



G. Richard

Audio and cross-modal Generative AI

Gaël RICHARD

Professor, Telecom Paris, Institut polytechnique de Paris


Winter School on Generative AI – Feb 27th-28th, 2024, EMINES conference Center, Morocco




Content

- **A few examples of audio generative AI**
- **What is an audio signal ?**
- **Generative audio**
 - A short (and incomplete) historical perspective
 - Deep neural audio synthesis
 - *Autoregressive, VAE, VQ-VAE*
 - *Neural discrete representation (or tokenisation)*
 - *GANs, Diffusion models*
- **Cross modal audio generation : some examples**
- **Towards Hybrid Deep learning**
- **Conclusion**







A few examples of Audio Generative AI

Speech synthesis

- Several systems: Vall-E, VoiceBox (Meta), OpenAI, ...
- Vall-E( Microsoft)
 - Zero-shot TTS

Text	Speaker prompt	Ground Truth	VALL-E
They moved thereafter cautiously about the hut groping before and about them to find something to show that Warrenton had fulfilled his mission			

- ... or keeping the speaker emotion



Text	Emotion	Speaker prompt	VALL-E
We have to reduce the number of plastic bags.	Anger		
	Sleepy		
	Amused		



A few examples of Audio Generative AI

Speech synthesis

- « Deepfake » voices by many actors : Resemble.ai, Speechify, Respeecher,...
- An example on my voice with Speechify (*OpenAI's API*)

Text	Training prompt (in French)	Generated voice
Hi, Gaël Richard! It's time to listen to your voice clone in action. Your voice clone opens up a world of possibilities.		



A few examples of Audio Generative AI

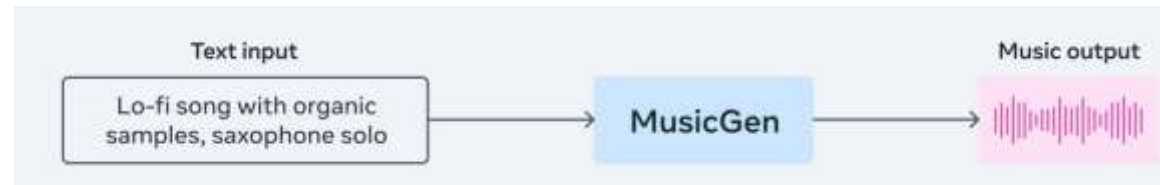
Audio/Music synthesis

- Several impressive models:



OpenAI (Jukebox, MuseNet,) (unseen lyrics rendition, completion, ..)

- MusicGen/AudioGen (AudioCraft, Meta): text-to-music or text-to audio generation



An example with MusicLM

Text	Generated music
Slow tempo, bass-and-drums-led reggae song. Sustained electric guitar. High-pitched bongos with ringing tones. Vocals are relaxed with a laid-back feel, very expressive.	



A few examples of Audio Generative AI

Cross modal audio synthesis

- Examples with MusicLM (text+ melody conditioning) generation,
- *Painting Caption Conditioning: An example with MusicLM*

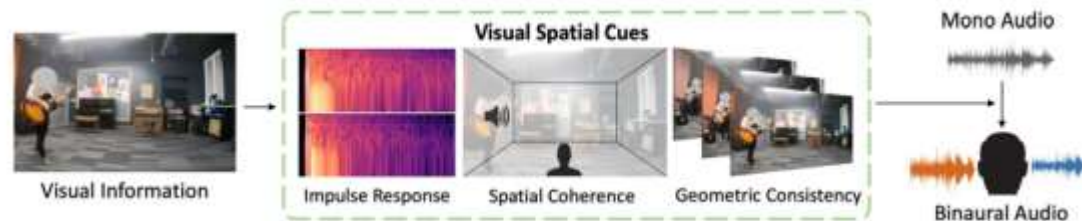
Guernica -
Pablo Picasso



"The grey, black, and white painting, on a canvas 3.49 meters tall and 7.76 meters across, portrays the suffering wrought by violence and chaos. Prominent in the composition are a gored horse, a bull, screaming women, a dead baby, a dismembered soldier, and flames." By [wikipedia](https://en.wikipedia.org/wiki/Guernica)



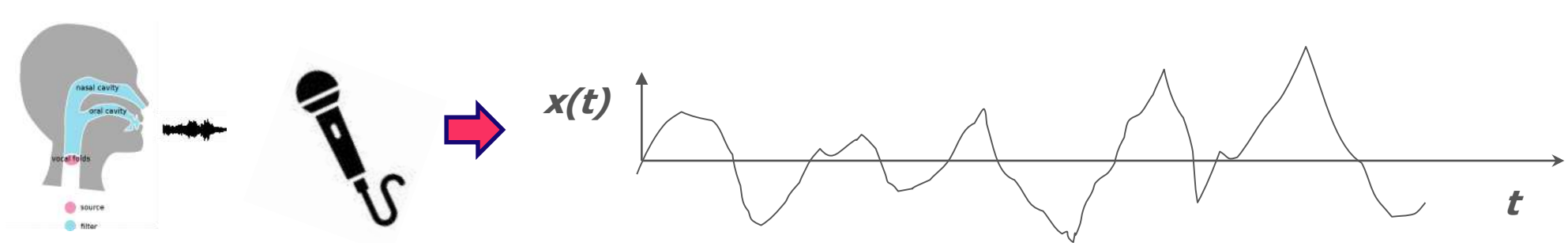
- Other examples: visually guided audio spatialization + the sound of pixel (2018)



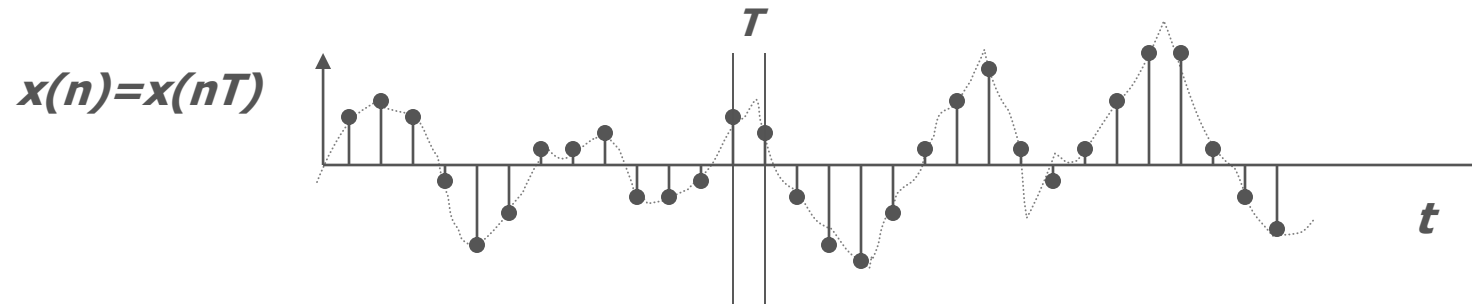
... but what is an audio signal ?

What is an audio signal

- The audio signal $x(t)$ is an continuous acoustic signal



- Let $x(nT)$ be the discrete signal sampled at time $t=nT$



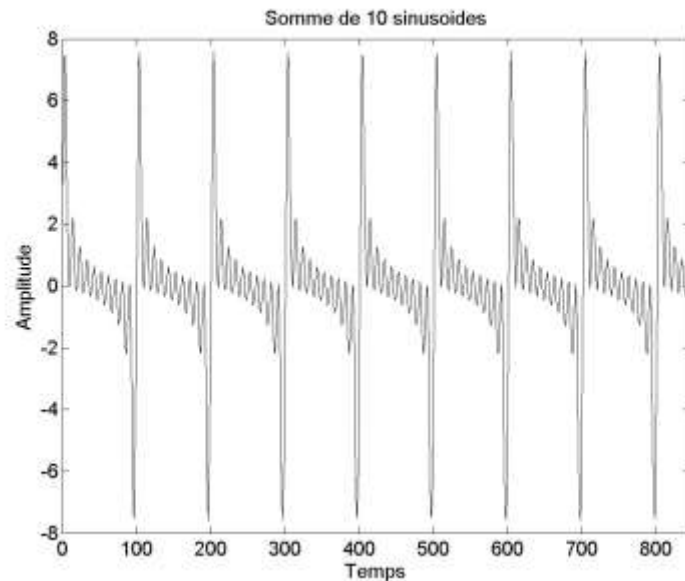
Time-Frequency representation

- Fourier Transform

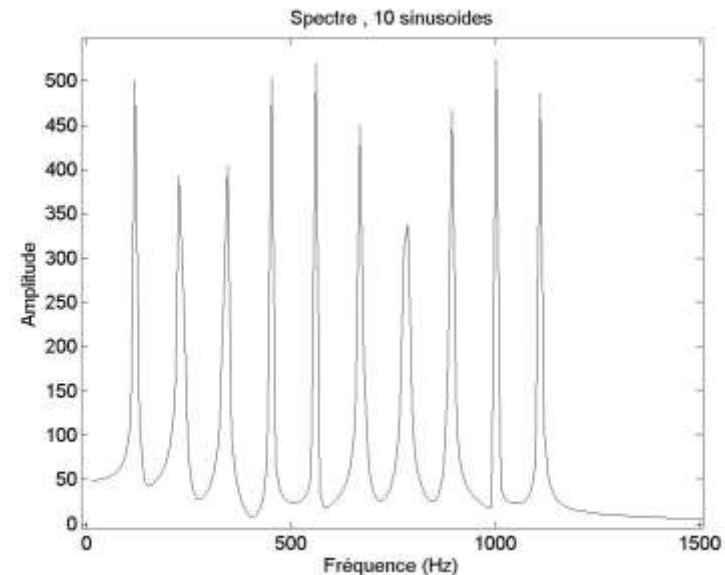
$$X_k = \sum_{n=0}^{N-1} x_n e^{-2j\pi nk/N}$$

$$x_n = \frac{1}{N} \sum_{k=0}^{N-1} X_k e^{2j\pi nk/N}$$

x_n

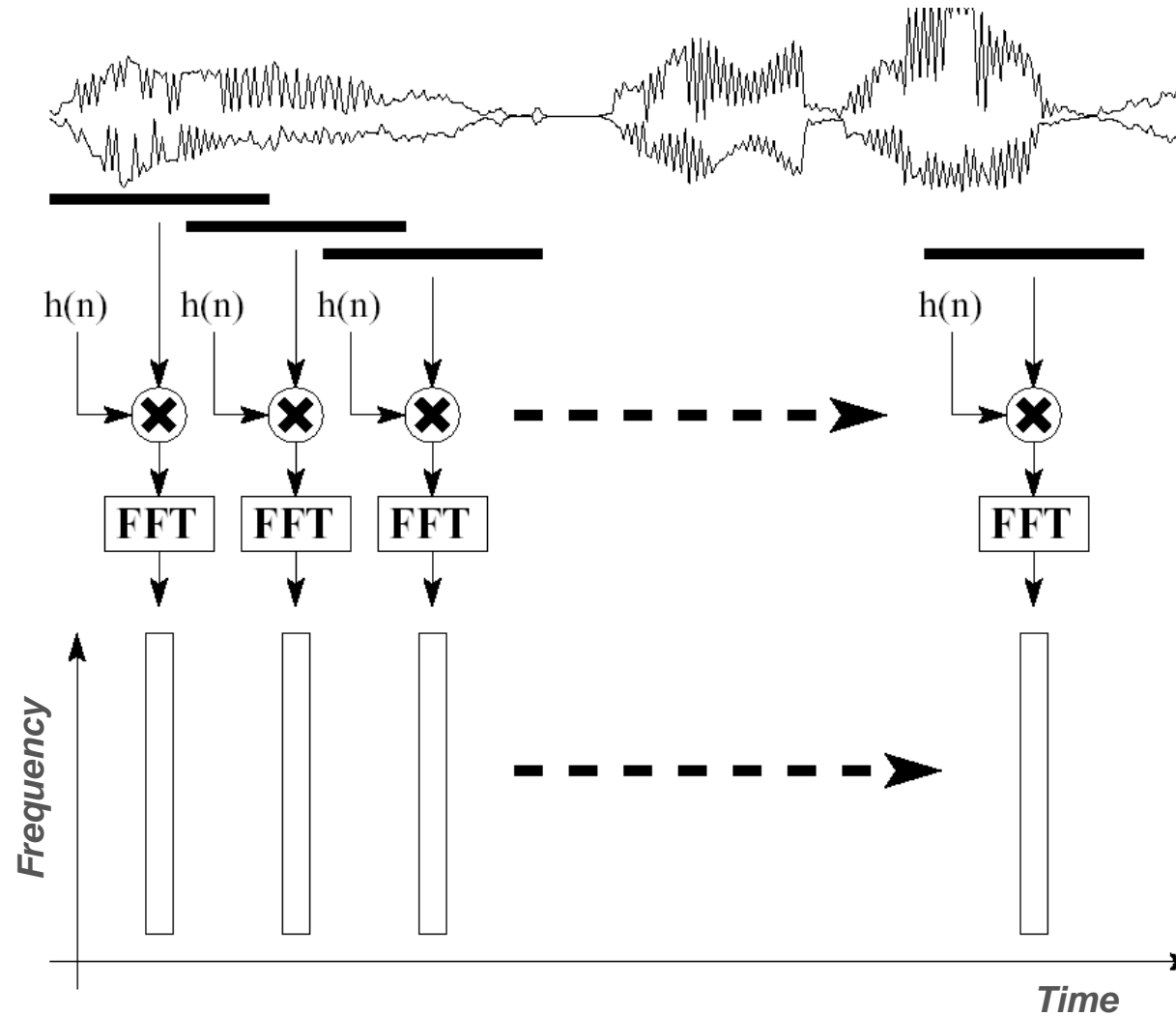


$|X_k|$



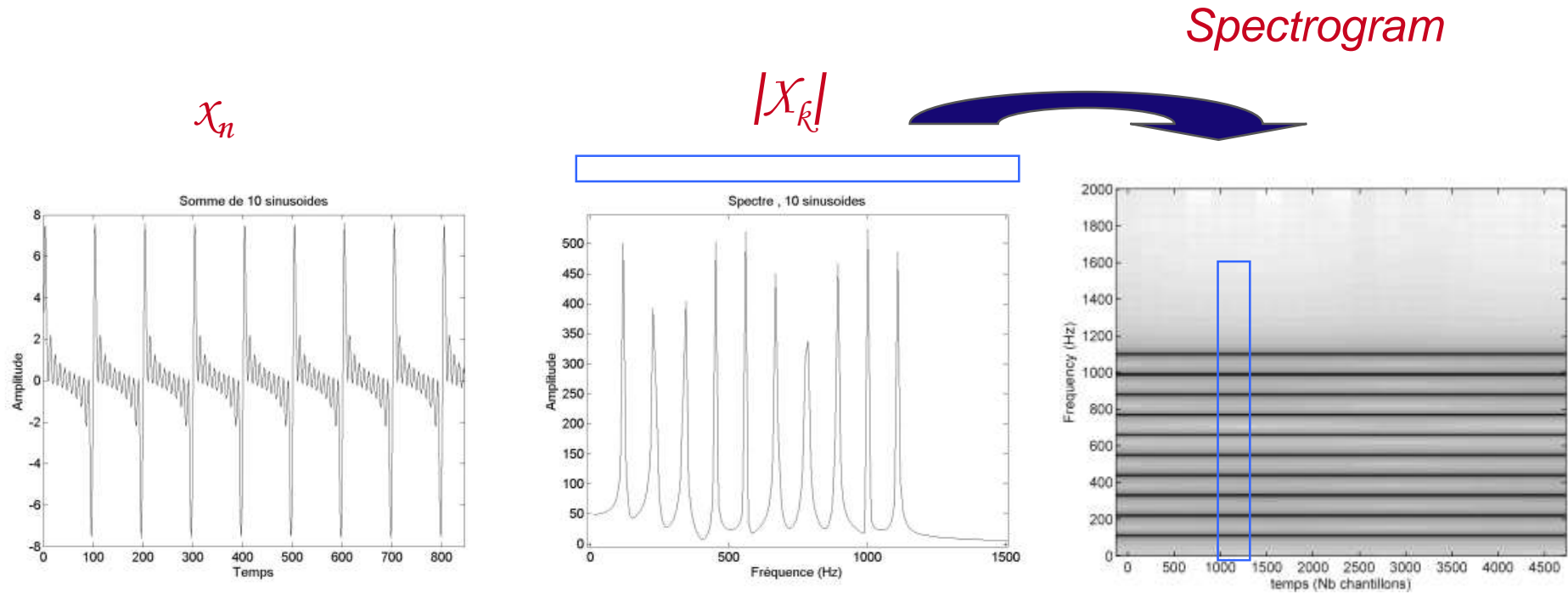
Spectral analysis of an audio signal (1)

(drawing from J. Laroche)



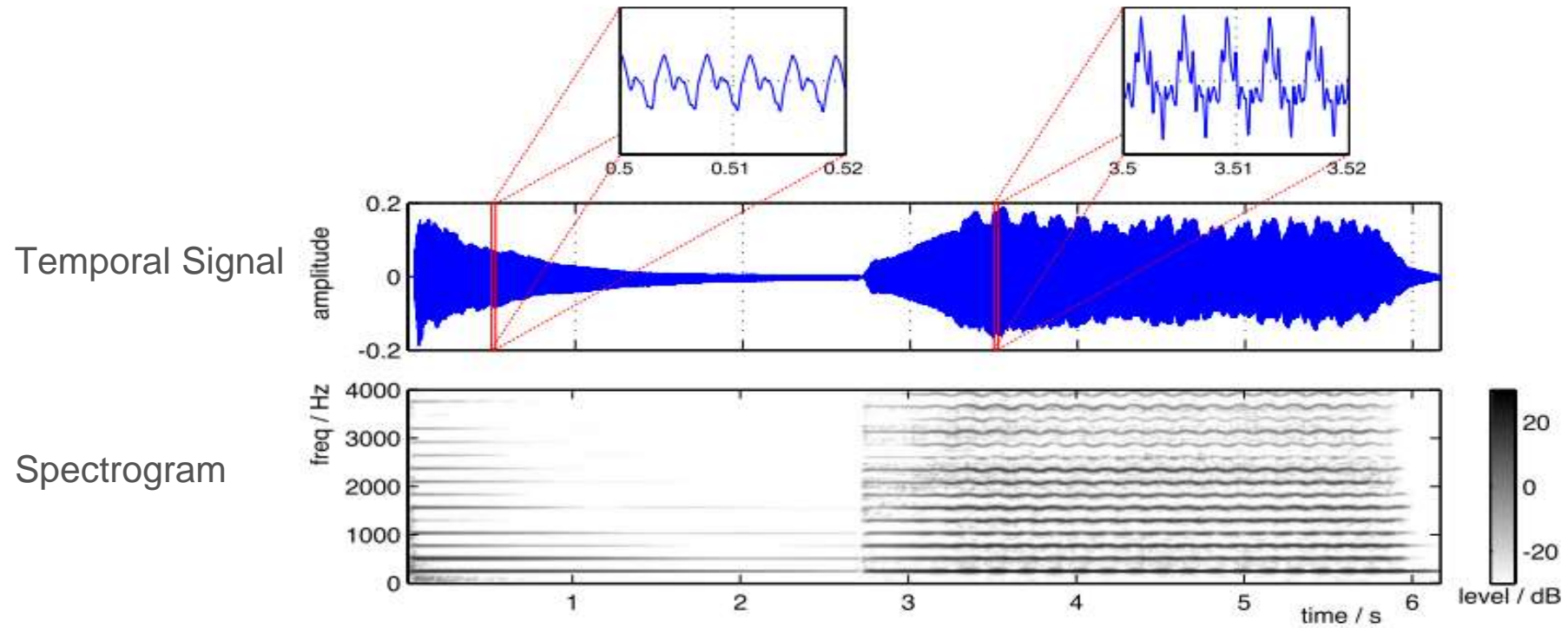
Spectral analysis of an audio signal (2)

- Spectrogram of a sum of 10 stable sinusoids



Audio signal representations

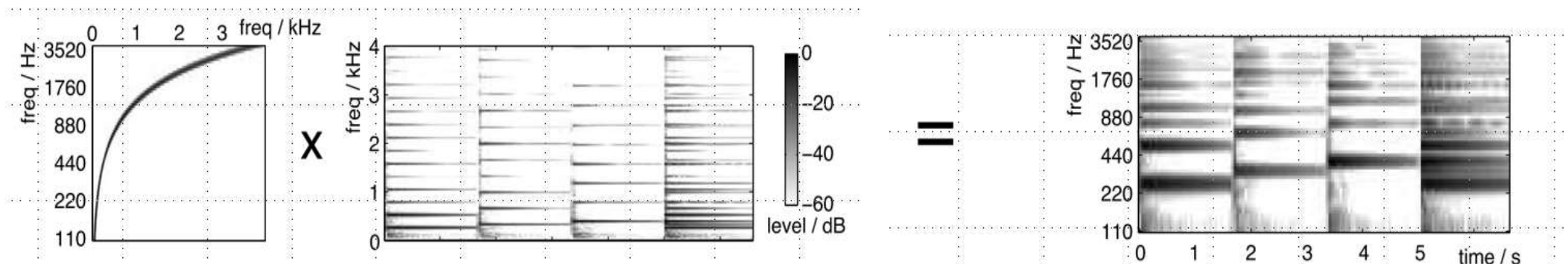
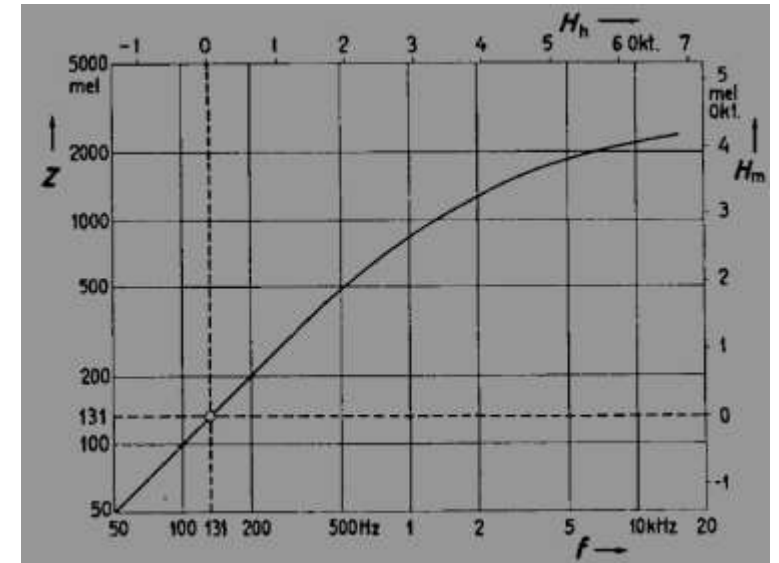
- Example on a music signal: note C (262 Hz) produced by a piano and a violin.



Towards a more specific representation

Mel-spectrogram

- Exploiting principles of sound perception
 - E.g. Tonal heights perception: Mel scale
 - From 0 à 500 Hz où 1 Mel = 1 Hz (linear)
 - Above 500 Hz, height perception (or « tonie ») grows logarithmically with frequency
- Example of analytical formula: $mel(f) = 1000 \log_2\left(1 + \frac{f}{1000}\right)$



About Generative audio ...

Generative audio ... an old domain

- ...generating speech with an instrument or a machine

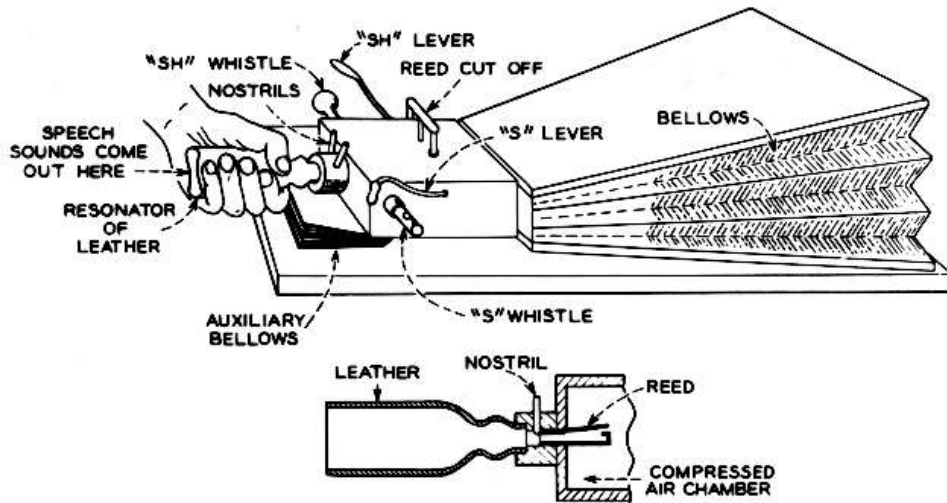
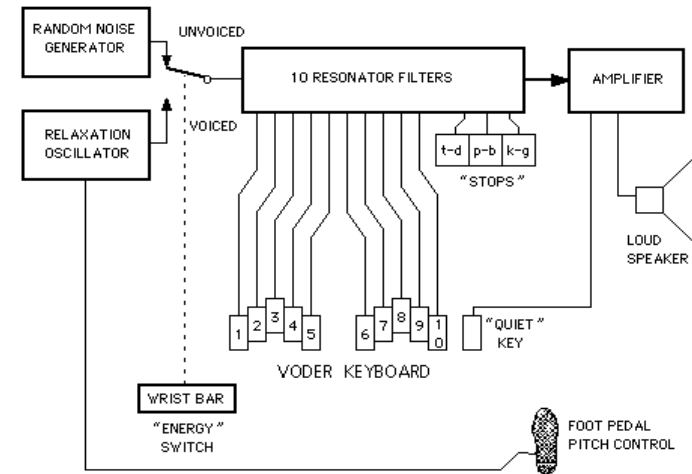


FIG. 10. Wheatstone's reconstruction of von Kempelen's speaking machine.¹

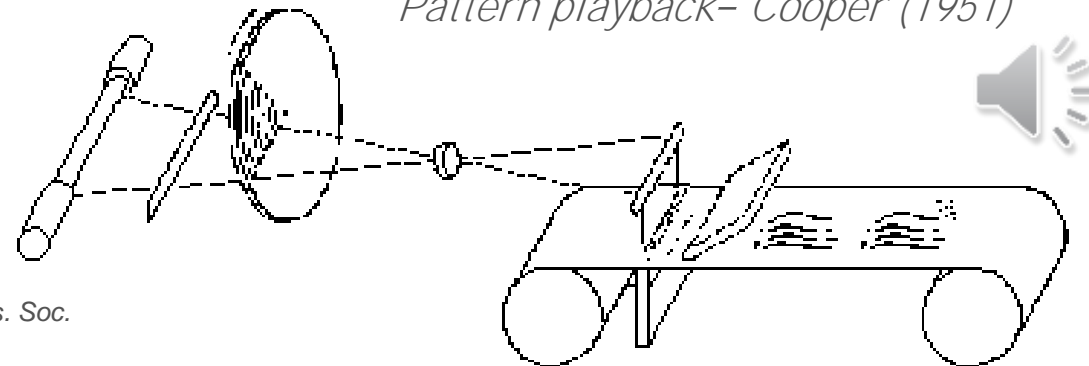
The Journal of the Acoustical Society of America

Van Kempelen machine (1791)

Voder - Dudley (1939)

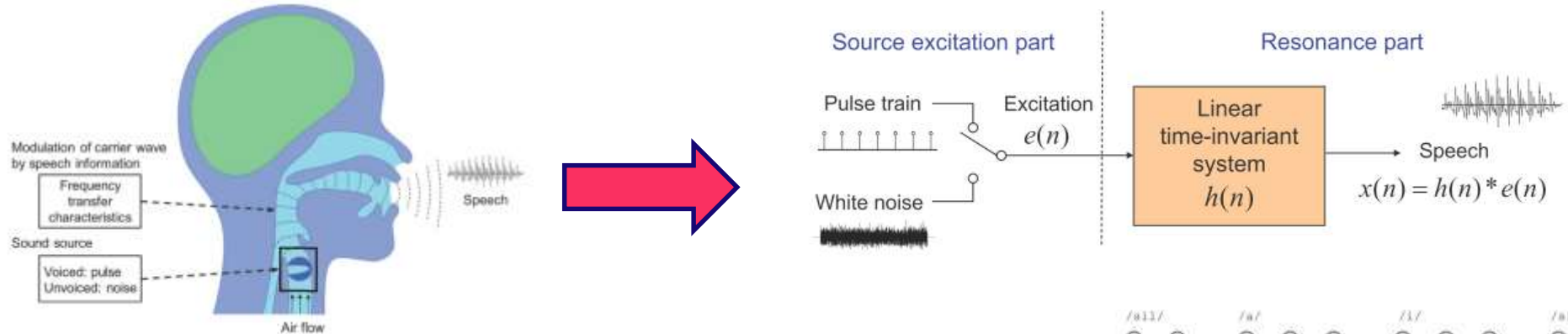


Pattern playback - Cooper (1951)



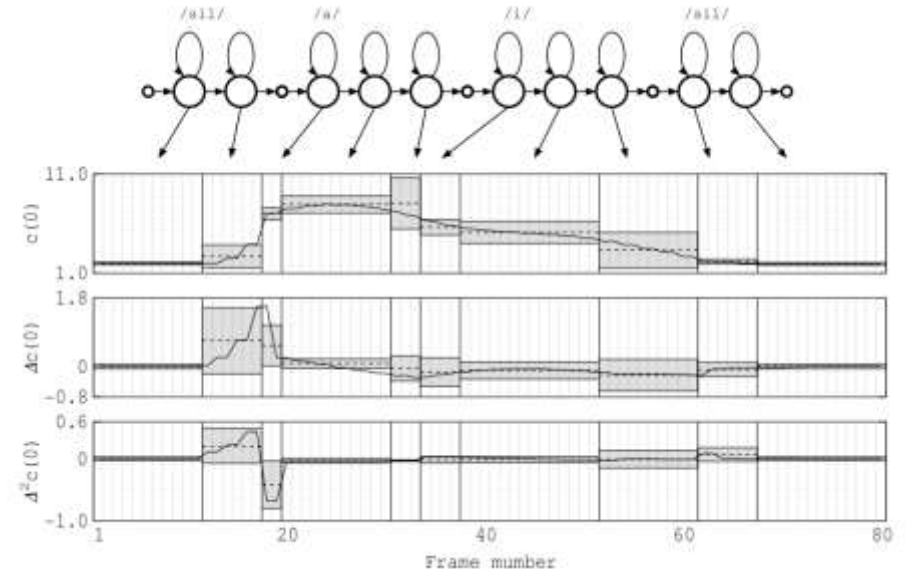
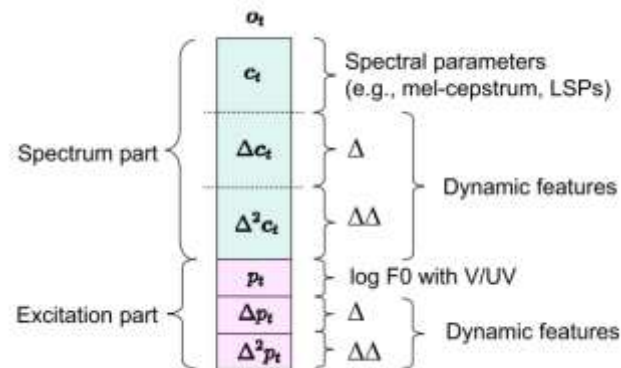
Generative audio ... an old domain

- ...generating speech with a simplified « speech production » model



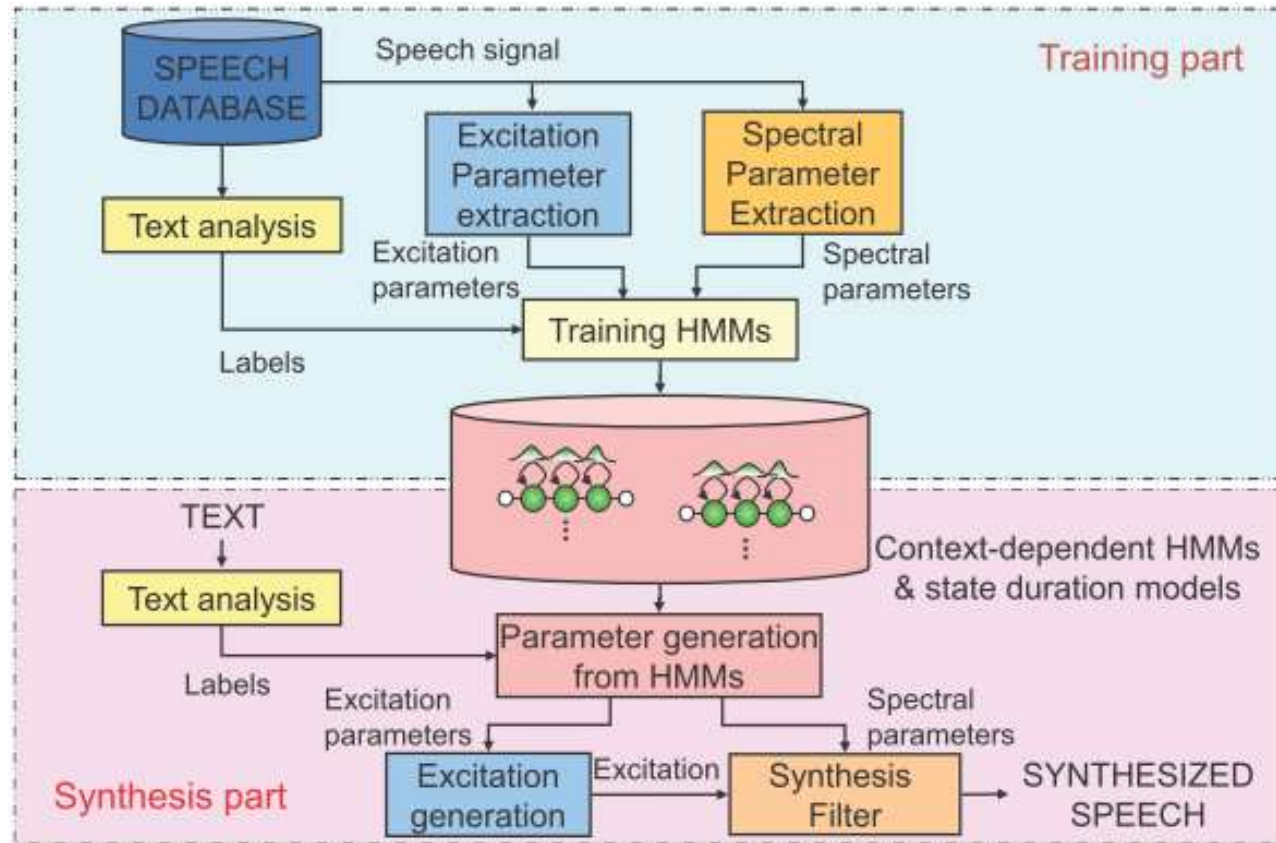
- ..Or using Hidden Markov Models (HMMs)

Observation vector



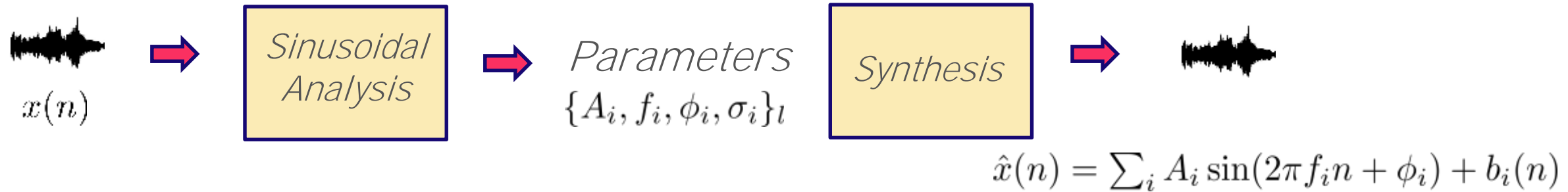
Generative audio ... an old domain

- ...generating speech with Hidden Markov Models (HMMs)







Generative audio ... an old domain

- ...generating and transforming sound using an analysis/synthesis model



⇒ Example on a piano signal

- Original signal: 
- Transposed by a third: 
- Signal S (« Sum of sinusoids with vibrato effect »): 
- Signal N (Noise): 

An « image of audio » (e.g spectrogram) is not the same as a natural image

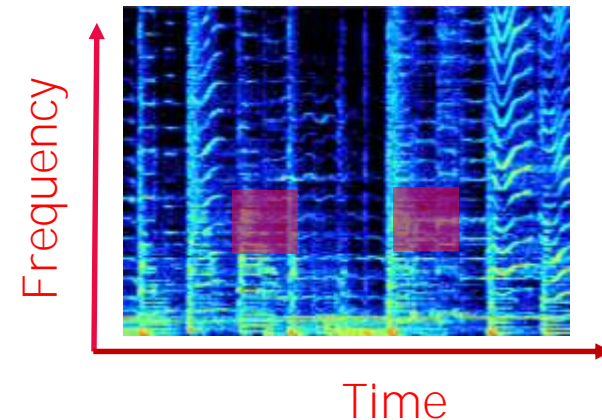
➤ Natural images

- the axes x and y represent the same concept (*spatial position*)
- the elements of an image have the same meaning independently of their positions over x and y.
- neighboring pixels:
 - usually highly correlated,
 - often belong to the same object



➤ Time-frequency audio representations (for example a spectrogram)

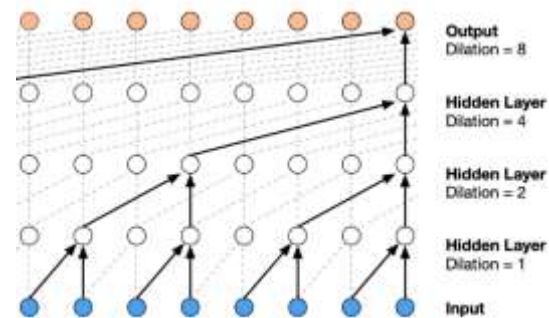
- the axes x and y represent profoundly different concepts (time and frequency).
- the elements of spectrogram (*such as the T/F area of a source*) have the same meaning independently of their position over time but not over frequency
- no invariance over y, even in the case of log-frequencies
- neighboring pixels:
 - are not necessarily correlated
 - a given sound source (such as an harmonic sound) can be distributed over the whole frequency in a sparse way (*the harmonics of a given sound can be spread over the whole frequency range*)



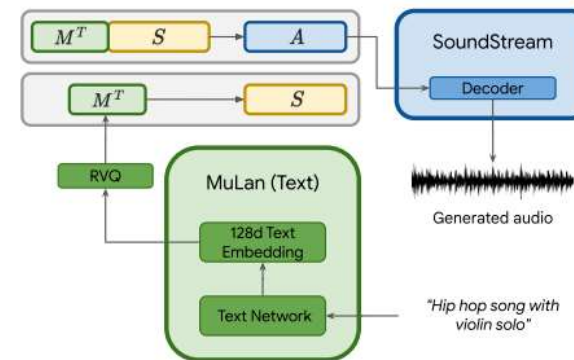
Deep neural audio synthesis

- Machine-learning based models "uses large amount of data and machine learning to generate sounds"
- A rapid growth and adoption of deep neural networks for audio synthesis

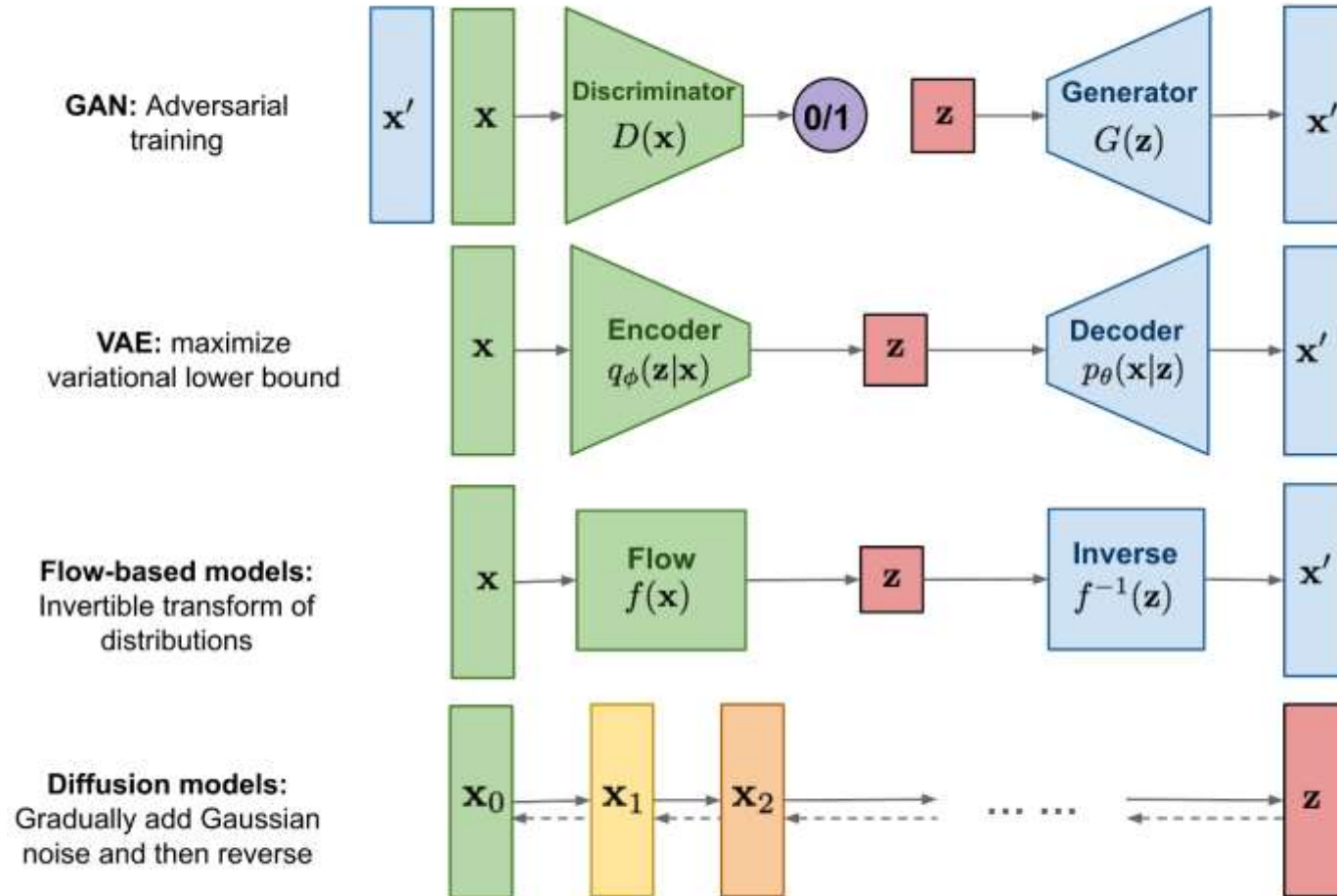
From wavenet (2016)
(autoregressive model)



MusicLM (2023)
(Generating Music from Text)



A large variety of generative models ...



Deep neural audio synthesis

Arch.	Name	Audio representation	Data	Conditioning
NAM	waveNet	van den Oord et al., 2016a	waveform	speech, piano speaker ID, text
	Universal music Translation	Mor et al., 2018	waveform	classical music -
	Hierarchical waveNet	Dieleman et al., 2018	waveform	piano music -
	SampleRNN	Mehri et al., 2017	waveform	speech, piano music -
	MelNet	Vasquez and Lewis, 2019	mag. spec.	speech, piano music speaker ID, text
	wavenetAE	Engel et al., 2017	waveform	tonal sounds pitch
	sparse Transformer	Child et al., 2019	waveform	piano music -
NFs	Parallel waveNet	van den Oord et al., 2018a	waveform	speech text, pitch
	ClariNet	Ping et al., 2018	waveform	speech text
	FlowwaveNet	Kim et al., 2018	waveform	speech text, Mel spec.
	waveGlow	Prenger et al., 2018	waveform	speech text, Mel spec.
	waveFlow	Ping et al., 2020	waveform	speech text, Mel spec.
	Blow	Serrà et al., 2019	waveform	speech speaker ID
VAEs	Planet Drums	Aouameur et al., 2019	Mel-scaled mag. spec.	drums instrument ID
	Jukebox	Dhariwal et al., 2020	waveform	music artist & genre ID, lyrics
	NOTONO	Bazin et al., 2020	mag. & IF	tonal instruments pitch
	FlowSynth	Esling et al., 2019	mag.	synth. sounds semantic tags
	Neural Granular Sound Synth.	Bitton et al., 2020	waveform	orchestral, drums, animals pitch, instrument ID
GANs	WaveGAN	Donahue et al., 2019	waveform	speech, drums, piano, birds -
	GANSynth	Engel et al., 2019	mag. & IF	tonal instruments pitch ID
	MelGAN	Kumar et al., 2019	mag. spec.	speech, music Mel-scaled spec. text
	GAN-TTS	Binkowski et al., 2020	waveform	speech pitch, text, speaker ID



J. Nistal, "Exploring Generative Adversarial Networks for Controllable Musical Audio Synthesis, PhD Thesis, IP Paris, 2022

Wavnet

a generative model, directly from the audio waveform

- The joint probability of a waveform $\mathbf{x} = \{x_1, \dots, x_T\}$

is factorised as a product of conditional probabilities :

$$p(\mathbf{x}) = \prod_{t=1}^T p(x_t | x_1, \dots, x_{t-1})$$

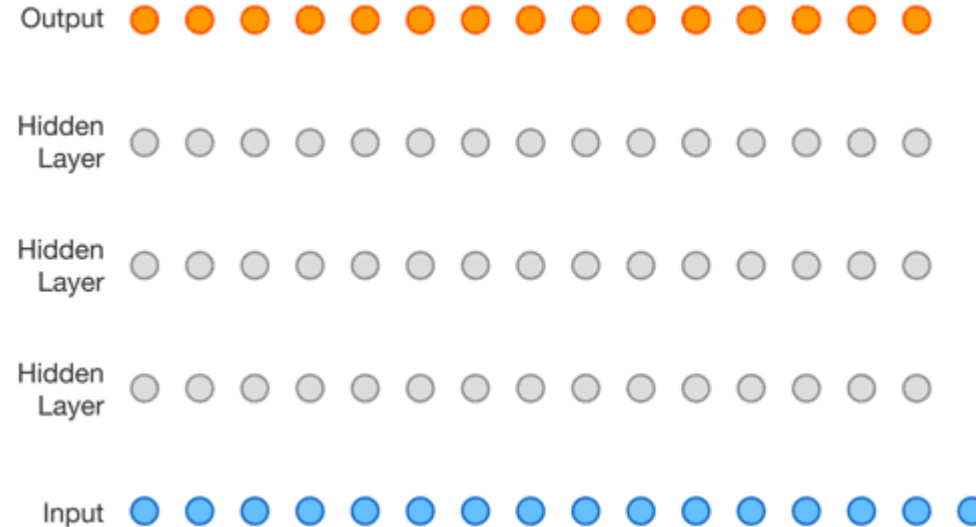
- the conditional probability distribution is modelled by a stack of convolutional layers;
- Output of the model: has the same time dimensionality as the input (no pooling)
- Output: a categorical distribution over the next value x_t with a softmax layer - optimized to maximize the log-likelihood of the data w.r.t. the parameters.



Wavnet

a generative model, directly from the audio waveform

- Dilated causal convolutions (the main ingredient!).
- Classic causal convolutions needs many layers to increase the receptive fields (RF)
- Dilated causal convolutions greatly increase RF



Wavnet

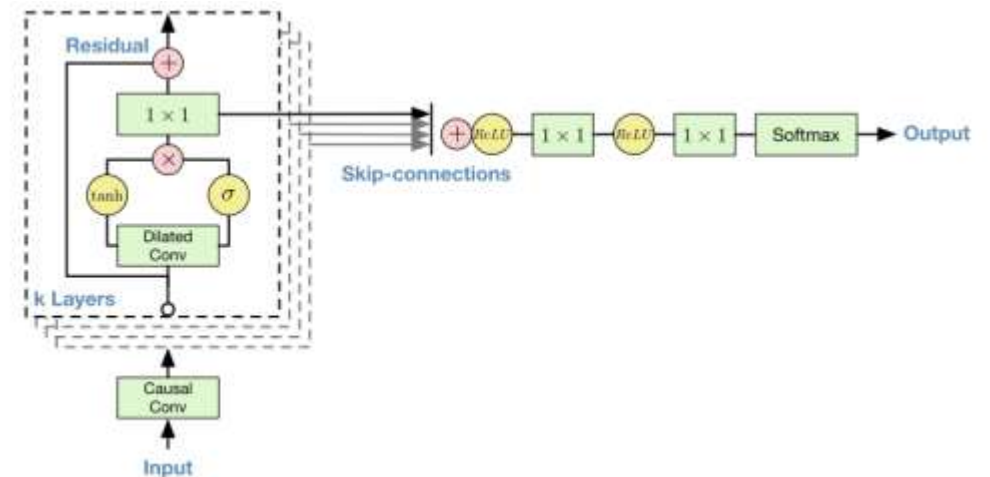
a generative model, directly from the audio waveform

- Condition distributions $p(x_t|x_1, \dots, x_{t-1})$ modelled using softmax distributions
- Use of mu-law to limit the number of “categories” (amplitude values):

$$f(x_t) = \text{sign}(x_t) \frac{\ln(1 + \mu |x_t|)}{\ln(1 + \mu)}$$

$$\mathbf{z} = \tanh(W_{f,k} * \mathbf{x}) \odot \sigma(W_{g,k} * \mathbf{x})$$

- Use of Gated recurrent units
- .. and residual and skip connections



Wavnet : sound examples

(from <https://www.deepmind.com/blog/wavenet-a-generative-model-for-raw-audio>)

- Speech ... but also music (no conditioning)



- But it is also possible to use conditions :

$$p(\mathbf{x} | \mathbf{h}) = \prod_{t=1}^T p(x_t | x_1, \dots, x_{t-1}, \mathbf{h})$$

$$\mathbf{z} = \tanh(W_{f,k} * \mathbf{x} + V_{f,k}^T \mathbf{h}) \odot \sigma(W_{g,k} * \mathbf{x} + V_{g,k}^T \mathbf{h})$$

- Speech (with condition on the text)



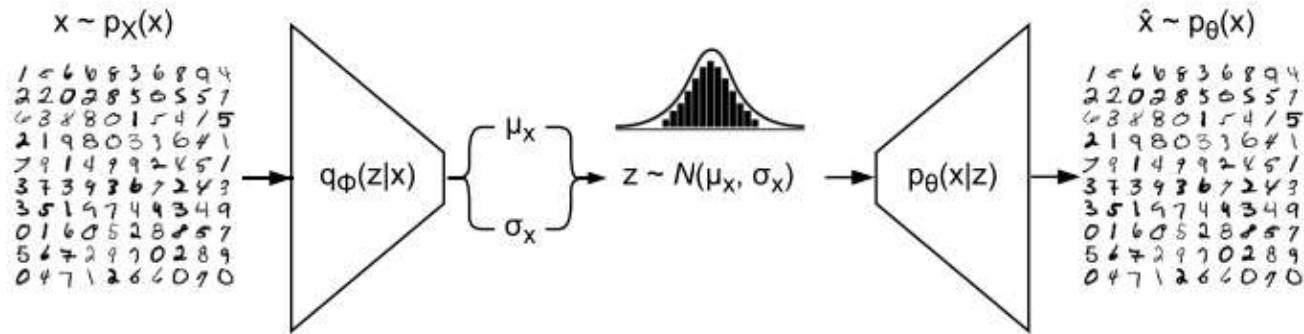
Wavnet and other neural autoregressive models

- Wavnet remains complex (sample is generated one at a time)
- Other neural autoregressive models

Arch.	Name	Audio representation	Data	Conditioning	
NAM	waveNet	van den Oord et al., 2016a	waveform	speech, piano	speaker ID, text
	Universal music Translation	Mor et al., 2018	waveform	classical music	-
	Hierarchical waveNet	Dieleman et al., 2018	waveform	piano music	-
	SampleRNN	Mehri et al., 2017	waveform	speech, piano music	-
	MelNet	Vasquez and Lewis, 2019	mag. spec.	speech, piano music	speaker ID, text
	wavenetAE	Engel et al., 2017	waveform	tonal sounds	pitch
	sparse Transformer	Child et al., 2019	waveform	piano music	-

Variational AutoEncoders

Schematic principle of Variational Autoencoders (VAEs)



The encoder $q_\phi(\mathbf{z}|\mathbf{x})$ approximates the true posterior distribution $q_\theta(\mathbf{z}|\mathbf{x})$
 The decoder $p_\theta(\mathbf{x}|\mathbf{z})$ generates an approximation \hat{x} from the encoding

Main idea of variational inference: :

- The complete model $p(\mathbf{x}, \mathbf{z}) = p(\mathbf{x}|\mathbf{z})p(\mathbf{z})$, but the data follows complex distributions

- Exploit an approximate of the true posterior: $q_\phi(\mathbf{z}|\mathbf{x})$
- Variational inference: minimizing the difference between the approximation and the true density:

$$q_\phi^*(\mathbf{z}|\mathbf{x}) = \operatorname{argmin}_{q_\phi(\mathbf{z}|\mathbf{x}) \in \mathcal{Q}} D_{\text{KL}}[q_\phi(\mathbf{z}|\mathbf{x})|p(\mathbf{z}|\mathbf{x})]$$

Variational AutoEncoders

$$q_{\phi}^*(\mathbf{z}|\mathbf{x}) = \operatorname{argmin}_{q_{\phi}(\mathbf{z}|\mathbf{x}) \in \mathcal{Q}} D_{\text{KL}}[q_{\phi}(\mathbf{z}|\mathbf{x})|p(\mathbf{z}|\mathbf{x})]$$

- This can be further expressed as :

$$\log p_{\theta}(\mathbf{x}) = D_{\text{KL}}(q_{\phi}(\mathbf{z}|\mathbf{x})|p_{\theta}(\mathbf{z}|\mathbf{x})) + \mathcal{L}(\phi, \theta, \mathbf{x})$$

- It describes the quantity to model $\log p_{\theta}(\mathbf{x})$ minus the error we make by using an approximate q instead of the true p .
- We can maximize the **Evidenced Lower Bound (ELBO)**

$$\mathcal{L}(\phi, \theta, \mathbf{x}) = -D_{\text{KL}}(q_{\phi}(\mathbf{z}|\mathbf{x})||p_{\theta}(\mathbf{z})) + \mathbb{E}_{q_{\phi}(\mathbf{z}|\mathbf{x})}(\log p_{\theta}(\mathbf{x}|\mathbf{z}))$$



Variational AutoEncoders in Audio/music

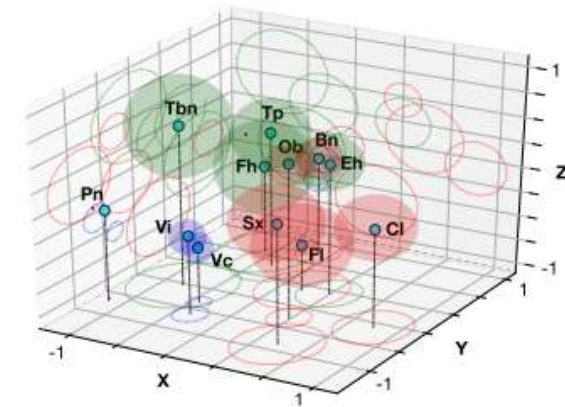
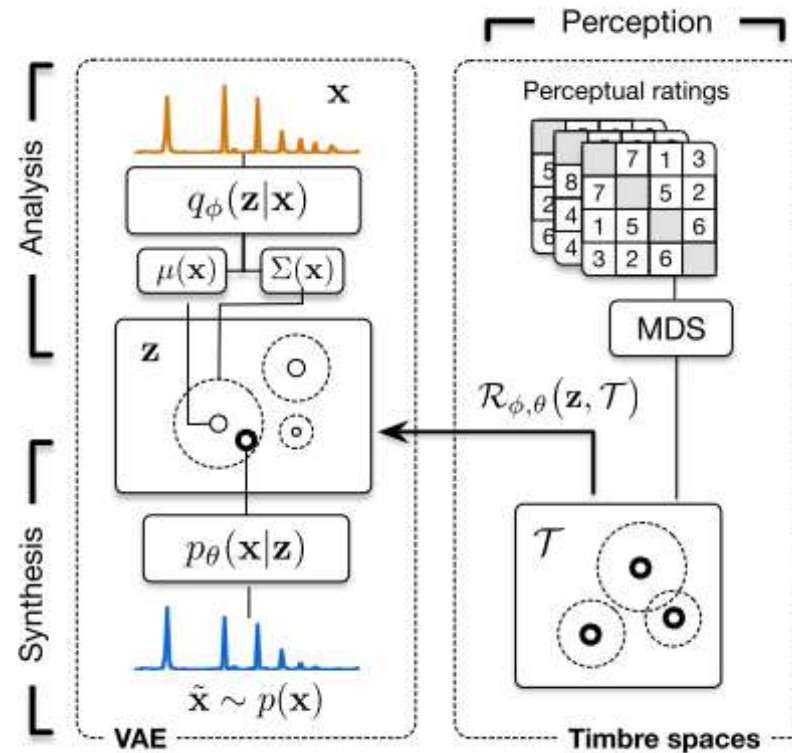
Many examples

Arch.	Name	Audio representation	Data	Conditioning
VAEs	Planet Drums	Aouameur et al., 2019	Mel-scaled mag. spec.	drums instrument ID
	Jukebox	Dhariwal et al., 2020	waveform	music artist & genre ID lyrics
	NOTONO	Bazin et al., 2020	mag. & IF	tonal instruments pitch
	FlowSynth	Esling et al., 2019	mag.	synth. sounds semantic tags
	Neural Granular Sound Synth.	Bitton et al., 2020	waveform	orchestral drums animals pitch instrument ID



Variational AutoEncoders in Audio/music

Regularizing the latent space with timbre spaces (perception)



Multi-dimensional scaling
(MDS)

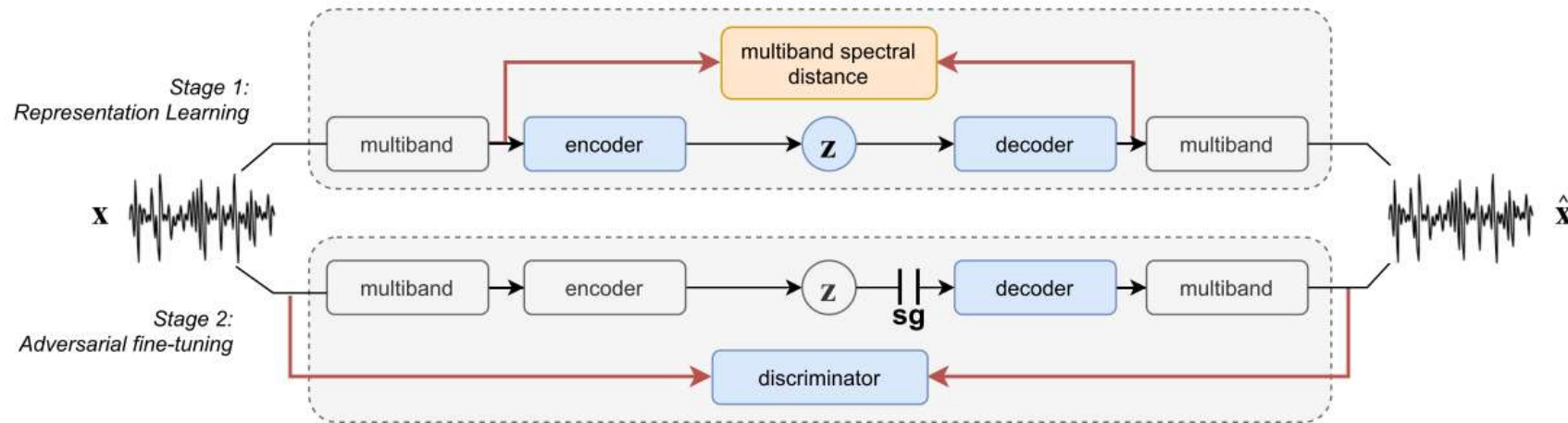
Variational AutoEncoders in Audio/music

Extensions

RAVE: Realtime Audio Variational autoEncoder

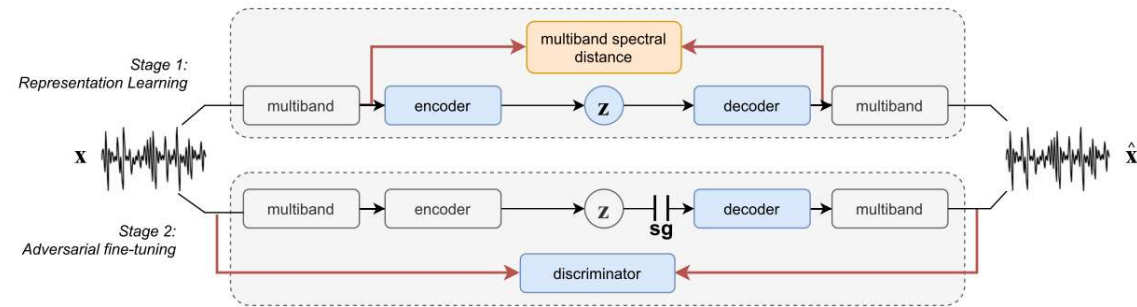
- Based on a two stage training:

1. representation learning with VAEs (stage 1)



Variational AutoEncoders in Audio/music

RAVE : some details



- The multispectral loss (from Engel2019 (DDSP))

$$S(\mathbf{x}, \mathbf{y}) = \sum_{n \in \mathcal{N}} \left[\frac{\|\text{STFT}_n(\mathbf{x}) - \text{STFT}_n(\mathbf{y})\|_F}{\|\text{STFT}_n(\mathbf{x})\|_F} + \log(\|\text{STFT}_n(\mathbf{x}) - \text{STFT}_n(\mathbf{y})\|_1) \right]$$

- Latent representation compactness
 - To avoid *posterior* collapse (e.g situation where **the learned latent space is ignored**)
 - Based on variance normalisation, rank estimation (using SVD on the latent space)



Variational AutoEncoders in Audio/music

RAVE : some results

- Evaluation (in 2021)

Model	MOS	95% CI	Training time	Parameter count
Ground truth	4.21	± 0.04	-	-
NSynth	2.68	± 0.04	~ 13 days	64.7M
SING	1.15	± 0.02	~ 5 days	80.8M
RAVE (Ours)	3.01	± 0.05	~ 7 days	17.6M

- Synthesis examples:

- Timbre transfer (model trained on speech, input :violin)



Violin input



Output (speech)

- Darbouka synthesis:



Original

- Reconstruction



Reconstructed

- Unconditional generation



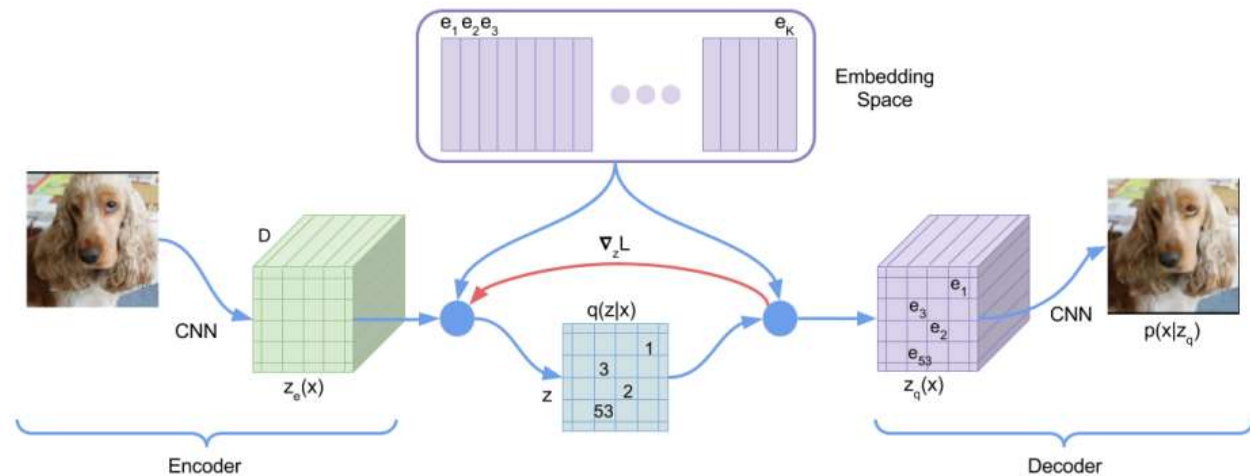
unconditioned



Vector-Quantized Variational AutoEncoders (VQ-VAEs)

- Combines VAEs with Vector quantization
- Helps to avoid *posterior* collapse of VAEs
- Offers the flexibility of a **discrete** neural representation

- Main principle

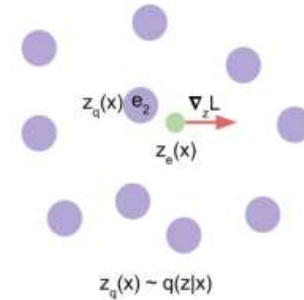


Vector-Quantized Variational AutoEncoders (VQ-VAEs)

- Discrete latent representation
- The discrete latent variables are obtained by nearest neighbour look-up

$$q(z = k|x) = \begin{cases} 1 & \text{for } k = \operatorname{argmin}_j \|z_e(x) - e_j\|_2, \\ 0 & \text{otherwise} \end{cases}$$

$$z_q(x) = e_k, \quad \text{where } k = \operatorname{argmin}_j \|z_e(x) - e_j\|_2$$



Vector-Quantized Variational AutoEncoders (VQ-VAEs)

- Learning
 - A loss function with three components
 1. A reconstruction loss (or data term)
 2. A dictionary learning term (VQ):
 3. A commitment loss (to force a joint learning of encoder and dictionary)

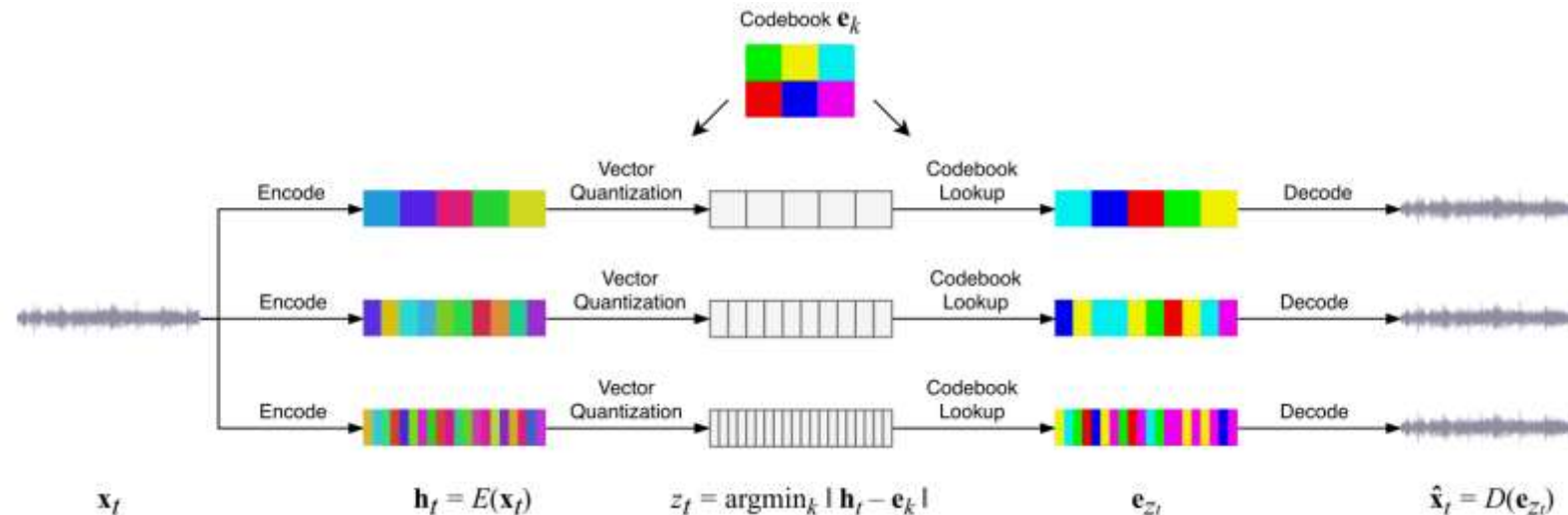
$$L = \log p(x|z_q(x)) + \|\text{sg}[z_e(x)] - e\|_2^2 + \beta \|z_e(x) - \text{sg}[e]\|_2^2$$



VQ-VAEs in Audio and Music

An example with Jukebox

- Based on hierarchical VQ-VAE (VQ-VAE2), trained with an additional spectral loss
- Combined with sparse transformers for learning the latent prior for generation

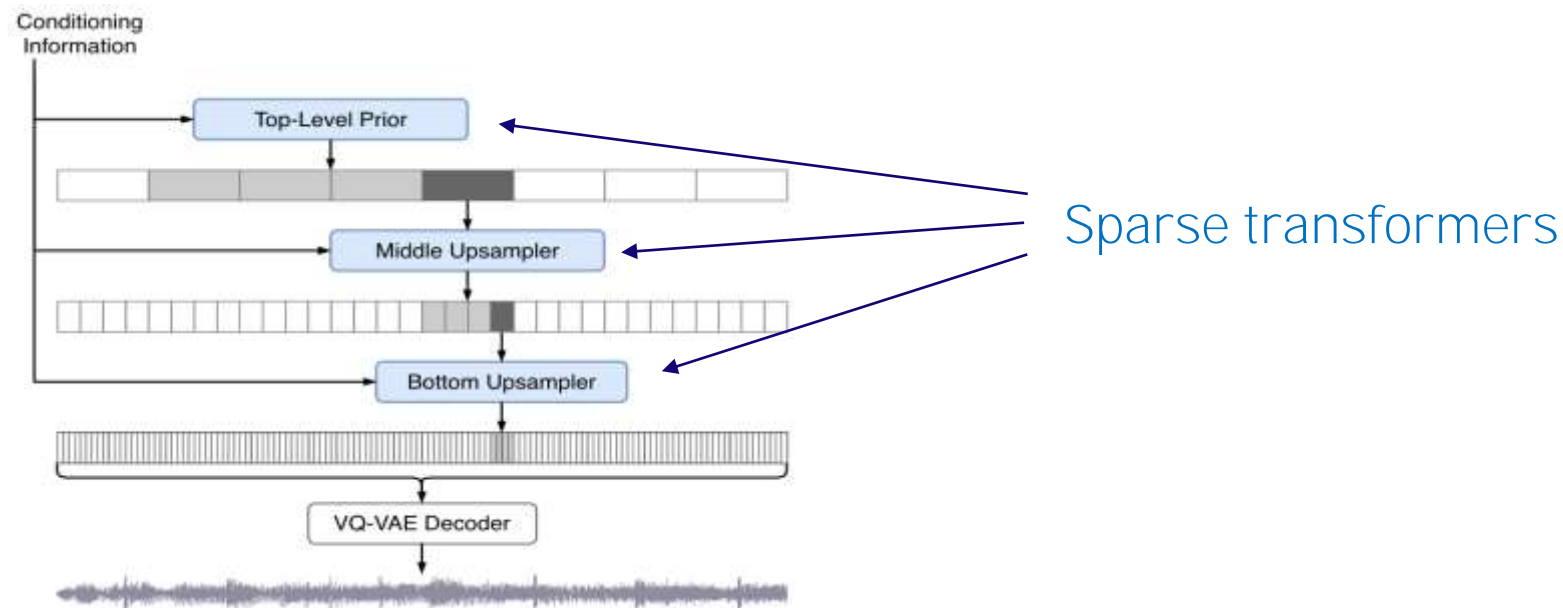


VQ-VAEs in Audio and Music

An example with Jukebox

- Learning the latent prior once the separate VQ-VAEs are trained

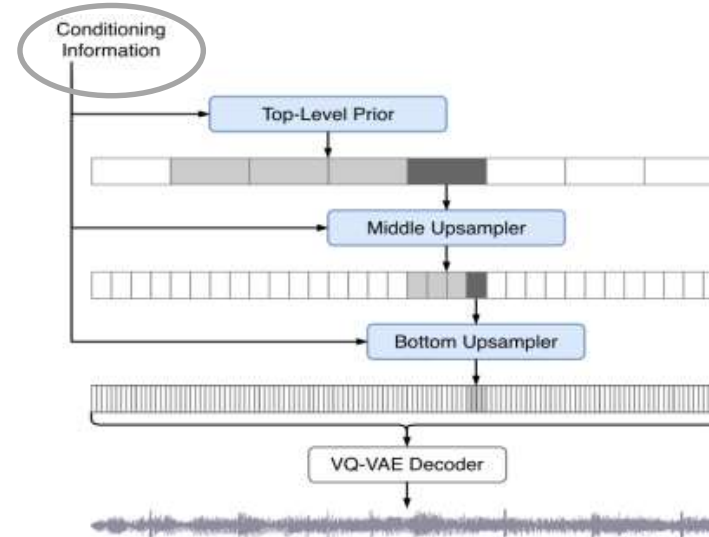
$$\begin{aligned} p(\mathbf{z}) &= p(\mathbf{z}^{\text{top}}, \mathbf{z}^{\text{middle}}, \mathbf{z}^{\text{bottom}}) \\ &= p(\mathbf{z}^{\text{top}})p(\mathbf{z}^{\text{middle}}|\mathbf{z}^{\text{top}})p(\mathbf{z}^{\text{bottom}}|\mathbf{z}^{\text{middle}}, \mathbf{z}^{\text{top}}) \end{aligned}$$



VQ-VAEs in Audio and Music

An example with Jukebox

- Conditioning for controlling the synthesis



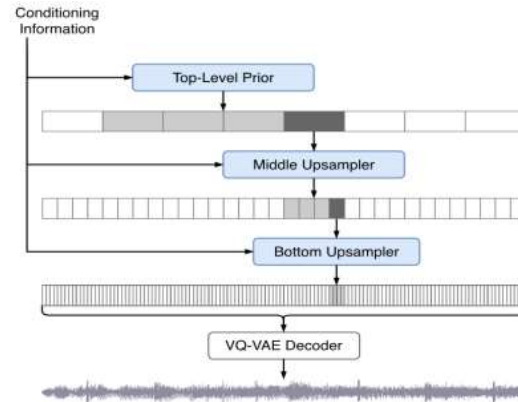
- **Artist, Genre, and Timing Conditioning** (to allow to learn patterns that depend on the structure... such as applause at the end)
- **Lyrics Conditioning** (with necessity to learn lyrics/audio alignment)



VQ-VAEs in Audio and Music

An example with Jukebox

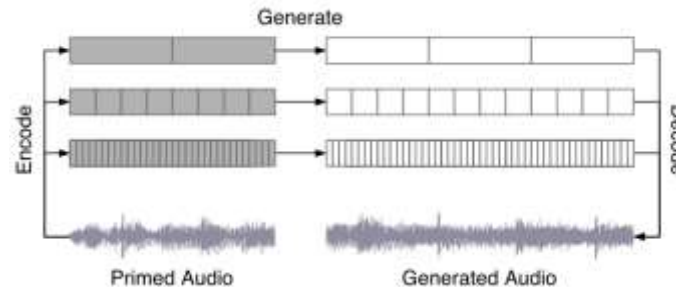
- Sampling methods for generating music



Windowed sampling for modelling sequences longer than initial context



Primed sampling: generate continuations by converting input into the VQ-VAE codes and sampling the subsequent codes in each level.



VQ-VAEs in Audio and Music

An example with Jukebox

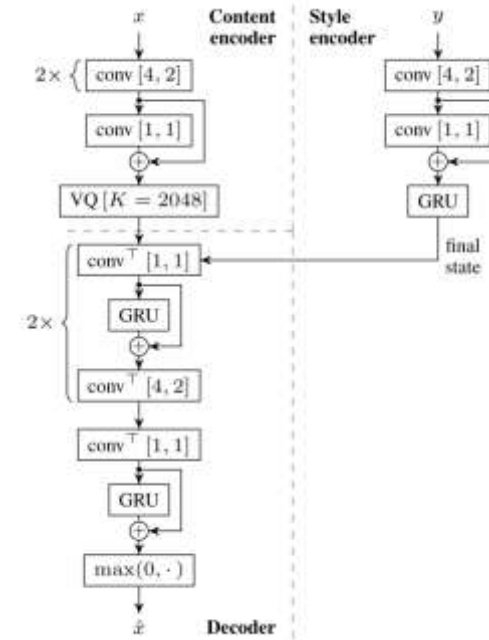
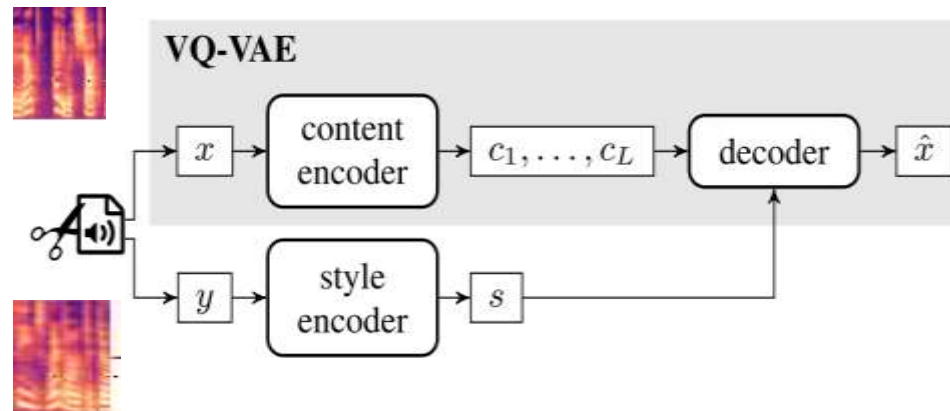
- Sound examples
 - Completion (with context of 12s of existing songs in the training)
 - Re-renditions (using pairs of lyrics-artist existing in the training)
 - Generation with novel lyrics (generated by GPT-2)
 - Generation with novel voices (by interpolating existing voice embeddings)
- Many raw examples at <https://jukebox.openai.com/>
- Some curated examples at <https://openai.com/blog/jukebox/>
 - One example of continuation with unknown lyrics:
<https://jukebox.openai.com/?song=795460096>
- Original model is rather slow at sampling (9 hours to render 1' of music)



VQ-VAEs in Audio and Music

Another example for one-shot music style transfer

- Content is encoded using a VQ-VAE
- Style is encoded using a self-supervised strategy (y is an audio-augmented version of a different segment than x , taken from the same recording)



VQ-VAEs in Audio and Music

Another example for one-shot music style transfer

- Many sound examples at: https://adasp.telecom-paris.fr/rc/demos_companion-pages/cifka-ss-vq-vae/#examples
- Two examples
 1. Synthetic example

Content input



Style input



Target



Output (VQ-VAE)



2. Real example

Content input



Style input



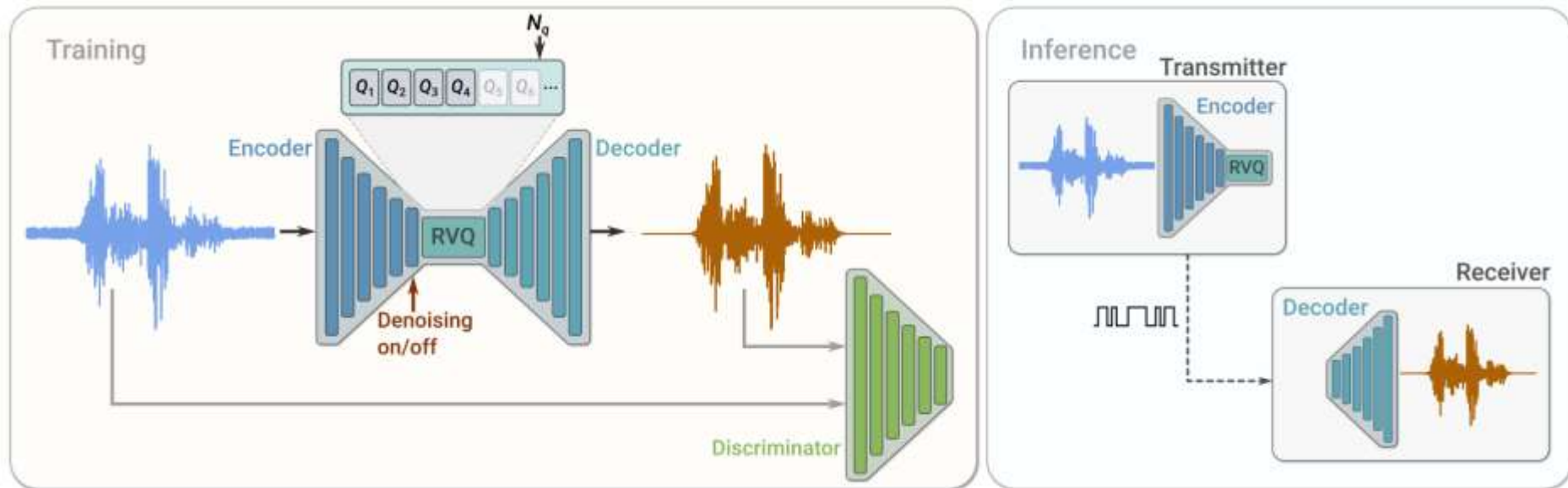
Output (VQ-VAE)



Discrete neural representation:

Soundstream: another powerful generative model

- Designed for audio compression
- Exploits Residual Vector Quantization (RVQ)
- trained end-to-end together with a discriminator using the mix of adversarial and reconstruction losses



Discrete neural representation: Soundstream: an other powerfull generative model

- Interest of RVQ

A concrete example with regular VQ :

- a codec with a target bitrate $R = 6000$ bps.
- For an audio at $F_s = 24000$ Hz (*striding factor of $M = 32$*), each second of audio is represented by $S = 75$ frames
- This leads to $r = 6000/75 = 80$ bits allocated to each frame.
- Using a plain vector quantizer, this requires storing a codebook with $N = 2^{80}$ vectors (this is Huge !!)

- RVQ = multi-stage Vector quantizer

- Cascade N_q layers of VQ
- Total rate budget is uniformly allocated to each VQ,
- $R_i = r / N_q = \log_2(N)$.
- Example: with $N_q = 8$, each quantizer uses a codebook of size $N = 2^{(r/N_q)} = 2^{(80/8)} = 1024$.

Algorithm 1: Residual Vector Quantization

Input: $y = \text{enc}(x)$ the output of the encoder, vector quantizers Q_i for $i = 1..N_q$

Output: the quantized \hat{y}

$\hat{y} \leftarrow 0.0$

residual $\leftarrow y$

for $i = 1$ to N_q **do**

$\hat{y} += Q_i(\text{residual})$

 residual $-= Q_i(\text{residual})$

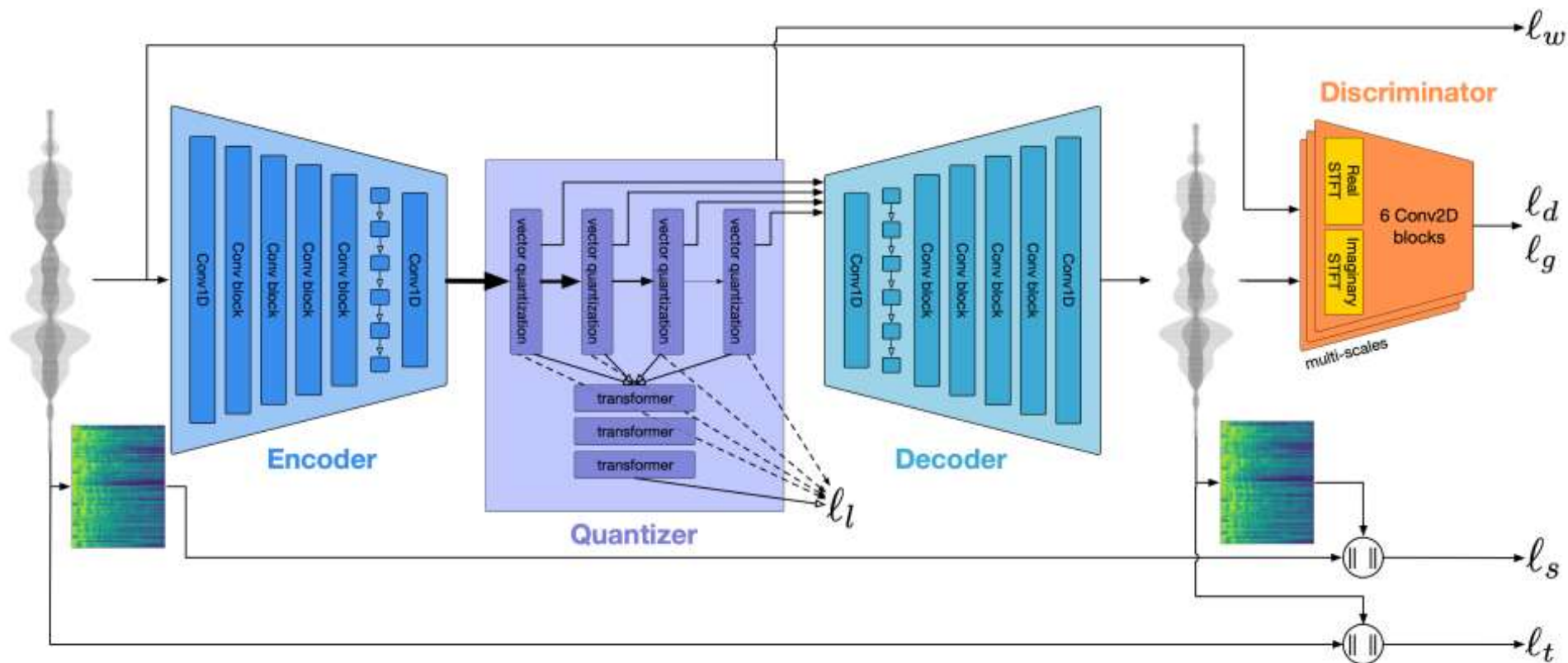
return \hat{y}



Discrete neural representation:

EncoDec: a slight extension of Soundstream

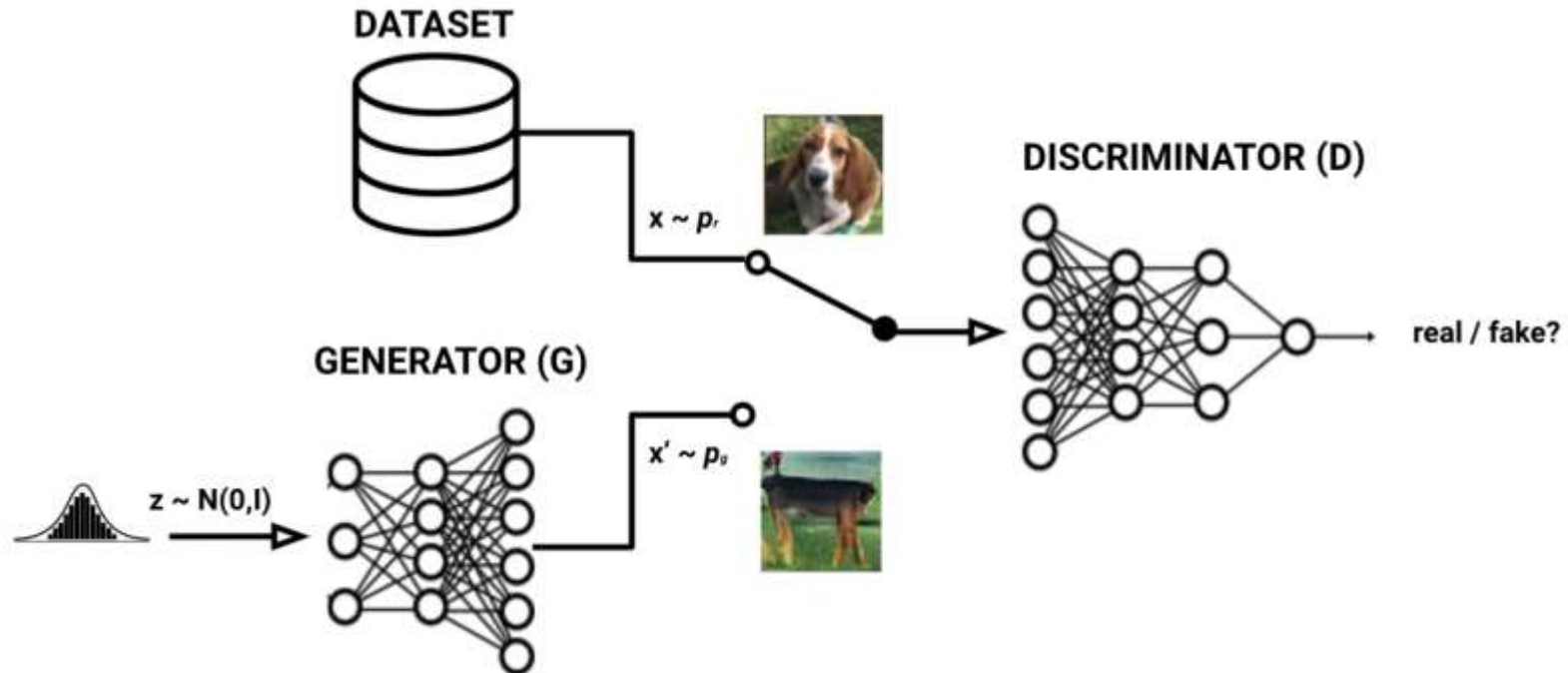
- E.g. Use of a small transformer model for better multi-stage VQ



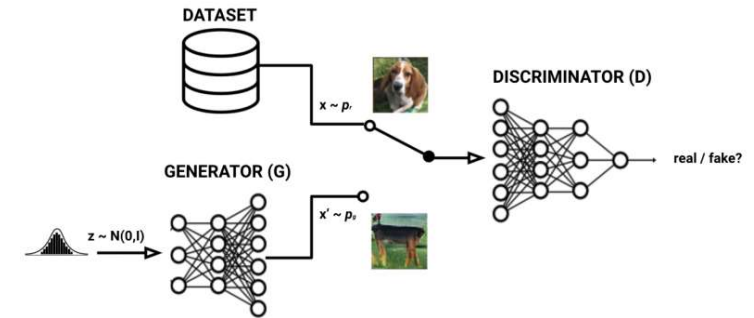
GANs, Diffusion models for audio generation

Generative Adversarial Networks (GANs)

- Principle of GANs



Generative Adversarial Networks (GANs)

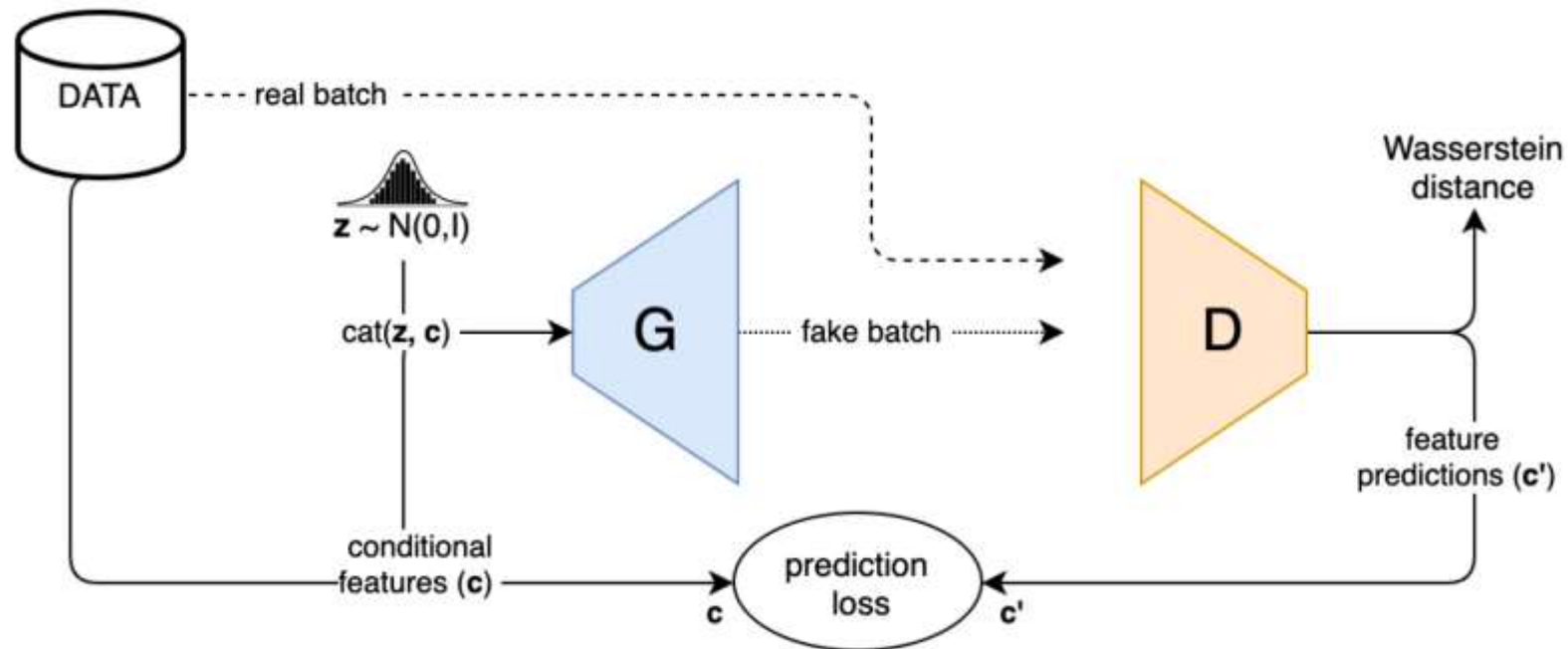


- More formally
 - a generative network $G_\theta(\mathbf{z})$ that outputs $x_g \sim p_g$ from a random input \mathbf{z}
After training, the output should follow the targeted probability distribution p_r
 - a discriminative network $D_\beta(\mathbf{x})$ trained to predict if the input comes from the real p_r or from the generated distribution p_g
- Optimization problem: a competitive objective

$$\min_{G_\theta} \max_{D_\beta} V(D_\beta, G_\theta) = E_{\mathbf{x} \sim p_r} [\log D_\beta(\mathbf{x})] + E_{\mathbf{x} \sim p_g} [1 - \log D_\beta(G_\theta(\mathbf{z}))]$$

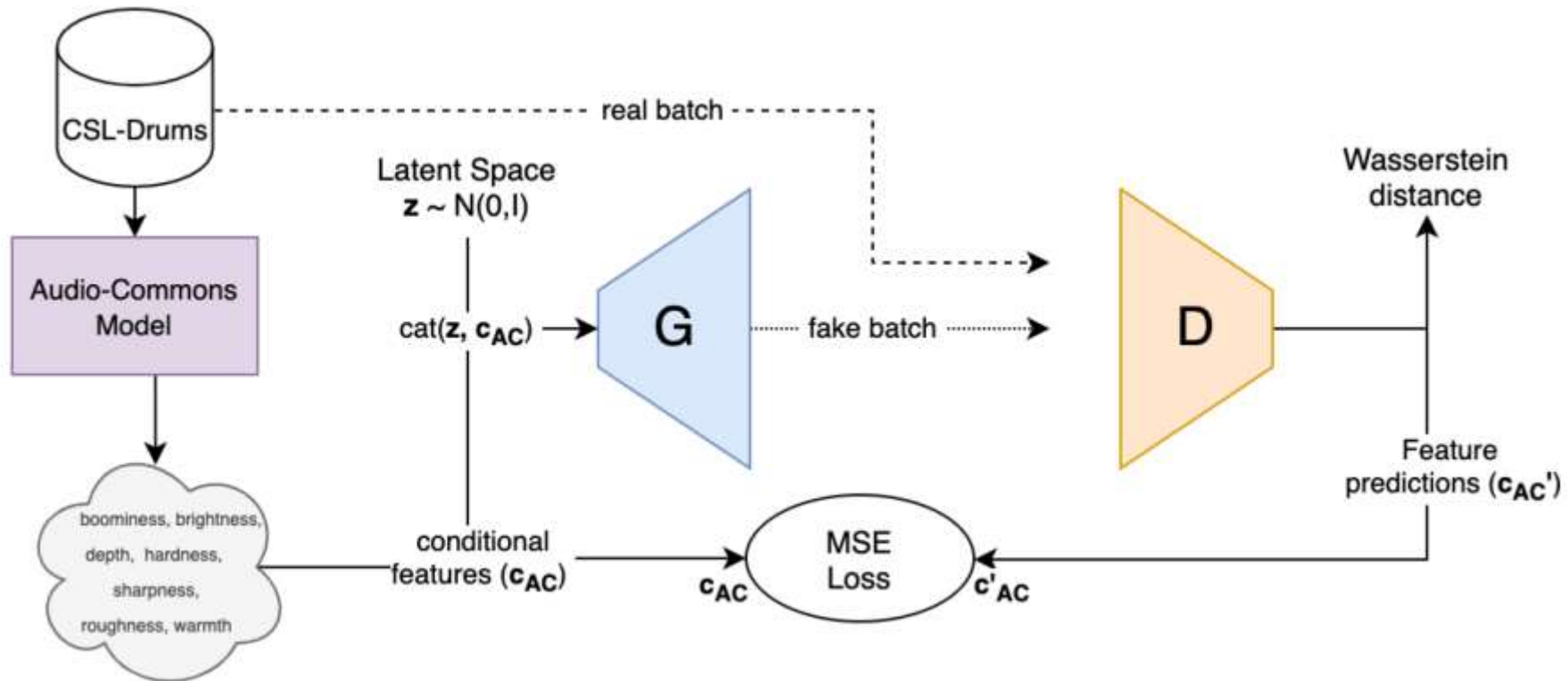
Generative Adversarial Networks (GANs)

- Principle of conditional GANs for audio synthesis



Generative Adversarial Networks (GANs)

- DrumGAN: Synthesis of Drum sounds with timbral feature Conditioning using GANs synthesis



Generative Adversarial Networks (GANs)

- DrumGAN: Demo
 - <https://sites.google.com/view/drumgan?pli=1>
- DrumGAN VST: A Plugin for Drum Sound Analysis/Synthesis with Autoencoding GANs
 - <https://cslmusicteam.sony.fr/drumgan-vst/>
 - Short demo on **Converting beatbox to drums**



Original

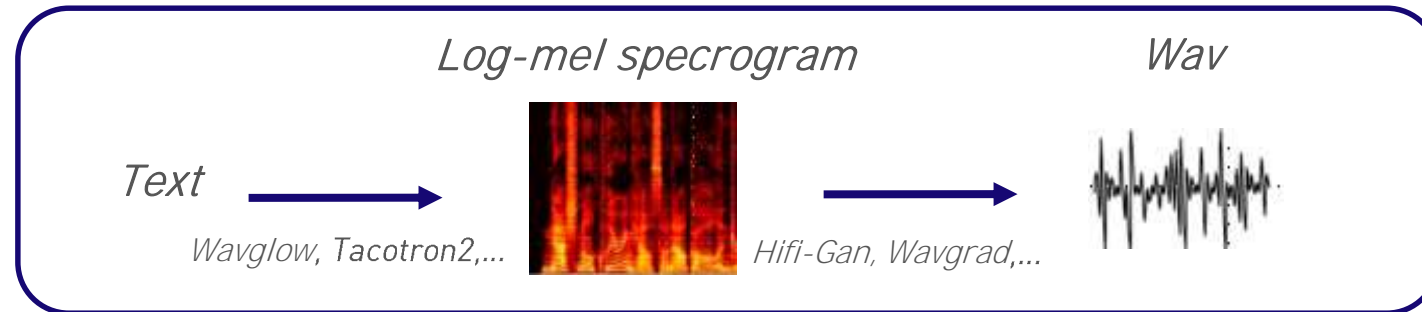


Original+decoded



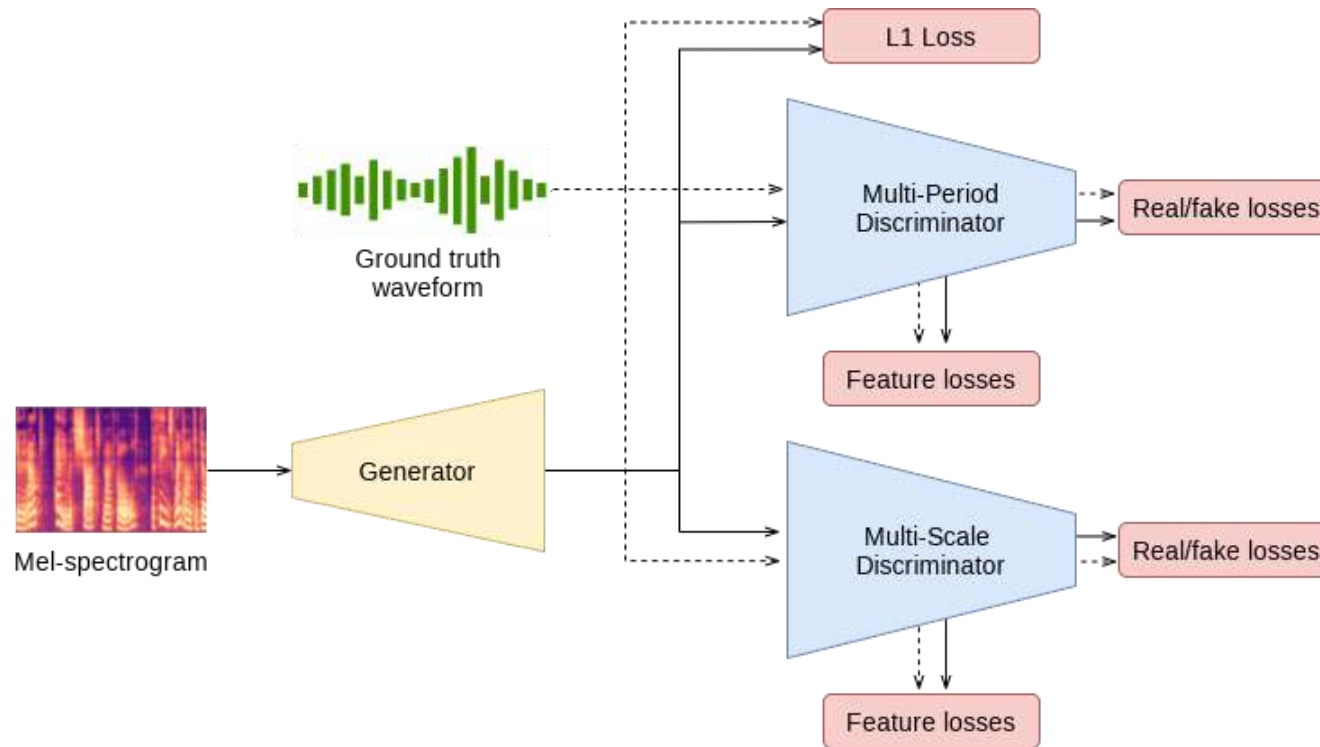
A classic pipeline for sound generation

- For example, a classic pipeline in recent Text-to-speech



Hifi-Gan

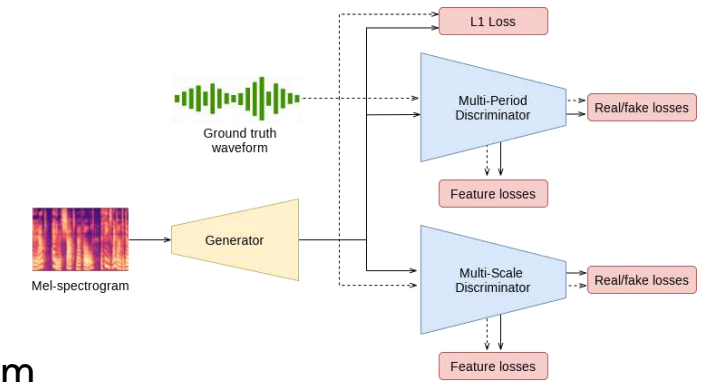
- High computational efficiency and high sample quality
- 1 Generator (CNN) and 2 Discriminators



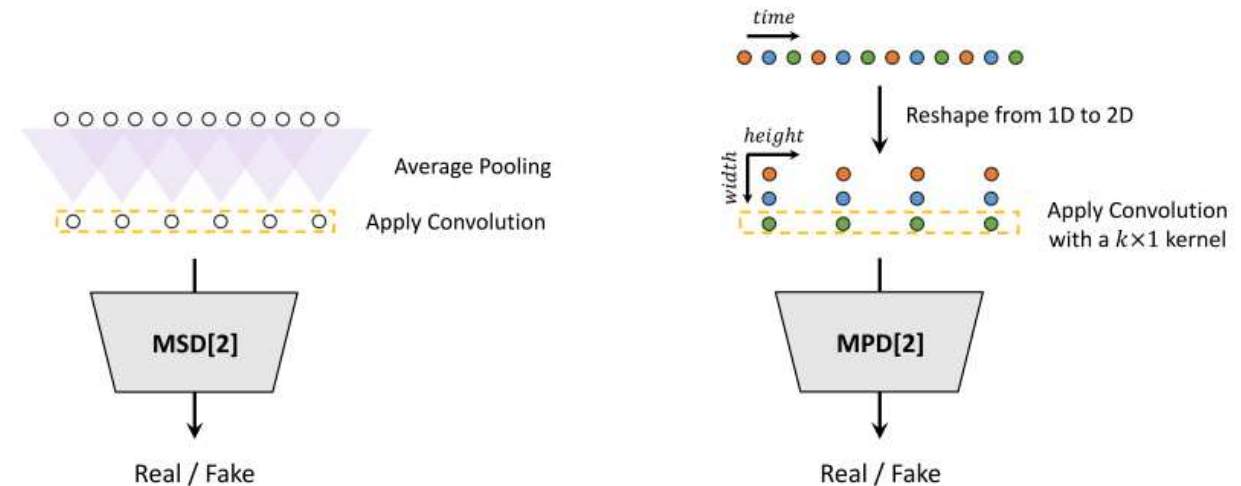
Hifi-Gan

2 Discriminators: MSD and MPD

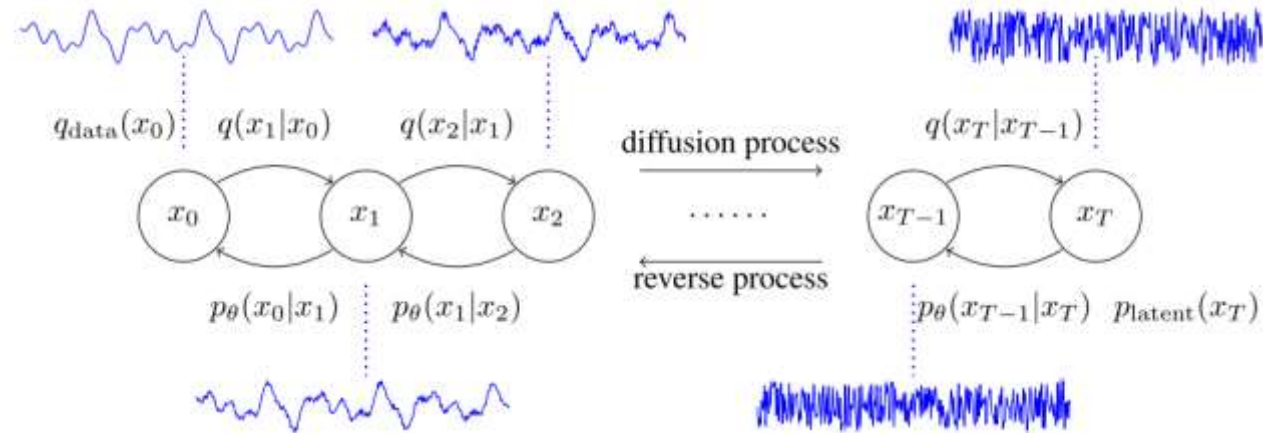
- **MPD = mixture of sub-discriminators**
- Sub-discriminators are designed to capture different implicit structures from each other by looking at different parts of an input audio
- each sub-discriminator only accepts equally spaced samples of an input audio
- the space (period) p is equal to $[2, 3, 5, 7, 11]$



- **MSD to evaluate audio sequence at multiple scale**
- MSD is a mixture of three sub-discriminators operating on different input scales: raw audio, $\times 2$ average-pooled audio, and $\times 4$ average-pooled audio



Diffusion models for audio synthesis ...



- Based on two processes: the diffusion process, and the reverse process
 - The **diffusion process** is defined by a fixed Markov chain from data x_0 to the latent variable x_T

$$q(x_1, \dots, x_T | x_0) = \prod_{t=1}^T q(x_t | x_{t-1})$$

where each of $q(x_t | x_{t-1})$ is fixed to $\mathcal{N}(x_t; \sqrt{1 - \beta_t} x_{t-1}, \beta_t I)$ for a small positive constant β_t

- The reverse process gradually converts the white noise signal into audio waveform through a Markov chain: .

$$p_{\text{latent}}(x_T) = \mathcal{N}(0, I), \quad \text{and} \quad p_{\theta}(x_0, \dots, x_{T-1} | x_T) = \prod_{t=1}^T p_{\theta}(x_{t-1} | x_t),$$



Diffusion models for audio synthesis ...

- The models are often strongly conditioned
- Example: wavgrad, specgrad conditioned on mel-spectrogram

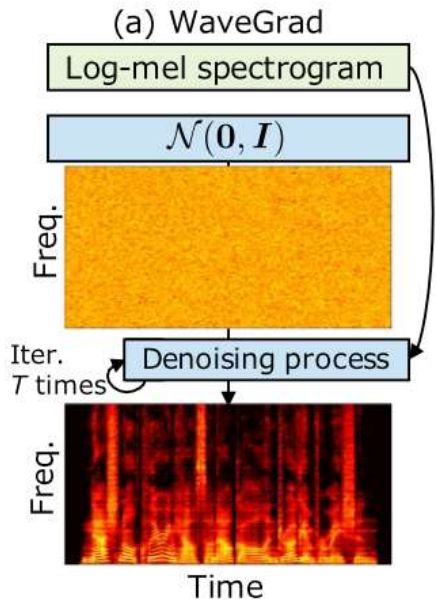
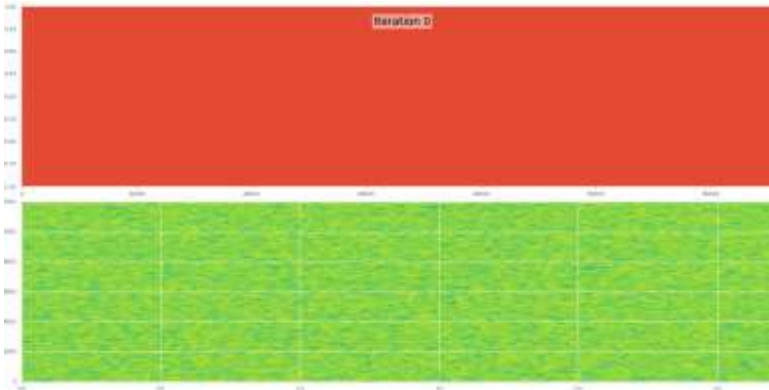


Illustration of the diffusion process (50 iterations)



Sound examples

reference



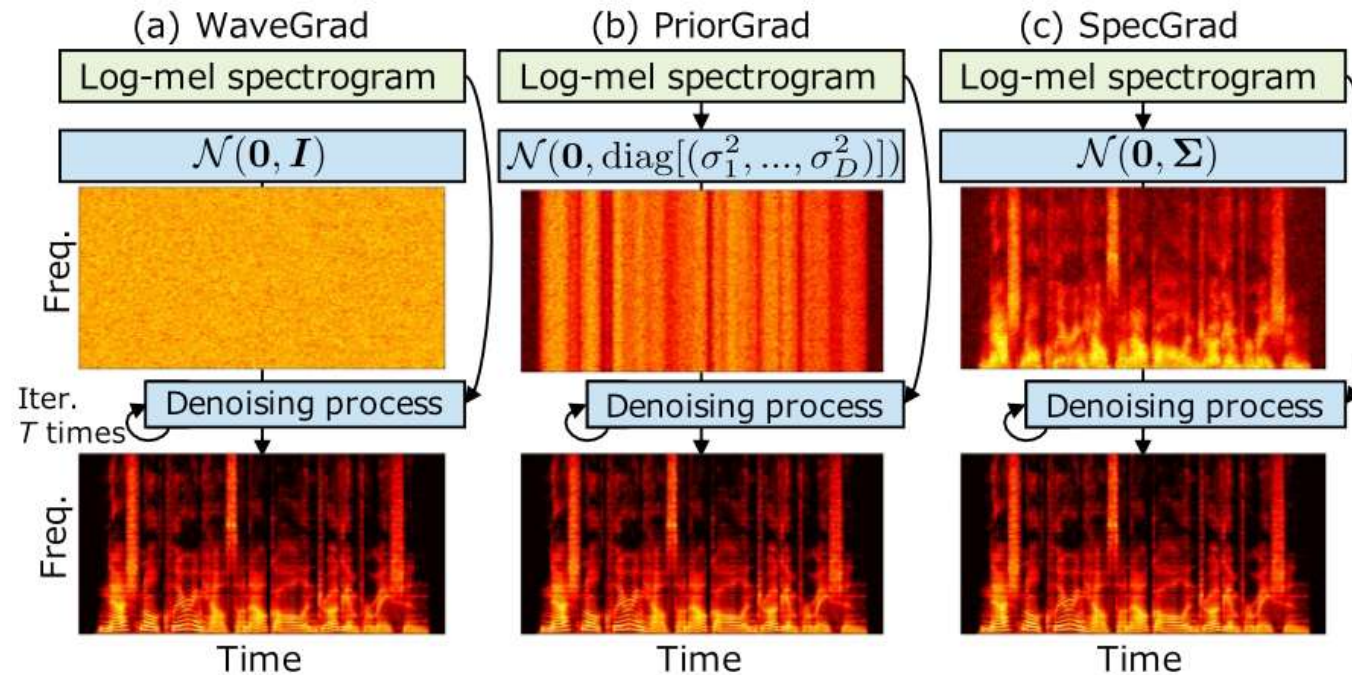
wavgrad



Diffusion models for audio synthesis ...

extensions of wavgrad

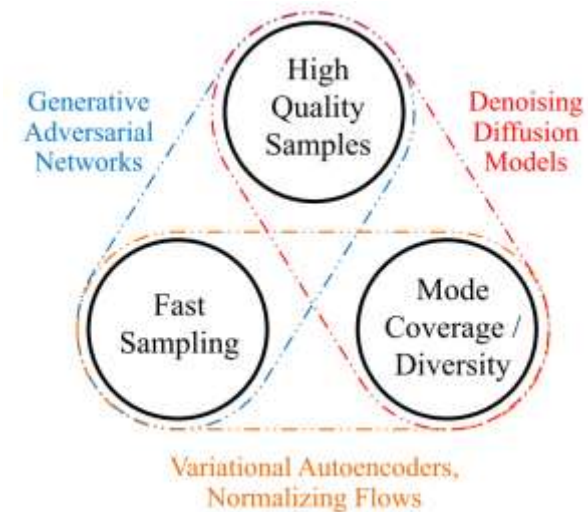
- The example of priorgrad, specgrad, ...



Diffusion models for audio synthesis ...

Combining diffusion models with GANs

The Generative learning trilemma



- **Example of Denoising Diffusion Gan:**

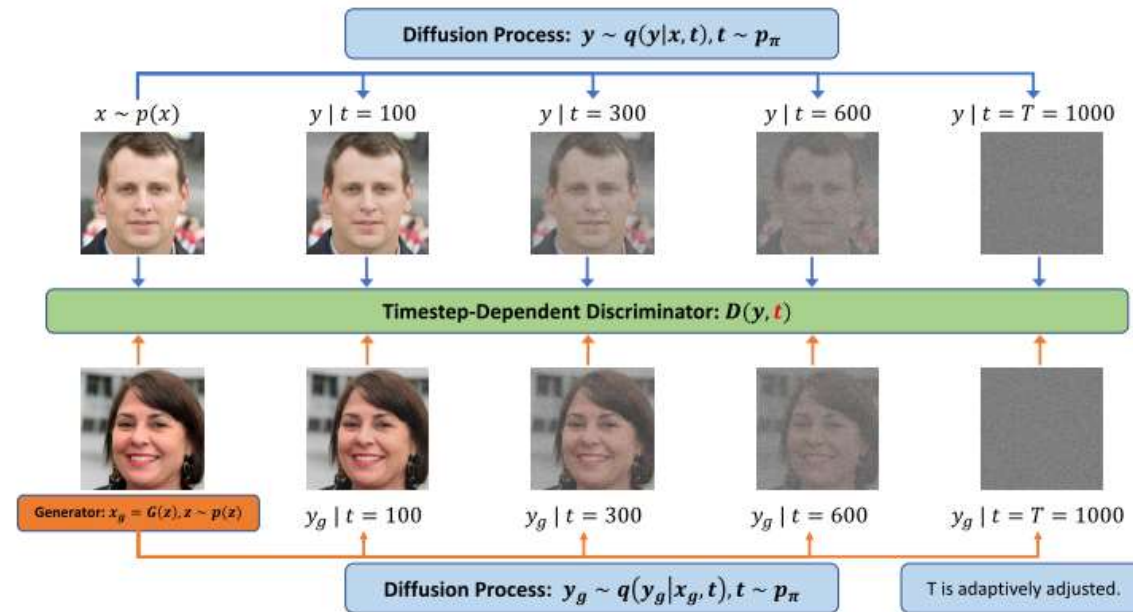
- Assumption: the slow sampling of diffusion models is due to the Gaussian assumption in the denoising distribution
- Propose to employ complex, multimodal denoising distributions.
- Propose denoising diffusion GANs, a diffusion model whose reverse process is parametrized by conditional GANs.
- Denoising diffusion GANs achieve several orders of magnitude speed-up compared to classic diffusion models for (image) generation

Diffusion models for audio synthesis ...

Combining diffusion models with GANs

Diffusion-Gan: Training GANS with Diffusion

- The discriminator learns to distinguish a diffused real image from a diffused fake image at all diffusion steps.
- Stabilizes the training of GANS ; Leads to improved performances (quality of images, complexity)



Diffusion models for audio synthesis ...

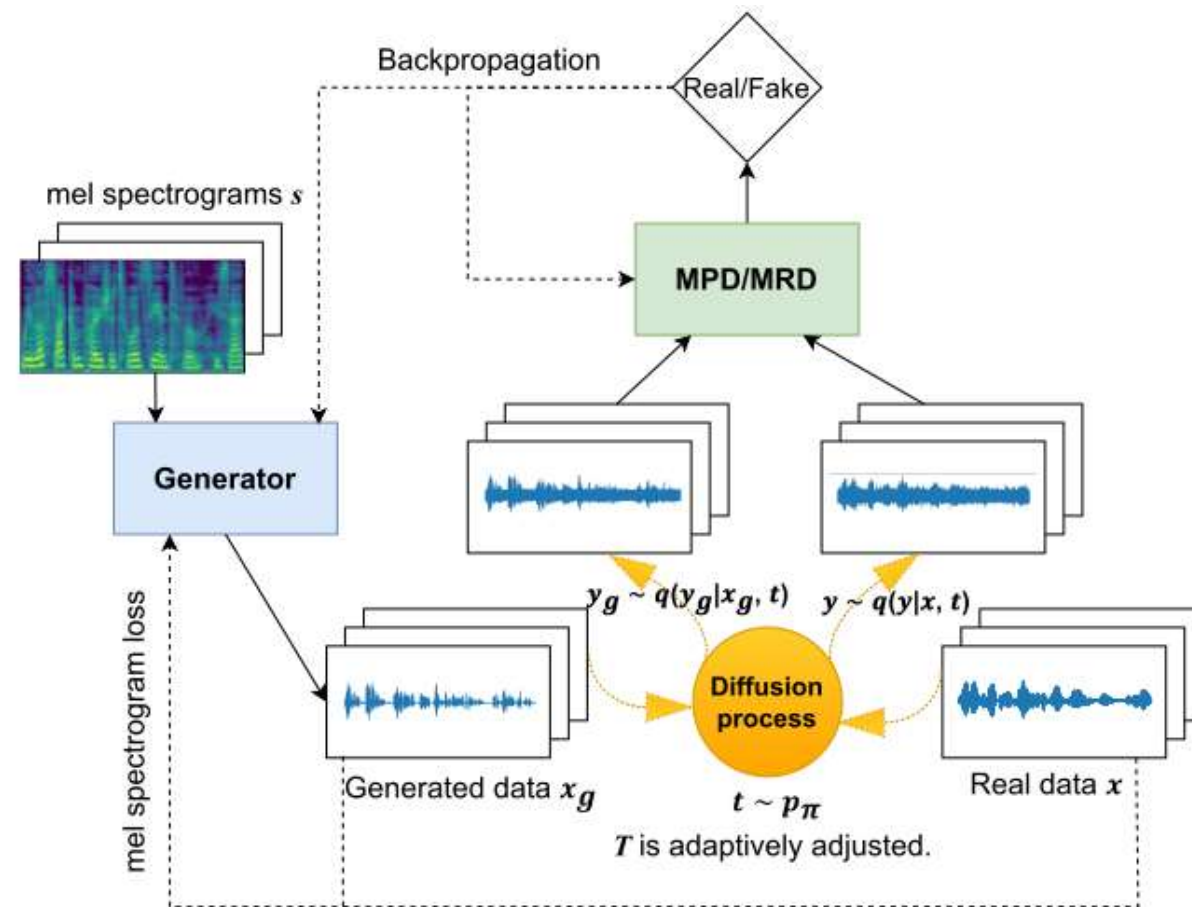
Combining diffusion models with GANs

SpecDiff-Gan

Combines principles of

- Diffusion-gans,
- Hifi-Gan
- and specgrad
- ...for *speech and music*

Ground truth	SpecDiff-Gan
speech	
piano	
drums	



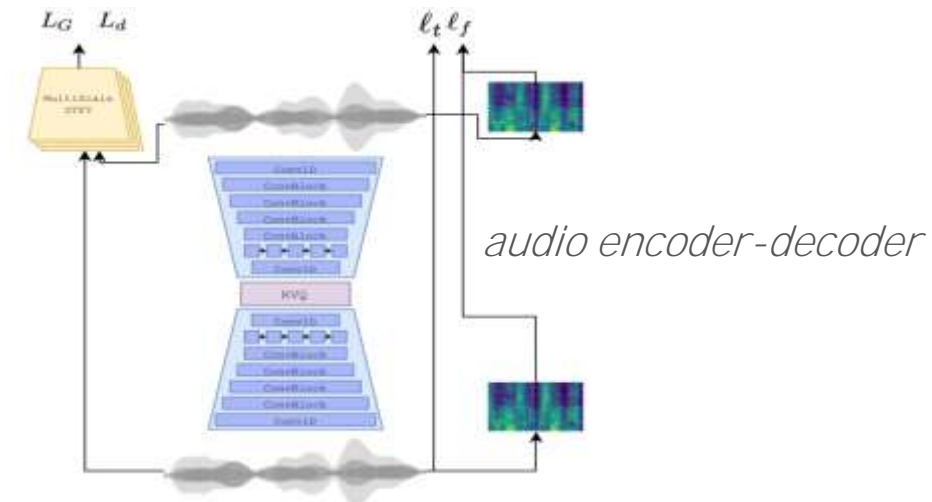
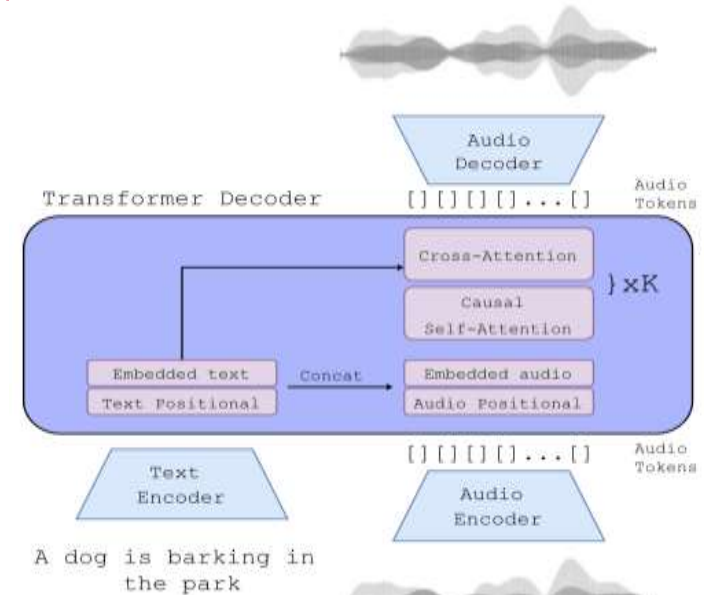
Cross Model audio generation : some examples

Towards « text-prompt » to audio

The exemple of AudioGen

- **AudioGen**

- 2 main steps:
 - (i) an audio encoder-decoder to learn a discrete audio representation (RVQ)
 - (ii) training a Transformer language model over the learnt codes obtained from the audio encoder, conditioned on textual features.
- **Some specificities:**
 - Text representation obtained using a pretrained T5 text encoder
 - For text adherence: cross-attention between audio and text to each attention block of the transformer.
 - Augmentation method that fuses pairs of audio samples and their respective text captions, thus creating new concept compositions during training
 - Uses Classifier Free Guidance (CFG) to improve generation for Low resolution (e.g. randomly unconditional training)



Towards « text-prompt » to audio

The exemple of AudioGen

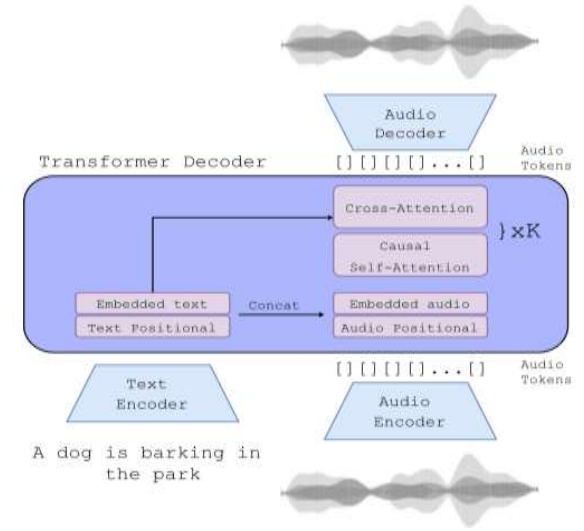
- Demo

Text prompt

a man speaks as birds chirp and dogs bark

male speech with horns honking in the background

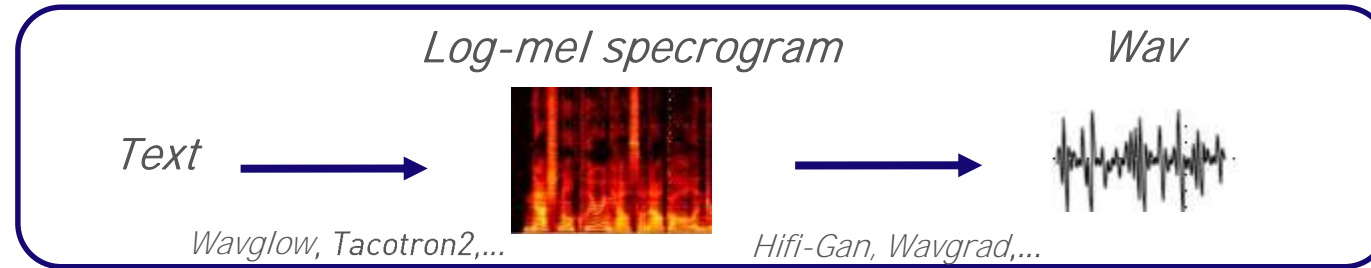
drums and music playing with a man speaking



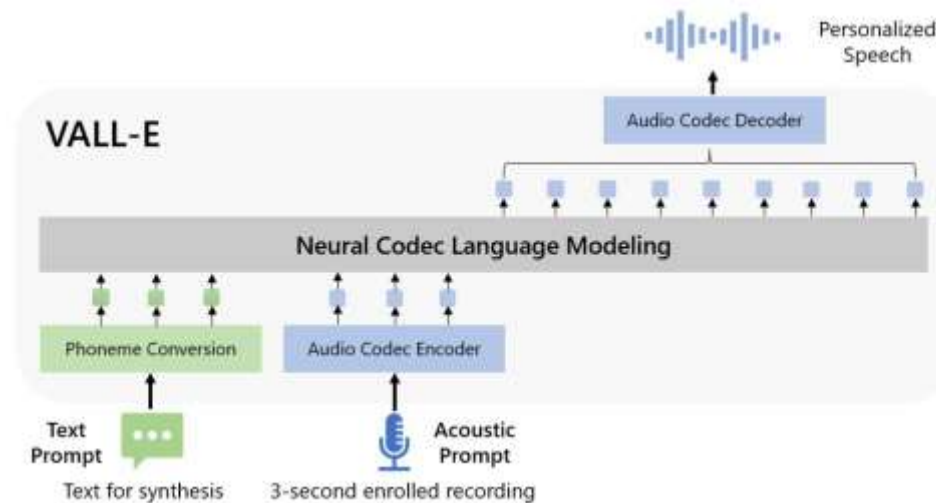
Vall-E: « text-to-speech (TTS) or *speech synthesis* »

another possible model of discrete neural model

- A classic pipeline in recent Text-to-speech



- Vall-E: A different pipeline with *discrete codes* as intermediate representation

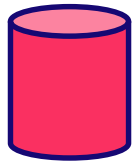


Vall-E: « text-to-speech (TTS) or *speech synthesis*

another possible model of discrete neural model

- **TTS as Conditional Codec Language Modeling**

$$\mathcal{D} = \{x_i, y_i\}$$



y_i : Audio signal

x_i : corresponding phoneme transcription

- **Use of an Tokenizer: a pre-trained neural audio codec (Encodec)**

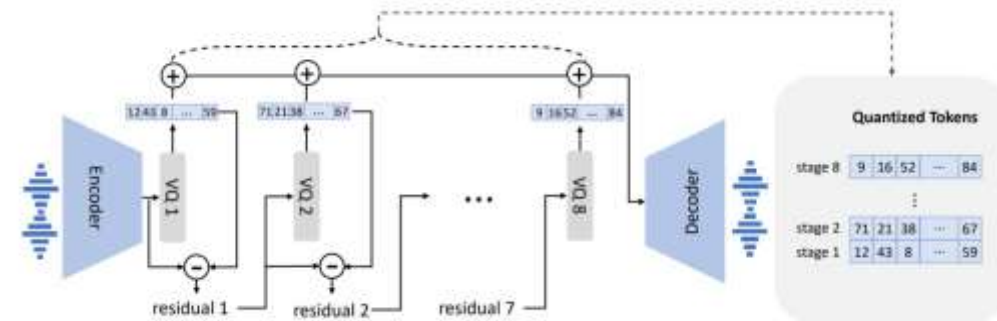
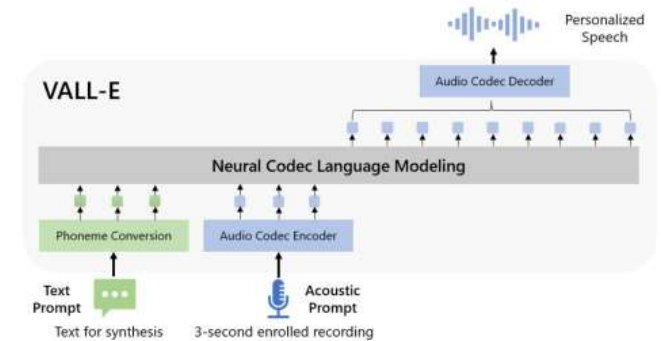
$$\text{Encodec}(y) = \mathbf{C}^{T \times 8}$$

$\mathbf{c}_{t,:}$: 8 codes for frame t

$\mathbf{c}_{:,j}$: code sequence for codebook j

- **Train a neural LM to generate acoustic codes with an optimisation objective:**

$$\max (p(\mathbf{C} | \mathbf{x}, \tilde{\mathbf{C}})) : \text{where } \tilde{\mathbf{C}} \text{ is the acoustic prompt}$$



Vall-E: « text-to-speech (TTS) or *speech synthesis*

another possible model of discrete neural model

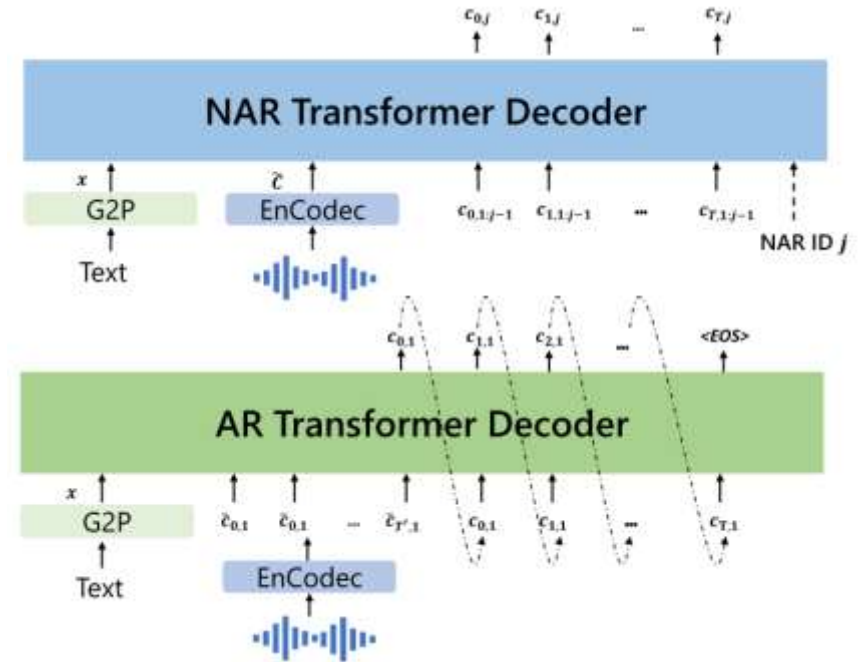
- The conditional codec language modelling

- Association of 2 transformer models
- An autoregressive model (AR) for the first codebook (e.g. *good quality*)

$$p(\mathbf{c}_{:,1} | \mathbf{x}, \tilde{\mathbf{C}}_{:,1}; \theta_{AR}) = \prod_{t=0}^T p(\mathbf{c}_{t,1} | \mathbf{c}_{<t,1}, \tilde{\mathbf{c}}_{:,1}, \mathbf{x}; \theta_{AR})$$

- An non-autoregressive (NAR) for the remaining ones (e.g. *less complex*)

$$p(\mathbf{C}_{:,2:8} | \mathbf{x}, \tilde{\mathbf{C}}; \theta_{NAR}) = \prod_{j=2}^{j=8} p(\mathbf{c}_{:,j} | \mathbf{C}_{:,<j}, \mathbf{x}, \tilde{\mathbf{C}}; \theta_{NAR})$$



Vall-E: « text-to-speech (TTS) or *speech synthesis* »

another possible model of discrete neural model




- Inference: In-Context Learning via Prompting
 - Converts the text into a phoneme sequence and encodes the enrolled recording into an acoustic matrix, forming the phoneme prompt and acoustic prompt.
 - Both prompts are used in the AR and NAR models.
 - For the AR model, sampling-based decoding conditioned on the prompts is used
 - For the NAR model, greedy decoding is used to choose the token with the highest probability.
 - Finally, the neural codec decoder is used to generate the waveform conditioned on the eight code sequences.









Vall-E demo

(replay)

- Vall-E( Microsoft)
- Zero-shot TTS

Text	Speaker prompt	Ground Truth	VALL-E
They moved thereafter cautiously about the hut groping before and about them to find something to show that Warrenton had fulfilled his mission			

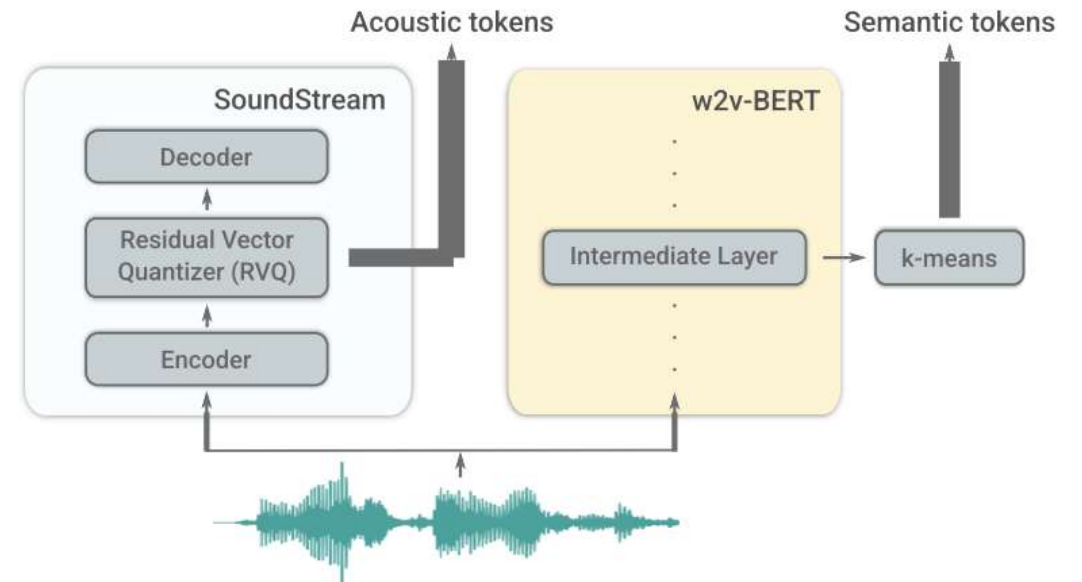
- ... or keeping the speaker emotion

Text	Emotion	Speaker prompt	VALL-E
We have to reduce the number of plastic bags.	Anger		
	Sleepy		
	Amused		



AudioLM: using language models for audio generation

- 3 main components:
 - (i) A tokenizer model, which maps the input audio into a sequence of discrete tokens from a finite vocabulary
 - (ii) A decoder-only Transformer language model that operates on the discrete tokens. At inference time, the model predicts the token sequence autoregressively.
 - (iii) A detokenizer model, which maps the sequence of predicted tokens back to audio
- Motivation for the dual-token model (for speech signal):
 - **Acoustic tokens:** speaker identity and recording conditions (mostly)
 - **Semantic tokens:** capture the linguistic content (mostly)



AudioLM: using language models for audio generation

- Hierarchical Modeling of Semantic and Acoustic Tokens
 - In all stages: a separate (decoder) Transformer is trained for predicting next tokens given previous tokens



- **At inference**
 - *Unconditional generation:* semantic tokens are sampled unconditionally and used as conditioning for acoustic modeling.
 - *Acoustic generation:* ground-truth semantic tokens are extracted from a test sequence as conditioning to generate the acoustic tokens.
 - *Generating continuations (from a short prompt):*
 - 1) generation of the continuation of semantic tokens autoregressively;
 - 2) concatenation of the entire semantic token sequence with the coarse acoustic tokens of the prompt and then feed as conditioning to the coarse acoustic model, which then samples the continuations of the corresponding acoustic tokens
 - 3) the coarse acoustic tokens are processed with the fine acoustic model.
 - 4) both the prompt and the sampled acoustic tokens are fed to the SoundStream decoder to reconstruct audio

AudioLM: using language models for audio generation

Demo

- **Speech (continuation)**

Original (speech)



Prompt



Continuation by AudioLM



- **Acoustic generation** « we sample the acoustic tokens given the semantic tokens extracted from the original samples from LibriSpeech test-clean»

Original (speech)



Generated (1)



Generated (2)



- **Generation without semantic tokens** : « Continuations with a language model trained on the acoustic tokens only (without semantic tokens)»

Example 1



Example 2



- **An interesting example : piano continuation**

Original



Prompt



Continuation by acoustic-only model



Continuation by AudioLM



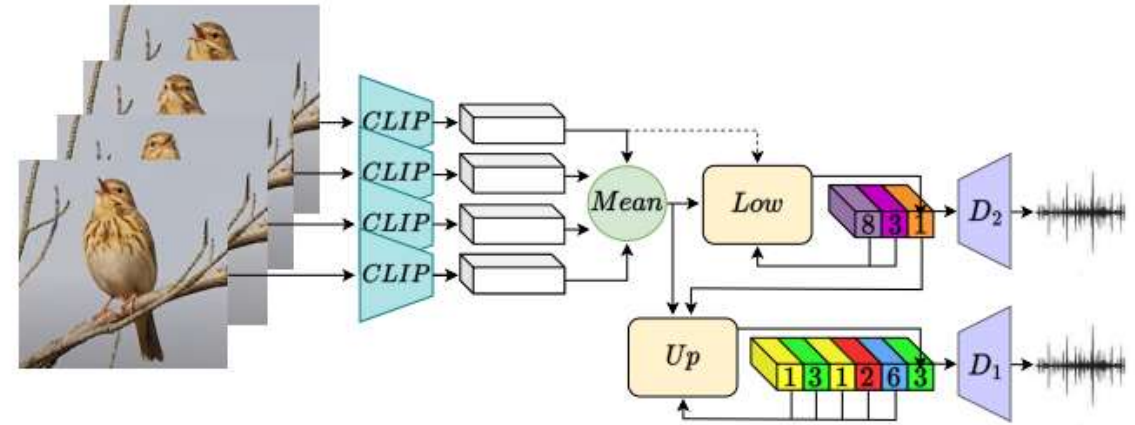
A prompt of a known piano sonata example (Beethoven N° 18) is continued in ... another known piano sonata (Beethoven – Moonlight sonata) !



Towards image-to-audio: the model IM2WAV

- IM2WAV: a Transformer-based audio Language Model (LM) conditioned on image representation

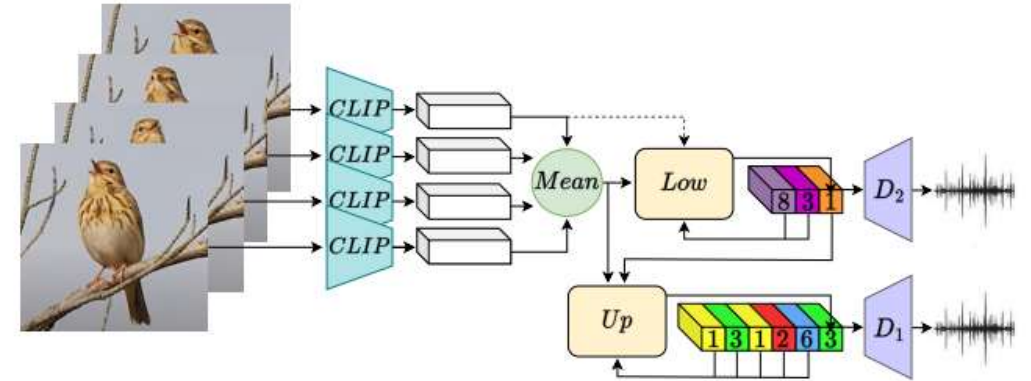
- 3 main components:
 - (i) an audio encoder-decoder with a discrete internal representation (VQ-VAE)
 - (ii) a pre-trained image encoder (CLIP)
 - (iii) an audio language model which operates over the discrete audio tokens (autoregressive sparse transformer)
- Some specificities:
 - CLIP embeddings trained in a multimodal context
 - Use Classifier Free Guidance (CFG) to improve generation for Low resolution (e.g. randomly unconditional training)



- **Parameters:**
 - 16 kHz Sampling frequency (4 s of sound)
 - 5 Conv. Layers for VQ-VAE (stride 2) Enc/dec.
 - 1st codebook after 3 layers (downsampling of 8)
 - 2nd codebook after 5 layers (downsampling of 32)
 - 2 k (resp. 5k) tokens in the UP (resp. LOW) model
 - Codebook: 2048 codes, embedding size of 128
 - Transformer: 48 layers, sparse attention

The IM2WAV model

- Demo



DALL-E Image Guided Audio Generation Example

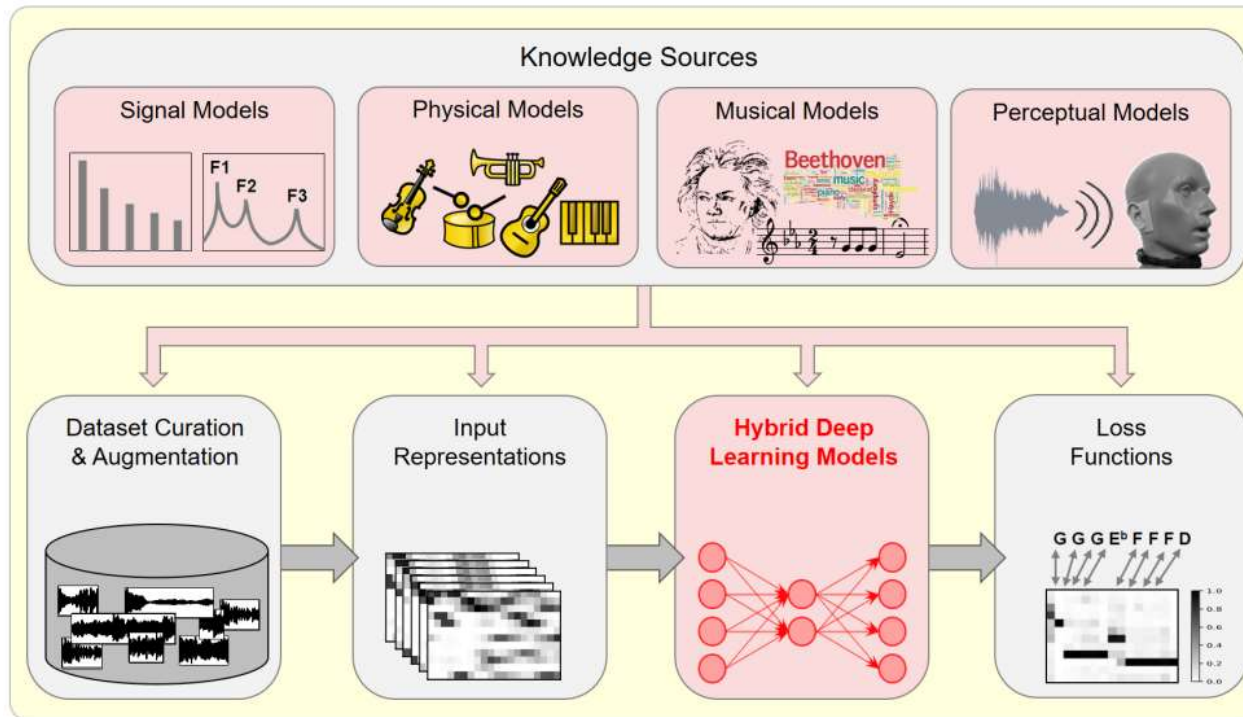


Towards hybrid deep learning ...

Towards Hybrid deep learning approaches

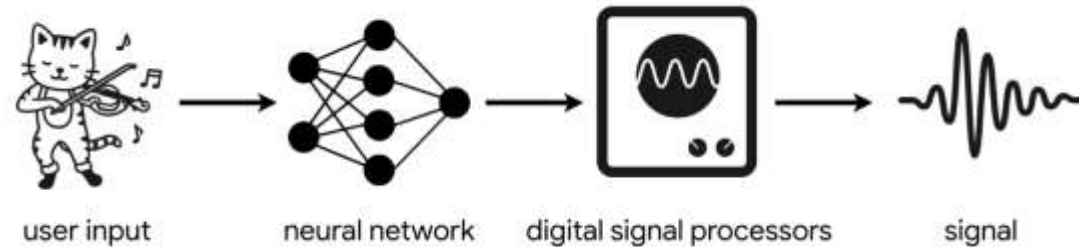
- Coupling model-based and deep learning:

Example with Hybrid deep model for Music signals



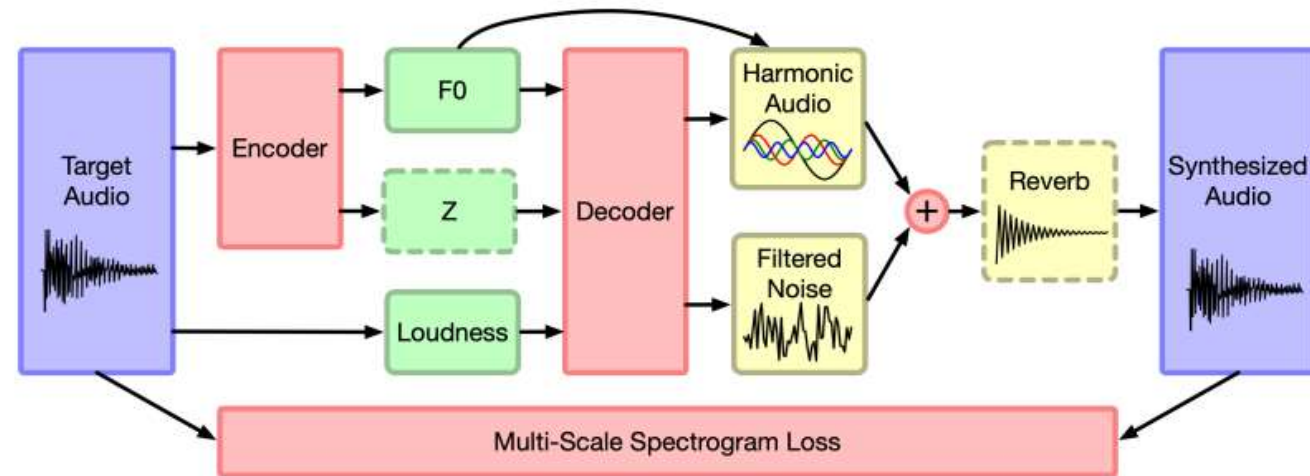
Towards Hybrid deep learning approaches

- Coupling model-based and deep learning
- For example, using deep learning for learning the parameters of a signal processing model



Towards Hybrid deep learning approaches

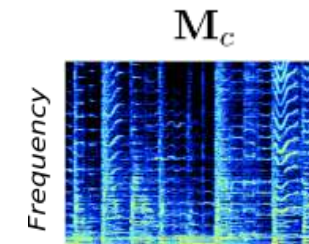
- The example of DDSP



- A multi-scale spectral loss $\mathcal{L}_{rec} = \sum_c \mathcal{L}_c$

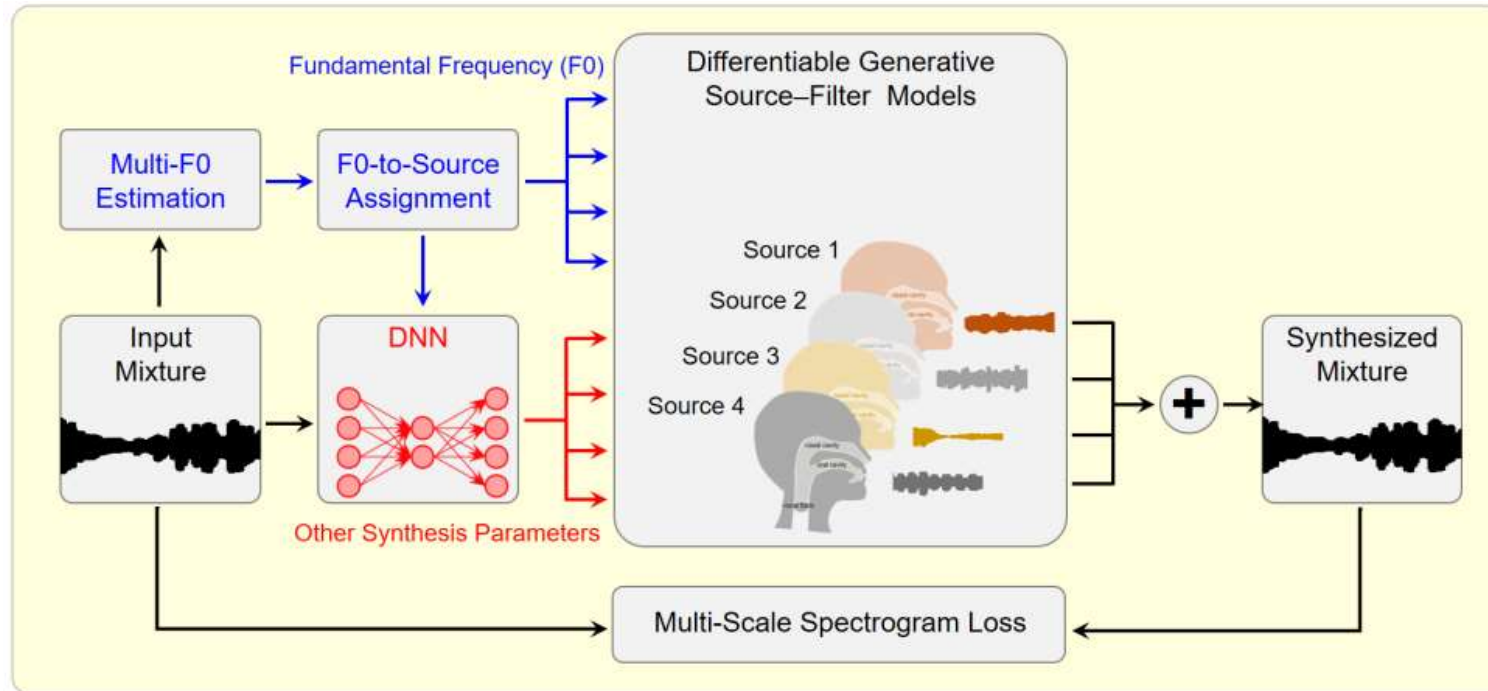
$$\text{With } \mathcal{L}_c = \|\mathbf{M}_c - \tilde{\mathbf{M}}_c\|_1 + \|\log(\mathbf{M}_c) - \log(\tilde{\mathbf{M}}_c)\|_1$$

and with $c = [2048, 1024, 512, 256, 128, 64]$ indicates the FFT size used to compute the STFT.



Towards Hybrid deep learning approaches: DDSP extensions and others...

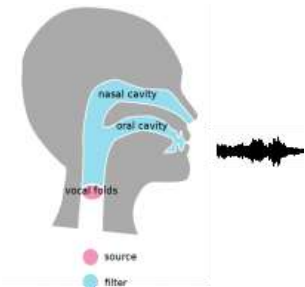
- An example for unsupervised singing voice separation



Towards Hybrid deep learning

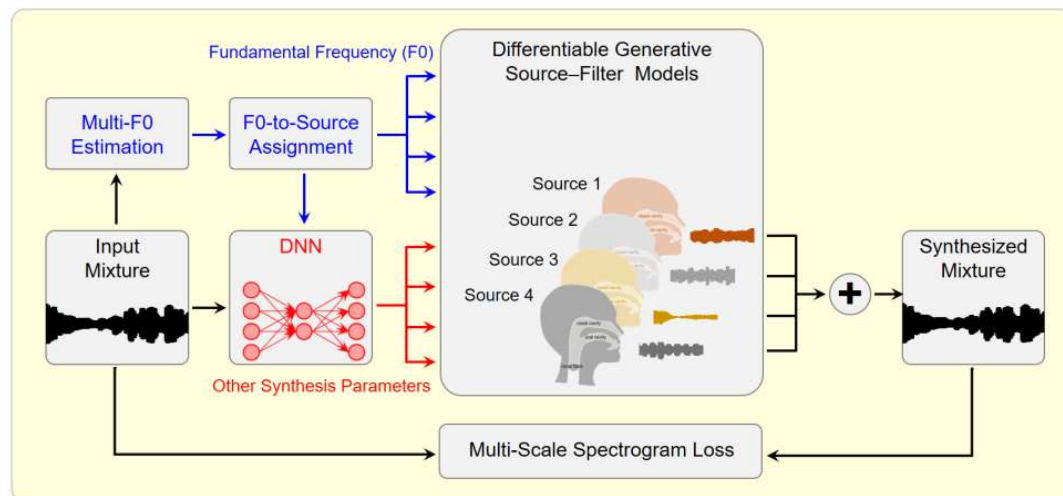
... by integrating our prior knowledge about the nature of the processed data.

Knowledge about « how the sound is produced » (e.g. sound production models)



Singing voice as a source / filter model :

- source = vibration of vocal folds
- Filter = resonances of vocal/nasal cavities



A new paradigm

- Model is at the « core » of neural architecture
- Source separation **by synthesis** (*no interference from other sources*)
- Learning only from the polyphonic recording (*no need of the true individual tracks*)


Novel sound transformation capabilities:

- Timbre/melody of the voice,
- Lyrics, translation
- Re-harmonization


Conclusion

- Generative AI goes beyond text generation...
- Generative Audio is gaining a strong interest and a variety of models and approaches are already proposed
- Note that I have not discussed the models for symbolic music (e.g. music scores as in MIDI)

Music score



MIDI representation (or piano roll)



Representation as sequence of tokens

```
NoteOn(50) TimeShift(9) NoteOn(60) NoteOn(65)
NoteOn(69) NoteOn(76) TimeShift(12) NoteOff(60)
NoteOff(65) NoteOff(69) NoteOff(76) TimeShift(3)
NoteOff(50) NoteOn(43) NoteOn(59) NoteOn(65)
NoteOn(69) NoteOn(76) TimeShift(24) NoteOff(All)
```

- ... which includes transformer models for symbolic music, « theme » transformer, Groove2Groove (style transfer), long context modelling (with specific positional encoding....)

Y.-J. Shih, S.-L. Wu, F. Zalkow, M. Müller, and Y.-H. Yang, "Theme Transformer: Symbolic music generation with theme-conditioned Transformer," *IEEE Transactions on Multimedia*, vol. 25, pp. 3495–3508, 2023.

O. Cifka, U. Simsekli, and G. Richard, "Groove2groove: One-shot music style transfer with supervision from synthetic data," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 28, pp. 2638–2650, 2020.

Manvi Agarwal, Changhong Wang, Gaël Richard, *Structure-Informed Positional Encoding For Music Generation*, Accepted for publication at ICASSP 2024.

