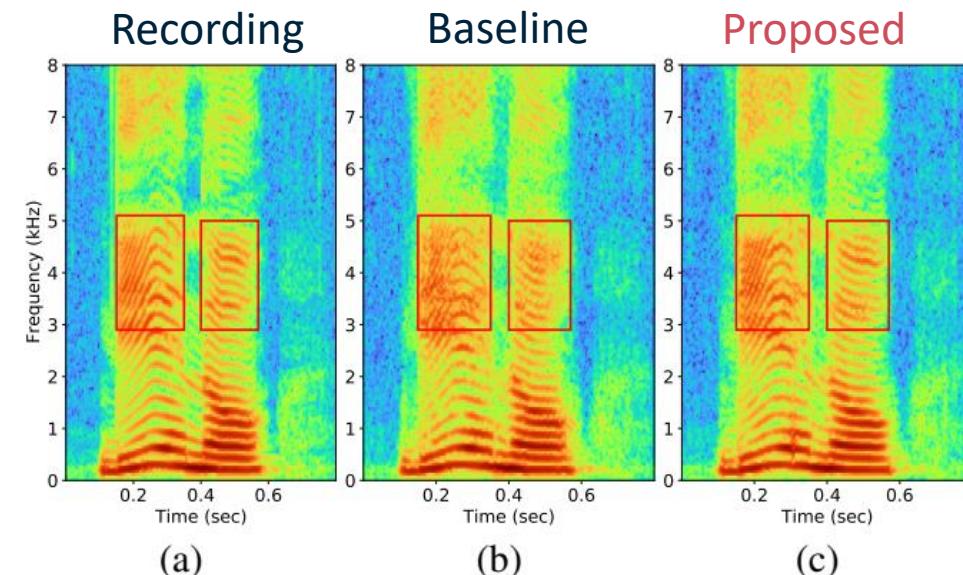


Fig. 2. Spectrograms of (a) natural speech, (b) generated speech from the conventional Parallel WaveGAN (S2), and (c) generated speech from the proposed Parallel WaveGAN (S7). As demonstrated in rectangle areas, our proposed method is able to model spectral harmonics more accurately.

Voicing-aware discriminators

Evaluation results

Table 2. MOS test results with 95% confidence intervals in analysis/synthesis: The speech samples were generated using the acoustic features extracted from the recorded speech. PWG denotes Parallel WaveGAN for short. Note that systems S2 and S3 used D^v as the primary discriminator. All the models were trained in a speaker-independent manner.

System	Model	Voiced segments	Unvoiced segments	Discriminator conditioning	MOS			
					F1	F2	M1	M2
S1	WaveNet	-	-	-	3.64±0.12	3.83±0.11	3.33±0.12	3.13±0.11
S2	PWG	-	-	-	3.61±0.11	3.55±0.11	3.57±0.12	3.61±0.11
S3	PWG-cGAN-D	-	-	Yes	4.04±0.10	3.95±0.10	3.91±0.11	3.97±0.10
S4	PWG-V/UV-D	D^v	D^v	Yes	3.60±0.12	3.59±0.11	3.34±0.11	3.48±0.11
S5	PWG-V/UV-D	D^{uv}	D^v	Yes	3.67±0.11	3.48±0.11	3.29±0.12	3.38±0.11
S6	PWG-V/UV-D	D^{uv}	D^{uv}	Yes	3.77±0.11	3.88±0.10	3.57±0.11	3.34±0.11
S7	PWG-V/UV-D (proposed)	D^v	D^{uv}	Yes	4.11±0.10	4.05±0.10	4.04±0.10	4.08±0.10
R1	Recordings	-	-	-	4.63±0.08	4.67±0.07	4.61±0.08	4.64±0.08

Table 3. MOS test results with 95% confidence intervals: Acoustic features generated from the FastSpeech 2 acoustic model were used to compose the input auxiliary features.

System	Model	MOS			
		F1	F2	M1	M2
S1	FastSpeech 2 + WaveNet	3.90±0.11	3.81±0.10	3.43±0.11	3.09±0.10
S2	FastSpeech 2 + PWG	3.76±0.11	3.62±0.11	3.63±0.11	3.78±0.10
S3	FastSpeech 2 + PWG-cGAN-D	4.02±0.10	4.03±0.10	4.16±0.10	4.06±0.10
S7	FastSpeech 2 + PWG-V/UV-D (proposed)	4.20±0.10	4.18±0.09	4.21±0.09	4.31±0.09
R1	Recordings	4.63±0.08	4.67±0.07	4.61±0.08	4.64±0.08

Voicing-aware discriminators

Evaluation results

Table 2. MOS test results with 95% confidence intervals in analysis/synthesis: The speech samples were generated using the acoustic features extracted from the recorded speech. PWG denotes Parallel WaveGAN for short. Note that systems S2 and S3 used D^v as the primary discriminator. All the models were trained in a speaker-independent manner.

System	Model	Voiced segments	Unvoiced segments	Discriminator conditioning	MOS			
					F1	F2	M1	M2
S1	WaveNet	-	-	-	3.64±0.12	3.83±0.11	3.33±0.12	3.13±0.11
S2	PWG	-	-	-	3.61±0.11	3.55±0.11	3.57±0.12	3.61±0.11
S3	PWG-cGAN-D	-	-	Yes	4.04±0.10	3.95±0.10	3.91±0.11	3.97±0.10
S4	PWG-V/UV-D	D^v	D^v	Yes	3.60±0.12	3.59±0.11	3.34±0.11	3.48±0.11
S5	PWG-V/UV-D	D^{uv}	D^v	Yes	3.67±0.11	3.48±0.11	3.29±0.12	3.38±0.11
S6	PWG-V/UV-D	D^{uv}	D^{uv}	Yes	3.77±0.11	3.88±0.10	3.57±0.11	3.34±0.11
S7	PWG-V/UV-D (proposed)	D^v	D^{uv}	Yes	4.11±0.10	4.05±0.10	4.04±0.10	4.08±0.10
R1	Recordings	-	-	-	4.63±0.08	4.67±0.07	4.61±0.08	4.64±0.08

Table 3. MOS test results with 95% confidence intervals: Acoustic features generated from the FastSpeech 2 acoustic model were used to compose the input auxiliary features.

System	Model	MOS			
		F1	F2	M1	M2
S1	FastSpeech 2 + WaveNet	3.90±0.11	3.81±0.10	3.43±0.11	3.09±0.10
S2	FastSpeech 2 + PWG	3.76±0.11	3.62±0.11	3.63±0.11	3.78±0.10
S3	FastSpeech 2 + PWG-cGAN-D	4.02±0.10	4.03±0.10	4.16±0.10	4.06±0.10
S7	FastSpeech 2 + PWG-V/UV-D (proposed)	4.20±0.10	4.18±0.09	4.21±0.09	4.31±0.09
R1	Recordings	4.63±0.08	4.67±0.07	4.61±0.08	4.64±0.08

Voicing-aware discriminators



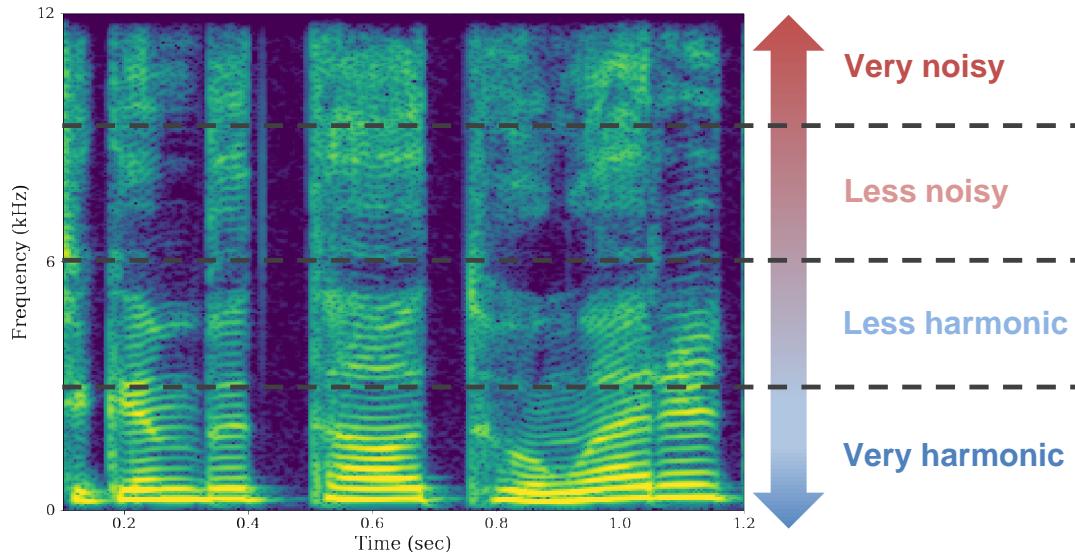
Demo samples

Parallel waveform synthesis

Toward high-quality synthesis: **Harmonic/noise generators**

Harmonic/noise generators

Harmonicity analysis in the frequency domain



Low frequency region

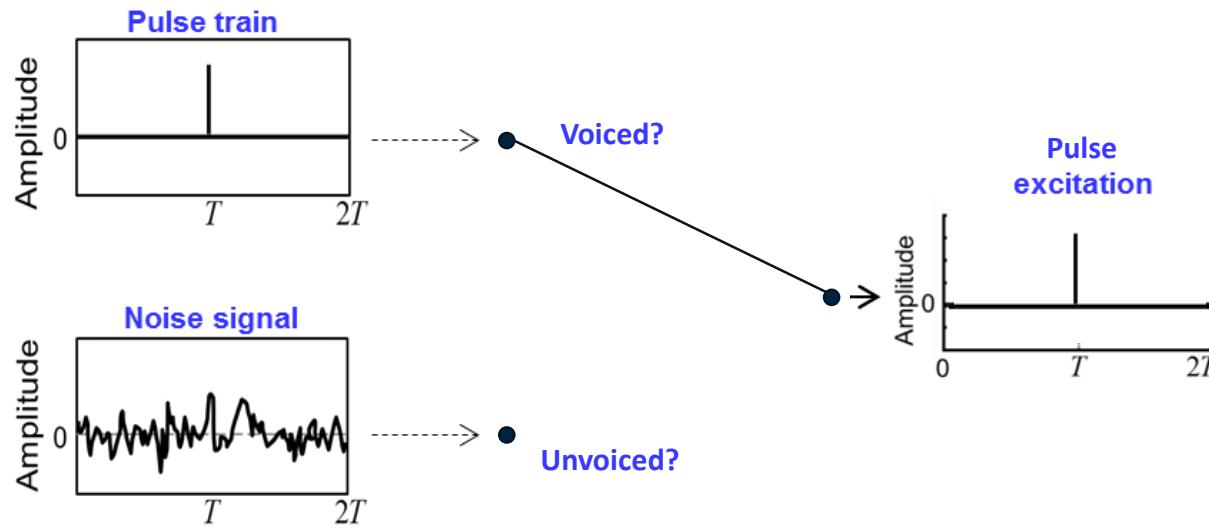
Harmonic characteristics > Noise characteristics

High frequency region

Harmonic characteristics < Noise characteristics

Harmonic/noise generators

Parametric LPC vocoder (binary decision)



Low frequency region

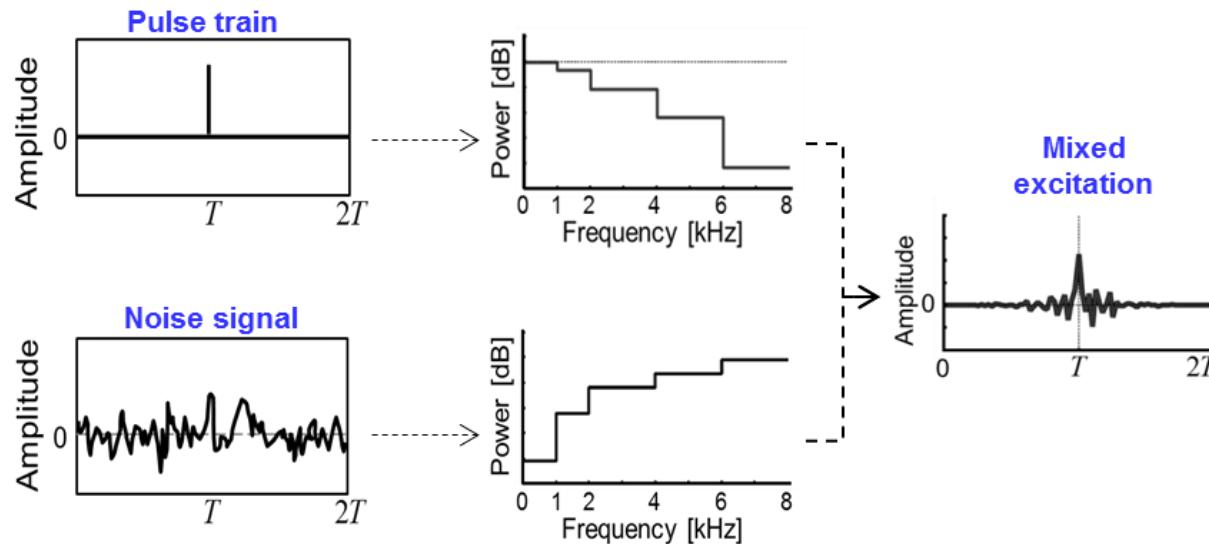
Harmonic characteristics > Noise characteristics

High frequency region

Harmonic characteristics < Noise characteristics

Harmonic/noise generators

Mixed excitation-based parametric vocoder



Low frequency region

Harmonic characteristics > Noise characteristics

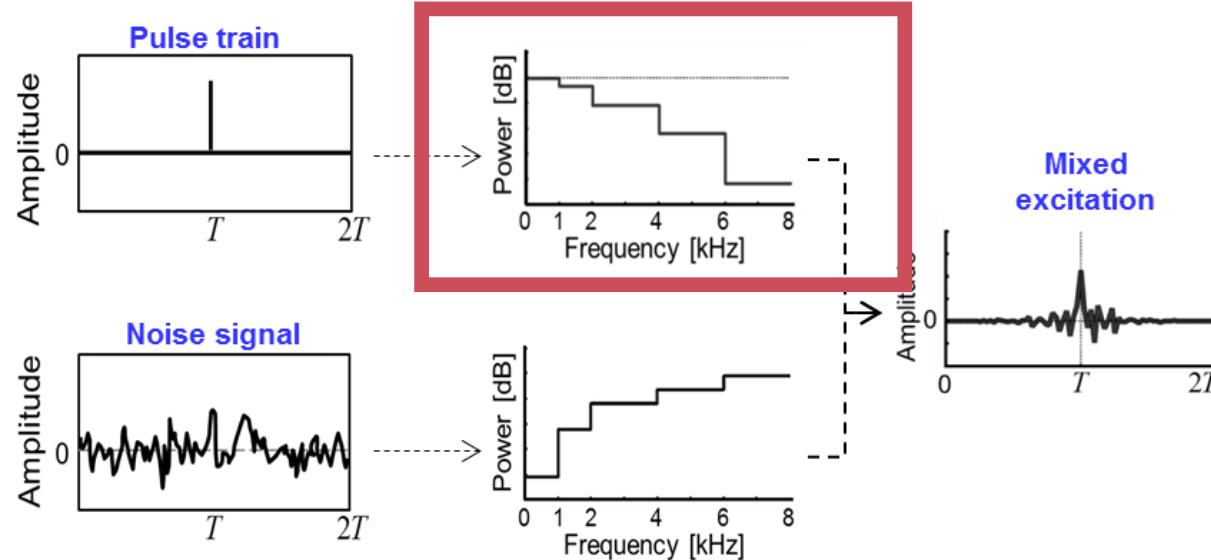
High frequency region

Harmonic characteristics < Noise characteristics

Harmonic/noise generators

Mixed excitation-based parametric vocoder

How periodic? → Harmonicity (ex. MELP and MBE vocoders)



Low frequency region

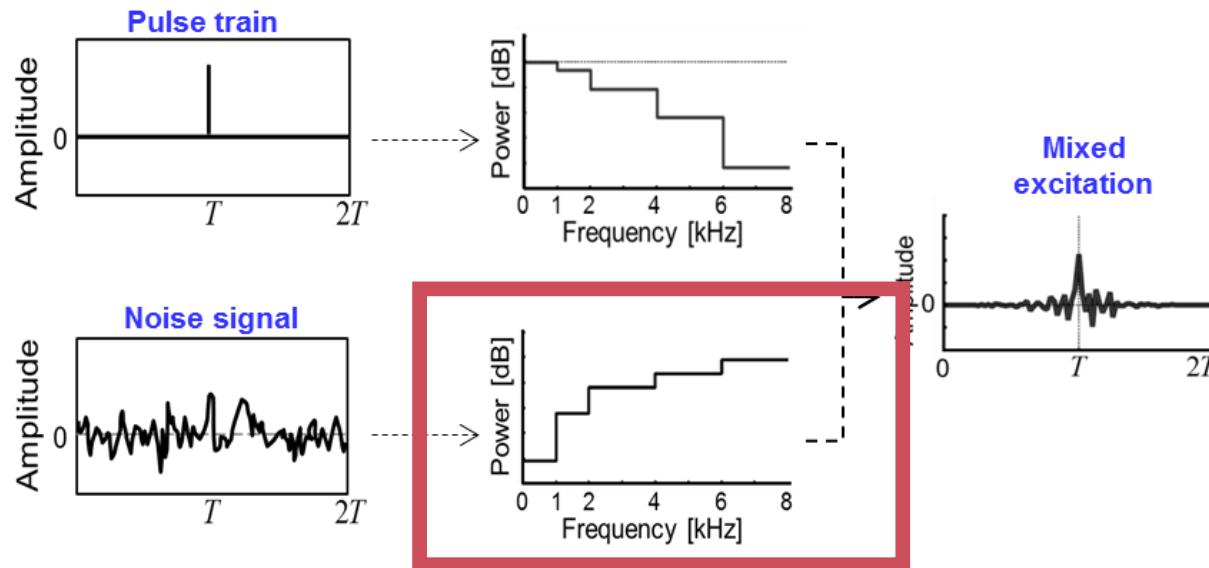
Harmonic characteristics > Noise characteristics

High frequency region

Harmonic characteristics < Noise characteristics

Harmonic/noise generators

Mixed excitation-based parametric vocoder



How aperiodic? → aperiodicity (ex. STRAIGHT and WORLD vocoders)

Low frequency region

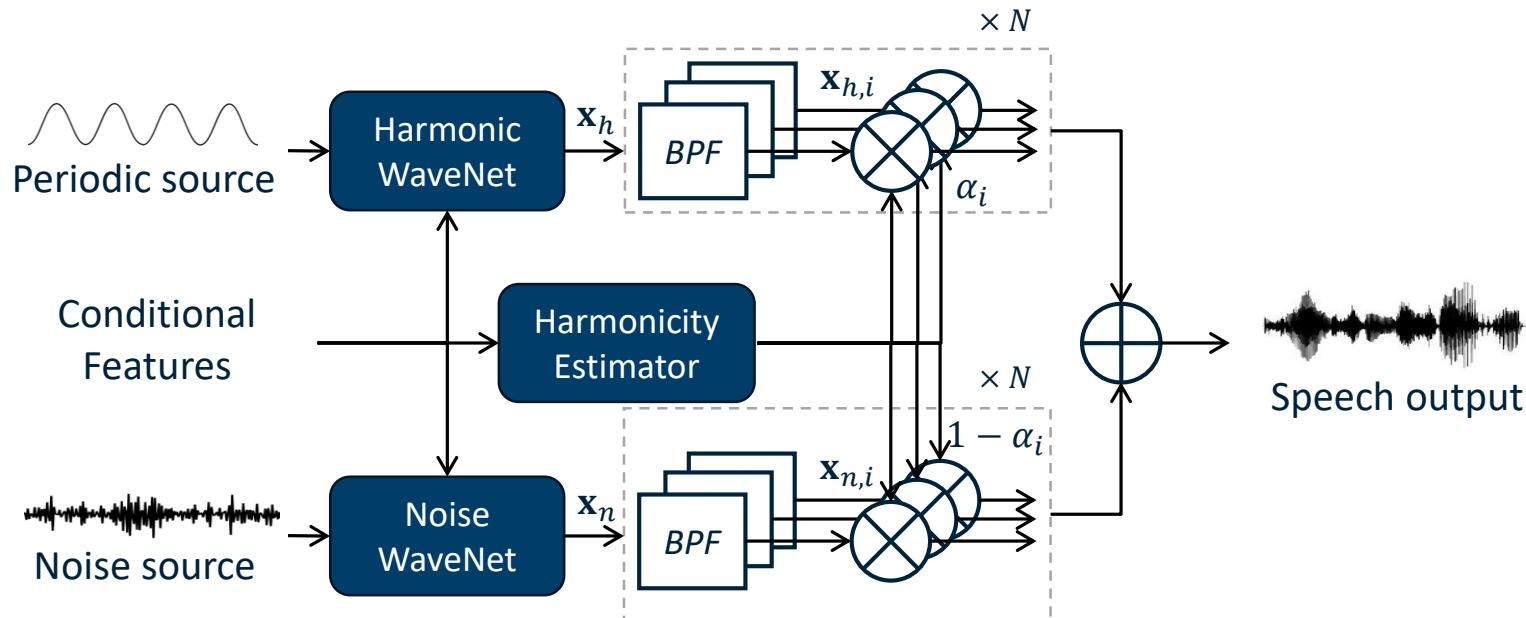
Harmonic characteristics > Noise characteristics

High frequency region

Harmonic characteristics < Noise characteristics

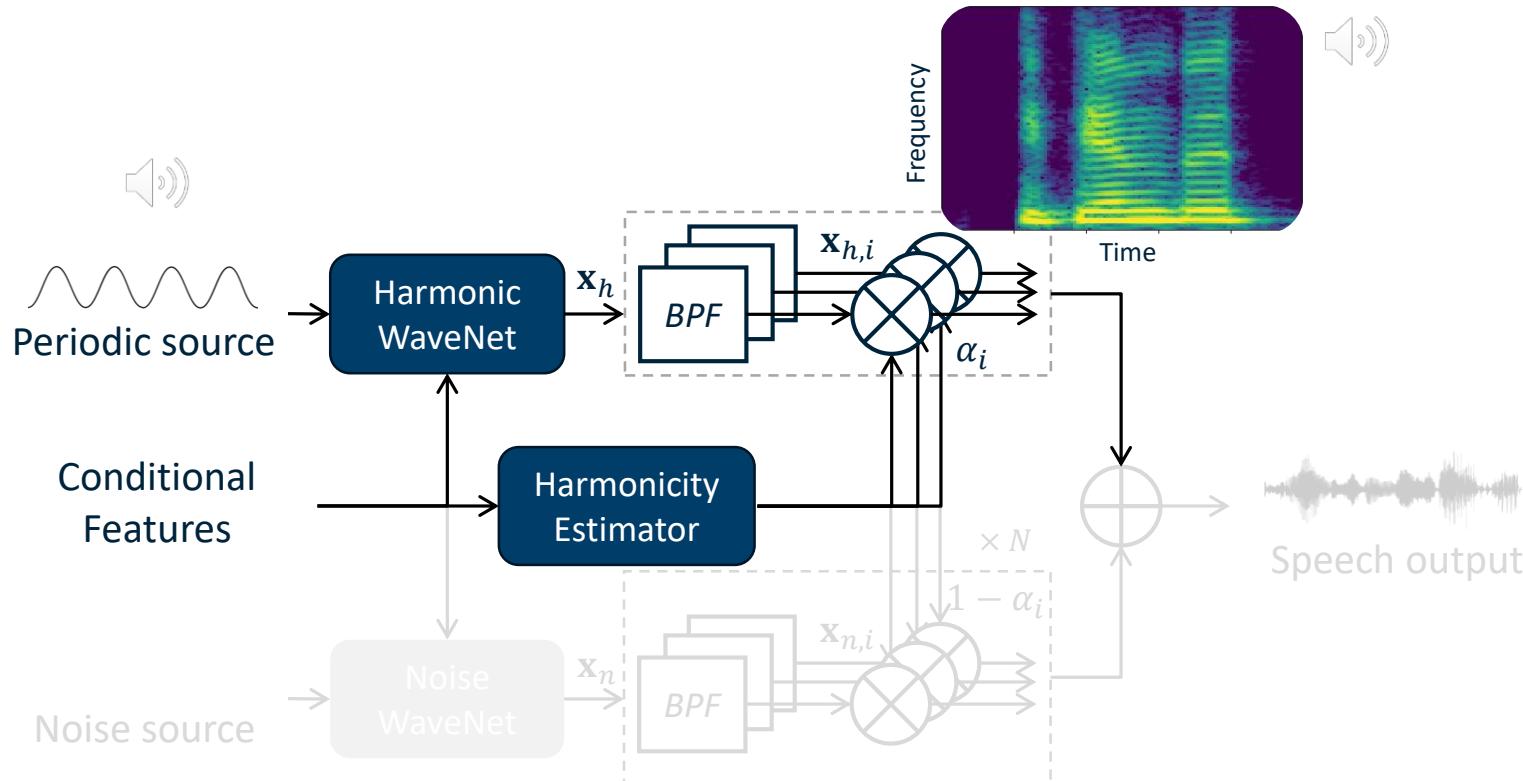
Harmonic/noise generators

Model architecture



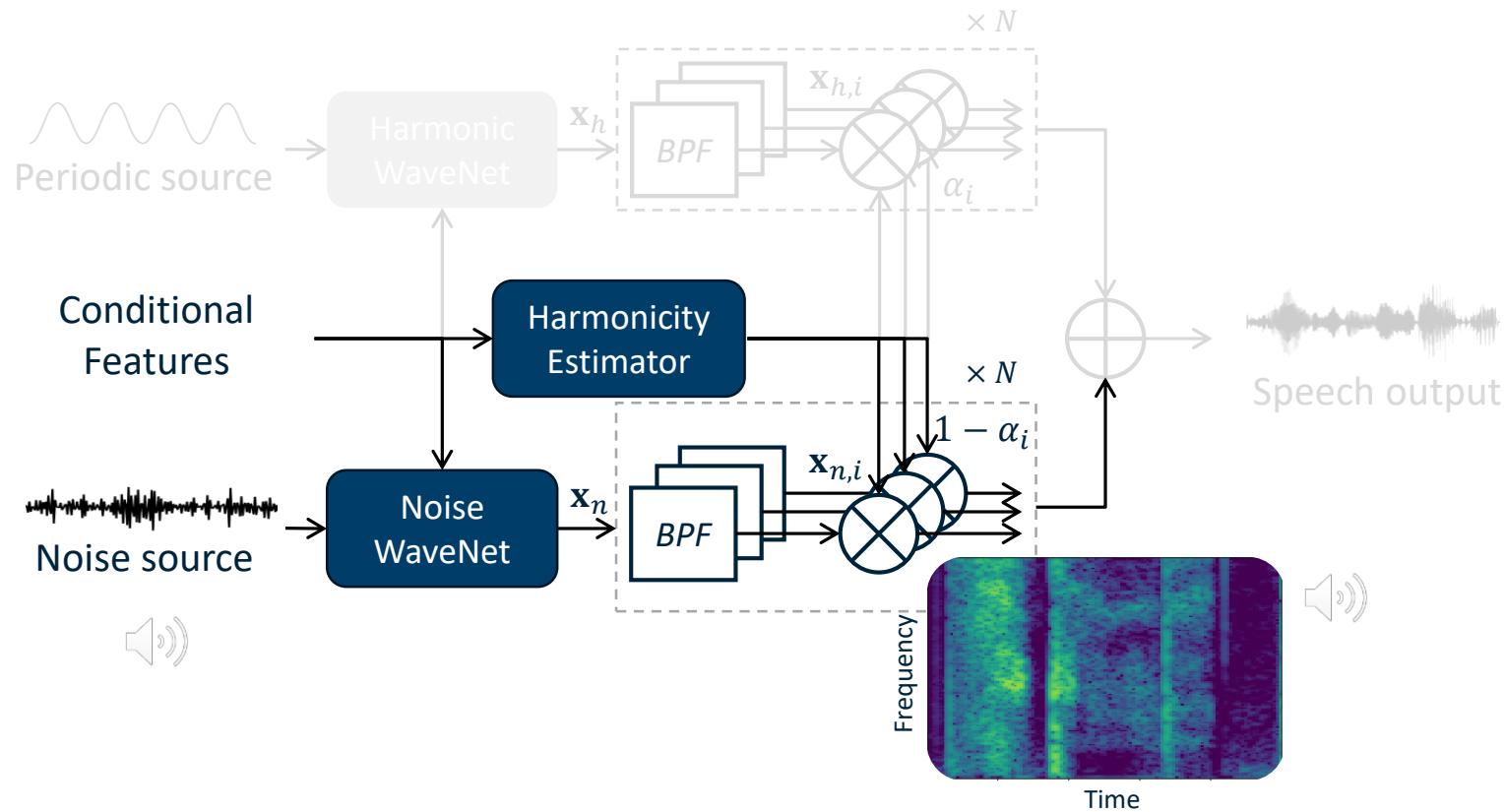
Harmonic/noise generators

Model architecture



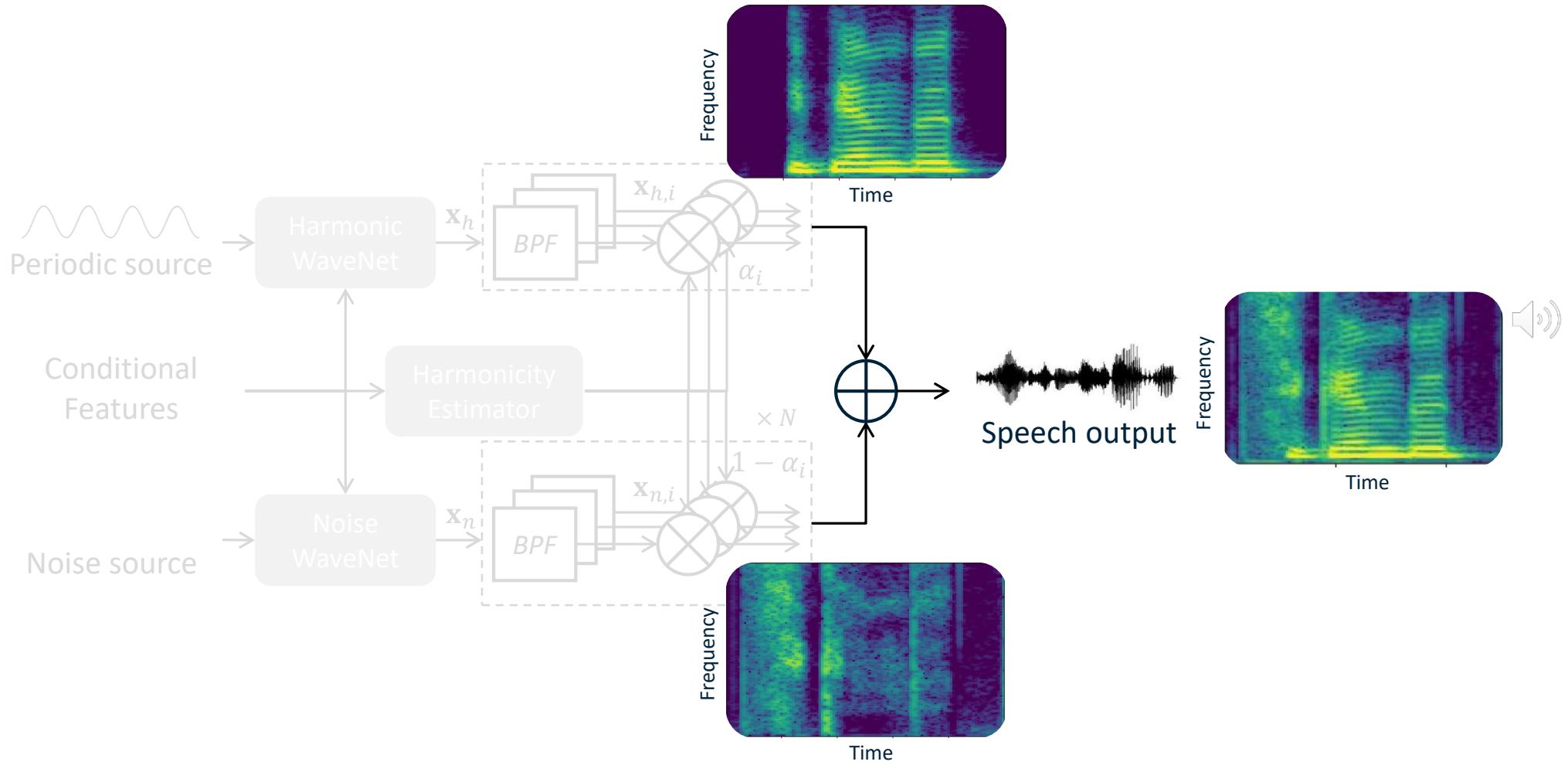
Harmonic/noise generators

Model architecture



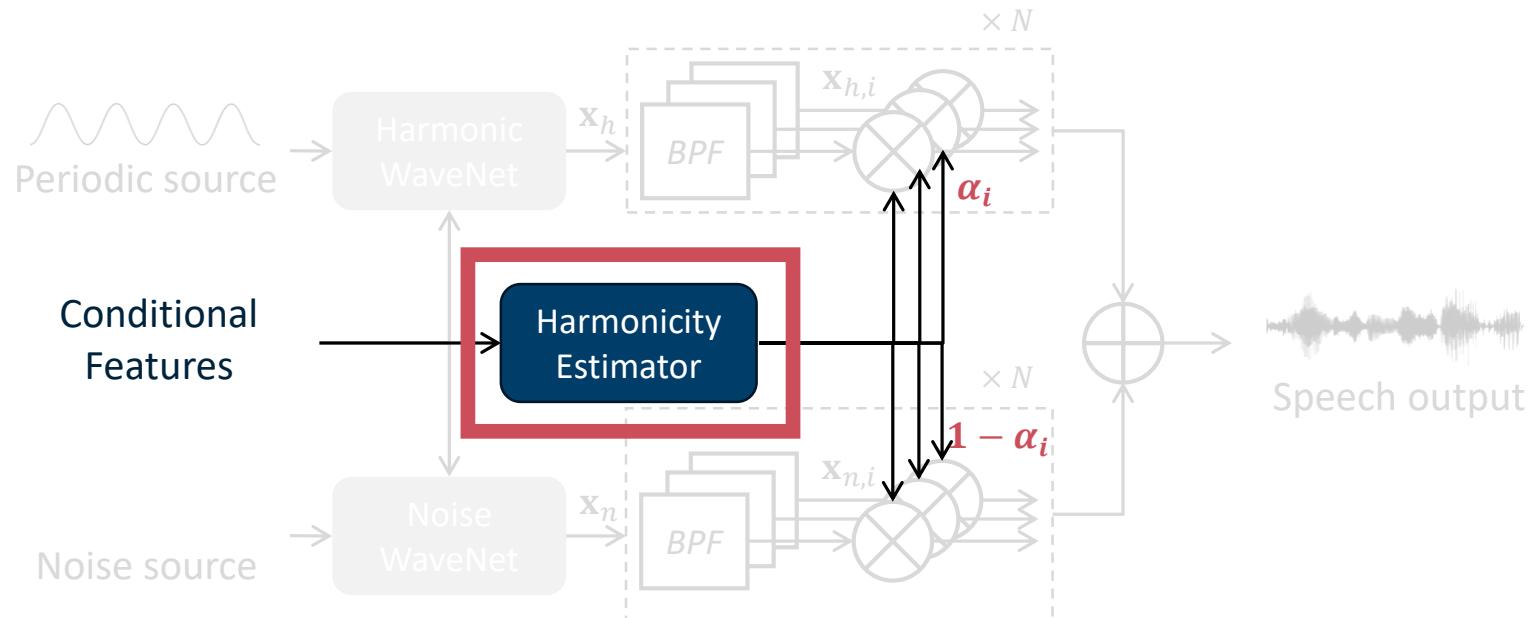
Harmonic/noise generators

Model architecture



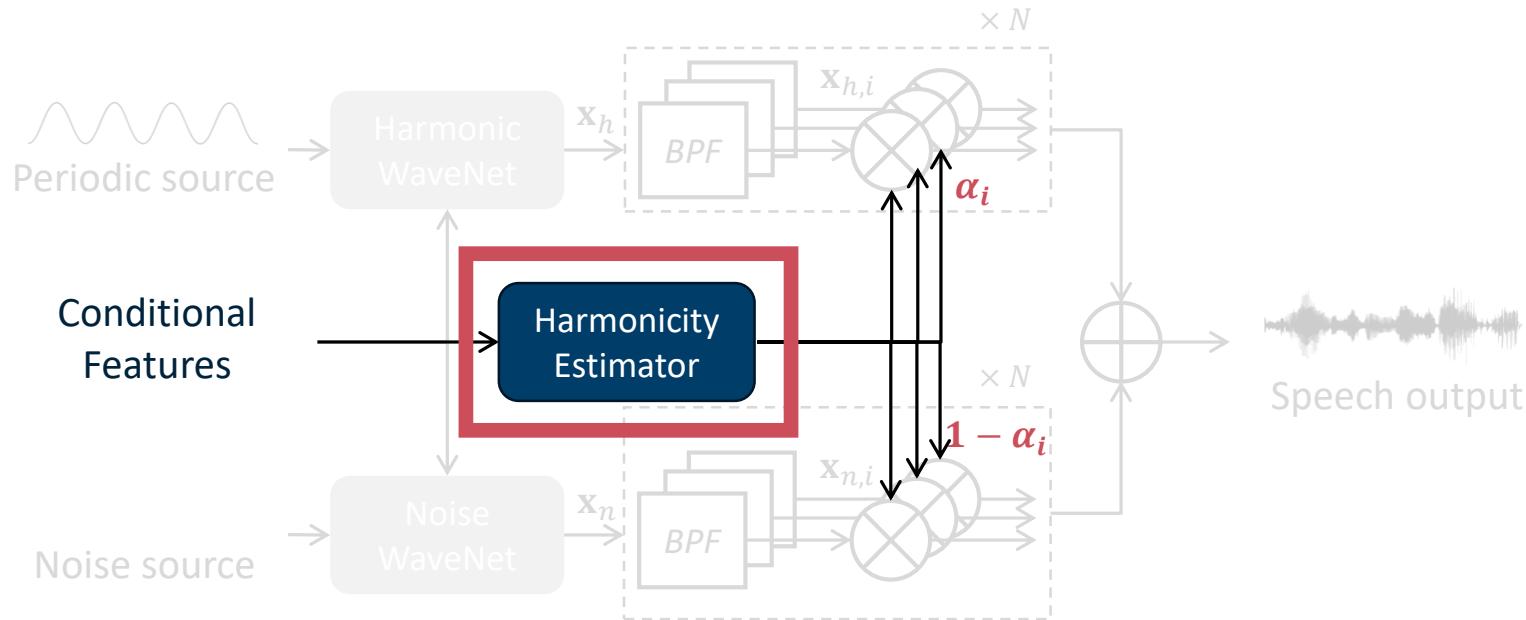
Harmonic/noise generators

Model architecture



Harmonic/noise generators

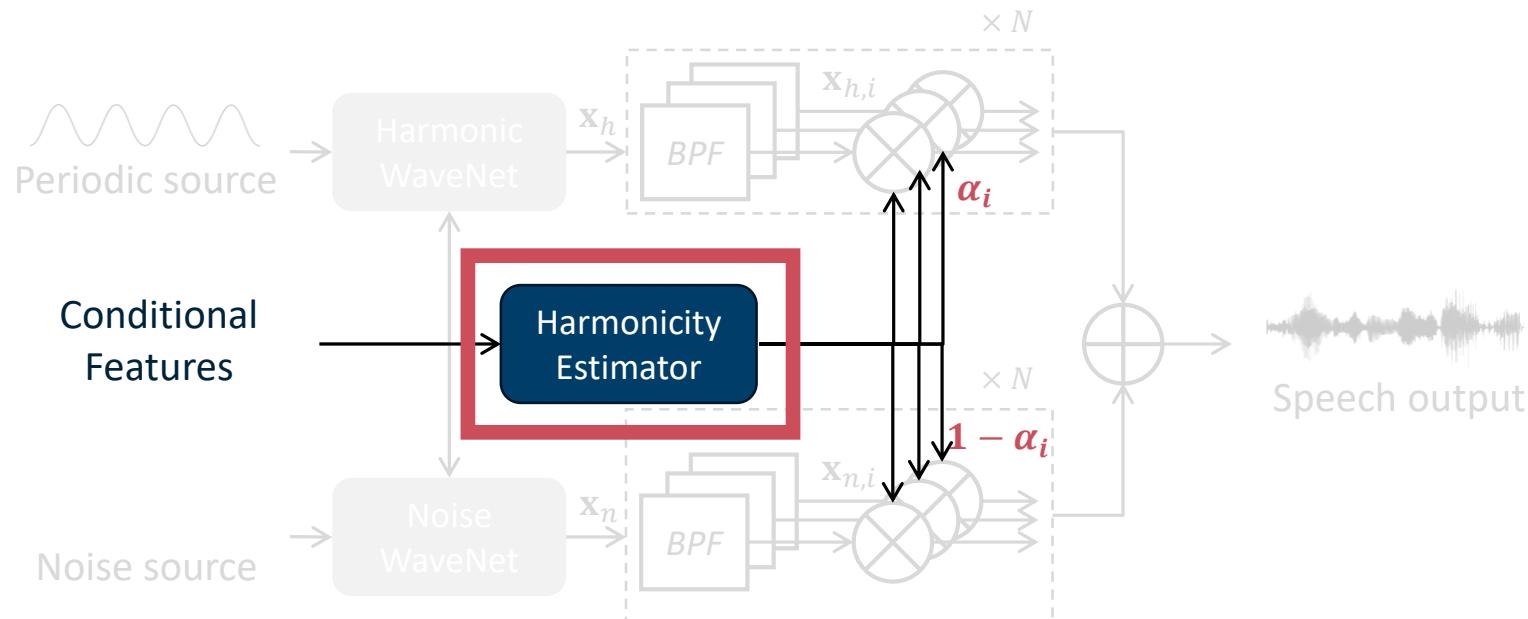
Model architecture



Parametric vocoders: Harmonicity has been estimated by **rule-based analysis** methods

Harmonic/noise generators

Model architecture



Parametric vocoders: Harmonicity has been estimated by rule-based analysis methods
Alternatively, we design **learnable harmonicities** optimized CNN blocks with input condition

Harmonic/noise generators

Evaluation results

Table 1. *The model size, inference speed, and MOS results with 95% confidence intervals: Acoustic features extracted from the recorded speech signal were used to compose the input acoustic features. The MOS results for highest score is in bold font.*

Label	Model	Use of HN model	Input signals for H-WaveNet	Type of HN model	Model size (M)	Inference speed	MOS
S1	WaveNet [21]	—	—	—	3.81	0.34×10^{-2}	4.22 ± 0.12
S2	PWG [7]	—	—	—	0.94	50.38	3.46 ± 0.37
S3	HN-PWG w/o noise [16]	Yes	Sine + V/UV	Full-band	0.94	47.91	4.02 ± 0.14
S4	HN-PWG	Yes	Sine + noise + V/UV	Full-band	0.94	47.93	4.18 ± 0.15
S5	Multi-band HN-PWG	Yes	Sine + noise + V/UV	Multi-band	0.99	47.87	4.29 ± 0.12
S6	Recordings	—	—	—	—	—	4.41 ± 0.12

S_i: i^{th} system; HN: harmonic-plus-noise; PWG: Parallel WaveGAN; H-WaveNet: harmonic WaveNet; V/UV: voicing flags upsampled from frame-level to sample-level. Note that inference speed, k , indicates that a system was able to generate waveforms k times faster than real-time. This evaluation was conducted on a server with a single NVIDIA Tesla V100 GPU.

Table 2. *Subjective MOS test results with 95% confidence intervals for the TTS systems with respect to the different vocoding models. The MOS results for highest score is in bold font.*

Label	Model	MOS
S-T1	WaveNet [21]	4.03 ± 0.19
S-T2	PWG [7]	3.56 ± 0.28
S-T3	HN-PWG w/o noise	2.60 ± 0.22
S-T4	HN-PWG	4.01 ± 0.17
S-T5	Multi-band HN-PWG	4.03 ± 0.16
S6	Recordings	4.41 ± 0.12

S-T_i: i^{th} system that generates speech waveform from the acoustic features predicted by TTS model.

Acoustic model: Tacotron 2

Parallel waveform synthesis

Summary

Summary

PARALLEL WAVEGAN: A FAST WAVEFORM GENERATION MODEL BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH MULTI-RESOLUTION SPECTROGRAM

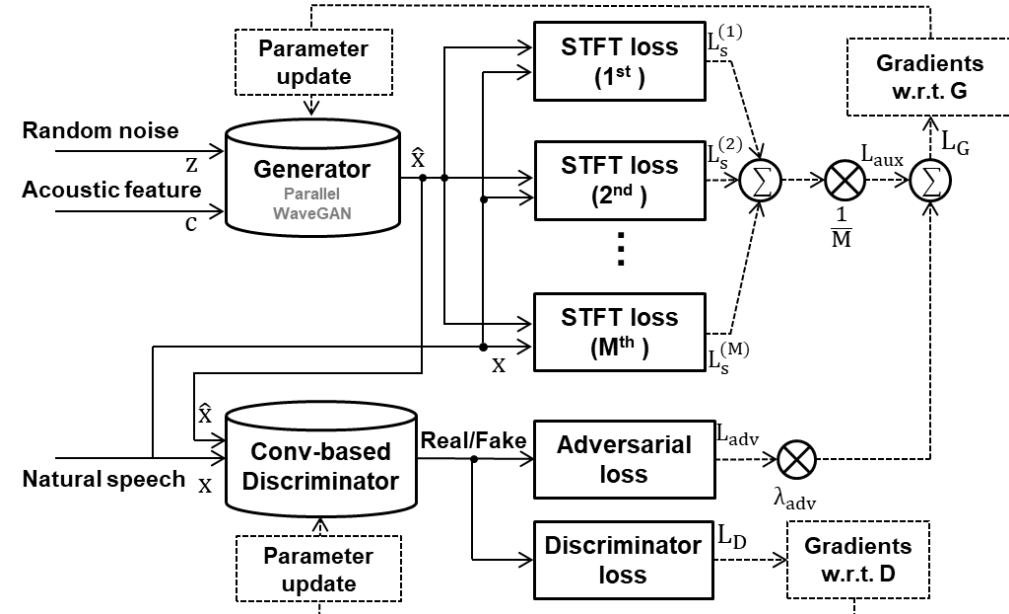
Ryuichi Yamamoto¹, Eunwoo Song² and Jae-Min Kim²

¹LINE Corp., Tokyo, Japan.

²NAVER Corp., Seongnam, Korea

ABSTRACT

We propose Parallel WaveGAN, a distillation-free, fast, and small-footprint waveform generation method using a generative adversarial network. In the proposed method, a non-autoregressive WaveNet is trained by jointly optimizing multi-resolution spectrogram and adversarial loss functions, which can effectively capture the time-frequency distribution of the realistic speech waveform. As our method does not require density distillation used in the conventional teacher-student framework, the entire model can be easily trained. Furthermore, our model is able to generate high-fidelity speech even with its compact architecture. In particular, the proposed Parallel WaveGAN has only 1.44 M parameters and can generate 24 kHz speech waveform 28.68 times faster than real-time on a single GPU environment. Perceptual listening test results verify that our proposed method achieves 4.16 mean opinion score within a Transformer-based text-to-speech framework, which is comparative to the best distillation-based Parallel WaveNet system.



Summary

IMPROVED PARALLEL WAVEGAN VOCODER WITH PERCEPTUALLY WEIGHTED SPECTROGRAM LOSS

Eunwoo Song¹, Ryuichi Yamamoto², Min-Jae Hwang³, Jin-Seob Kim¹, Ohsung Kwon¹, Jae-Min Kim¹

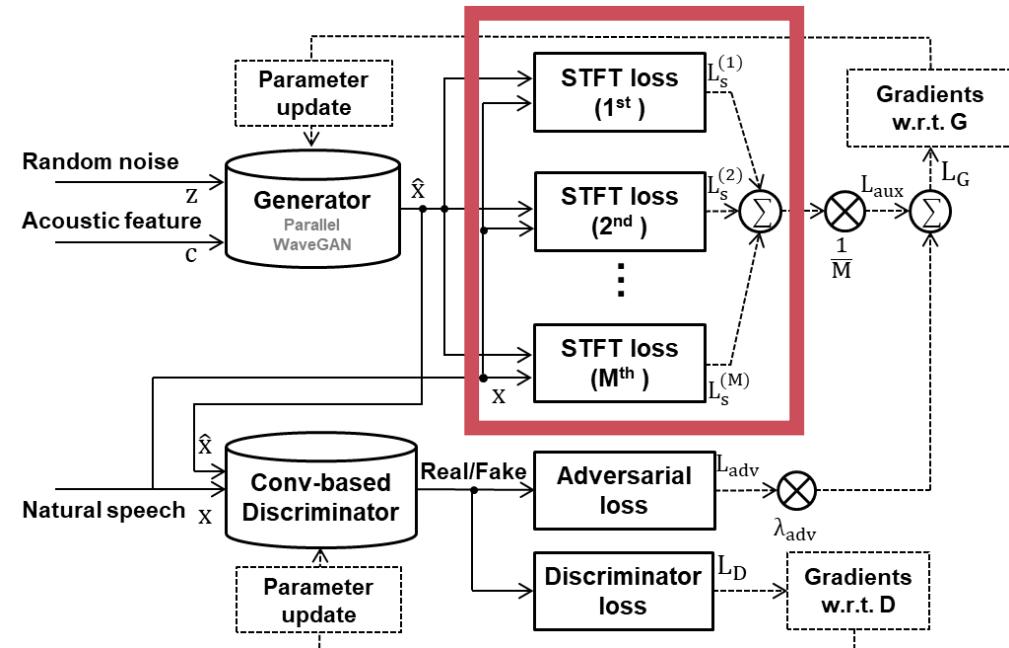
¹NAVER Corp., Seongnam, Korea

²LINE Corp., Tokyo, Japan

³Search Solutions Inc., Seongnam, Korea

ABSTRACT

This paper proposes a spectral-domain perceptual weighting technique for Parallel WaveGAN-based text-to-speech (TTS) systems. The recently proposed Parallel WaveGAN vocoder successfully generates waveform sequences using a fast non-autoregressive WaveNet model. By employing multi-resolution short-time Fourier transform (MR-STFT) criteria with a generative adversarial network, the light-weight convolutional networks can be effectively trained without any distillation process. To further improve the vocoding performance, we propose the application of frequency-dependent weighting to the MR-STFT loss function. The proposed method penalizes perceptually-sensitive errors in the frequency domain; thus, the model is optimized toward reducing auditory noise in the synthesized speech. Subjective listening test results demonstrate that our proposed method achieves 4.21 and 4.26 TTS mean opinion scores for female and male Korean speakers, respectively.



Summary

PARALLEL WAVEFORM SYNTHESIS BASED ON GENERATIVE ADVERSARIAL NETWORKS WITH VOICING-AWARE CONDITIONAL DISCRIMINATORS

Ryuichi Yamamoto¹, Eunwoo Song², Min-Jae Hwang³ and Jae-Min Kim²

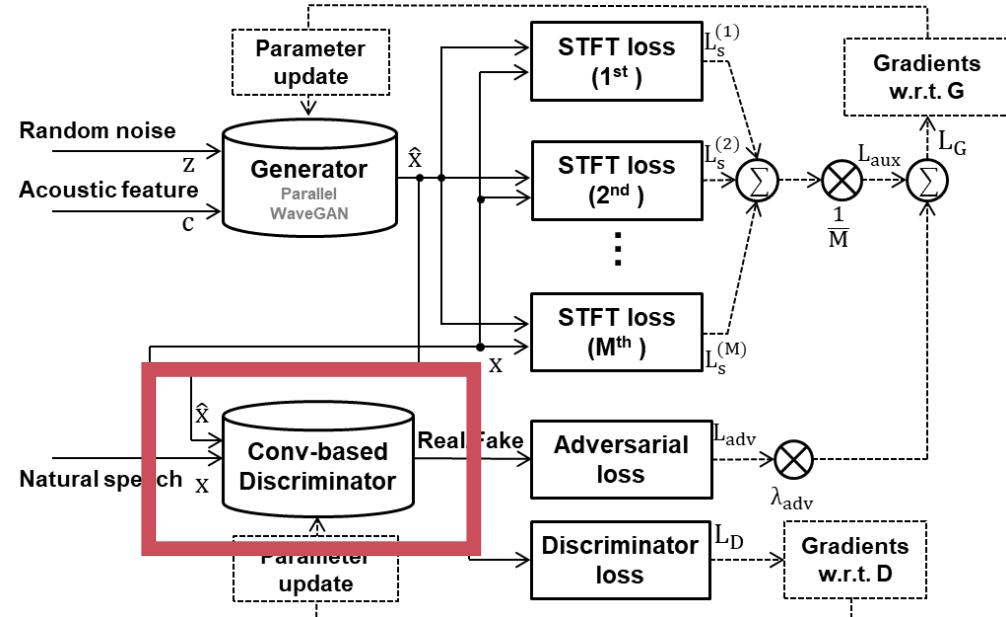
¹LINE Corp., Tokyo, Japan

²NAVER Corp., Seongnam, Korea

³Search Solutions Inc., Seongnam, Korea

ABSTRACT

This paper proposes voicing-aware conditional discriminators for Parallel WaveGAN-based waveform synthesis systems. In this framework, we adopt a projection-based conditioning method that can significantly improve the discriminator's performance. Furthermore, the conventional discriminator is separated into two waveform discriminators for modeling voiced and unvoiced speech. As each discriminator learns the distinctive characteristics of the harmonic and noise components, respectively, the adversarial training process becomes more efficient, allowing the generator to produce more realistic speech waveforms. Subjective test results demonstrate the superiority of the proposed method over the conventional Parallel WaveGAN and WaveNet systems. In particular, our speaker-independently trained model within a FastSpeech 2 based text-to-speech framework achieves the mean opinion scores of 4.20, 4.18, 4.21, and 4.31 for four Japanese speakers, respectively.



Summary

High-fidelity Parallel WaveGAN with Multi-band Harmonic-plus-Noise Model

Min-Jae Hwang^{1*}, Ryuichi Yamamoto^{2*}, Eunwoo Song³ and Jae-Min Kim³

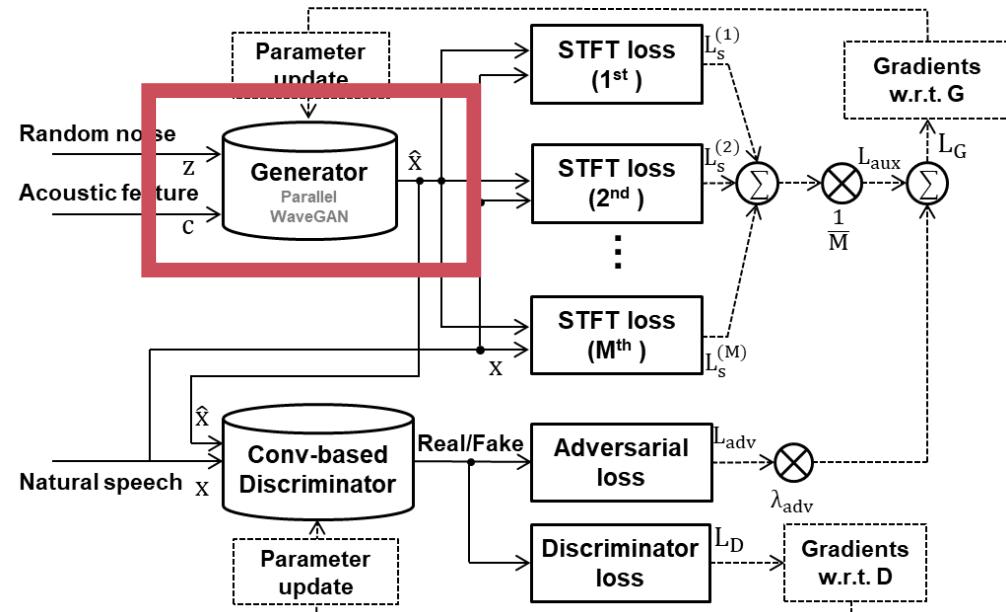
¹Search Solutions Inc., Seongnam, Korea

²LINE Corp., Tokyo, Japan

³NAVER Corp., Seongnam, Korea

Abstract

This paper proposes a multi-band harmonic-plus-noise (HN) Parallel WaveGAN (PWG) vocoder. To generate a high-fidelity speech signal, it is important to well-reflect the harmonic-noise characteristics of the speech waveform in the time-frequency domain. However, it is difficult for the conventional PWG model to accurately match this condition, as its single generator inefficiently represents the complicated nature of harmonic-noise structures. In the proposed method, the HN WaveNet models are employed to overcome this limitation, which enable the separate generation of the harmonic and noise components of speech signals from the pitch-dependent sine wave and Gaussian noise sources, respectively. Then, the energy ratios between harmonic and noise components in multiple frequency bands (i.e., subband harmonics) are predicted by an additional harmonicity estimator. Weighted by the estimated harmonics, the gain of harmonic and noise components in each subband is adjusted, and finally mixed together to compose the full-band speech signal. Subjective evaluation results showed that the proposed method significantly improved the perceptual quality of the synthesized speech.



Q / A



gregorio.song@gmail.com

