

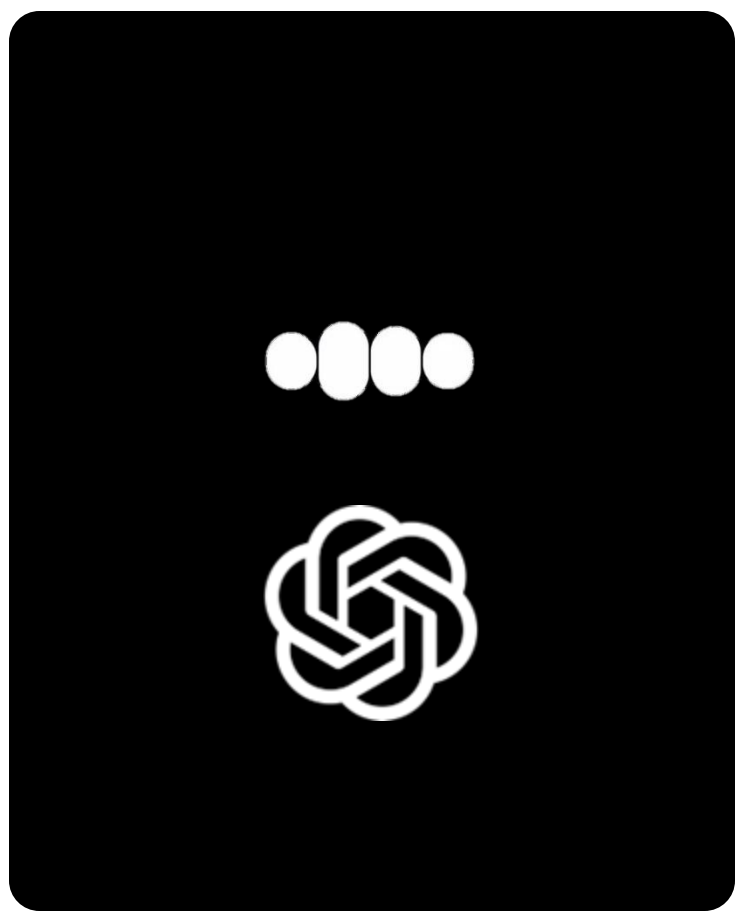
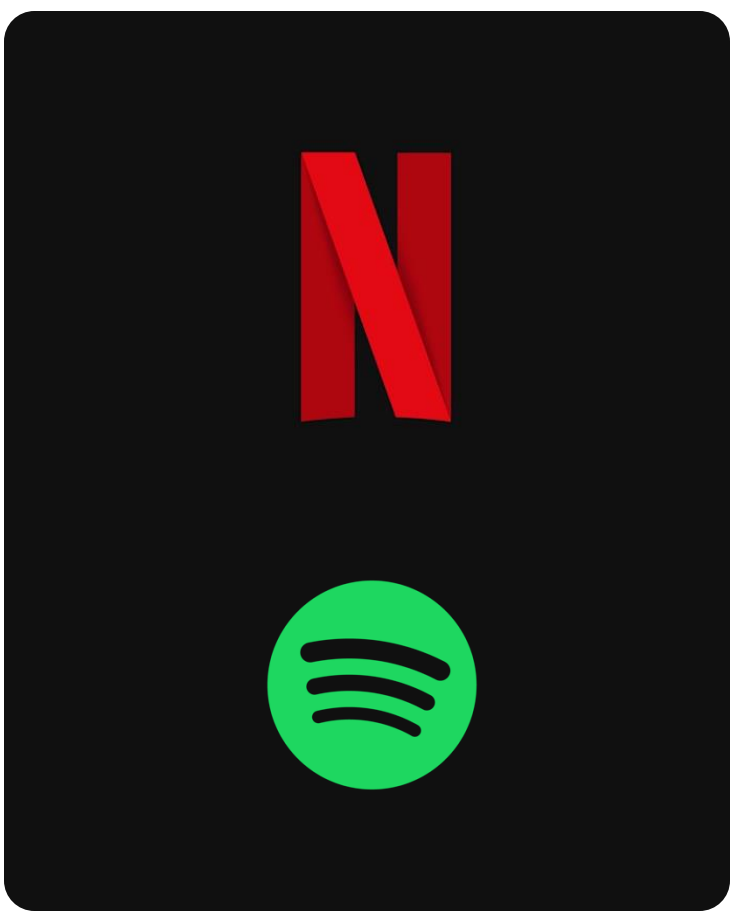


# High-Fidelity Audio Compression with Improved RVQGAN

Presented by: Nandor Kofarago



**ETH** zürich



# Audio sampling

**Human hearing**

20 Hz – 20 kHz



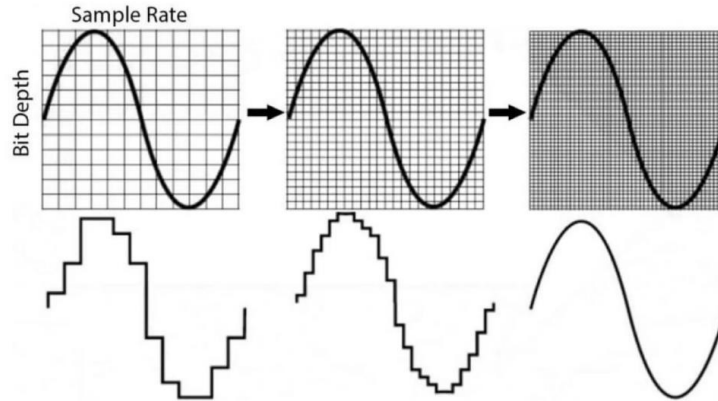
**Sample rate**

40 kHz



**Eliminate aliasing**

44.1 kHz



# Digital quantization

**Quantization**

16 bits



44.1 kHz x 16 bits x 2 channels

**Bitrate**

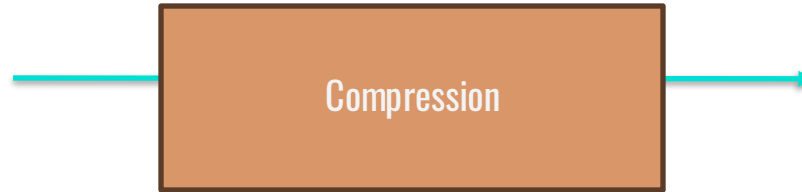
1411 kbps

Uncompressed audio: **630 MB / hour**

Tokenizing: **44100 tokens / s ?**

# Digital quantization

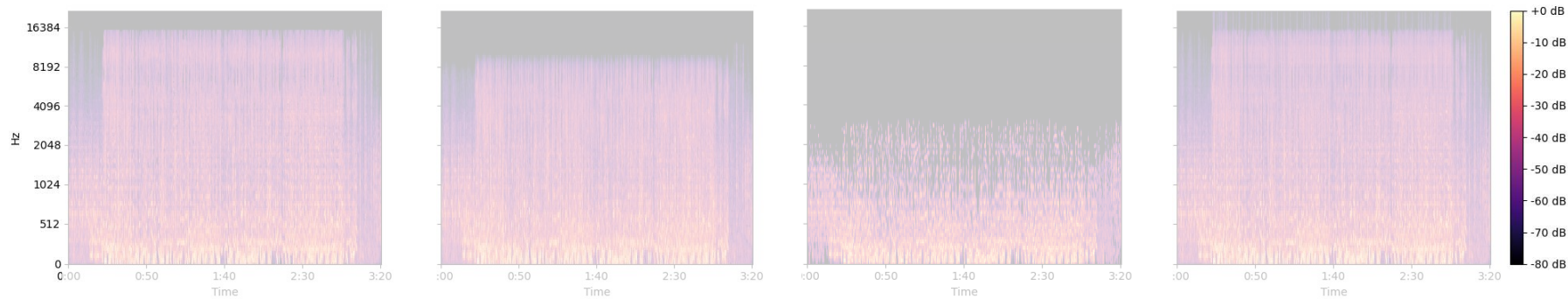
**Quantization**  
16 bits



**Bitrate**  
2 - 8 kbps

**Compressed audio: ~ 2 MB / hour**

# Demo



**Original**

**Opus – 8kbps**

**MP3 – 8 kbps**

**RVQGAN – 8 kbps**



# What do we need?

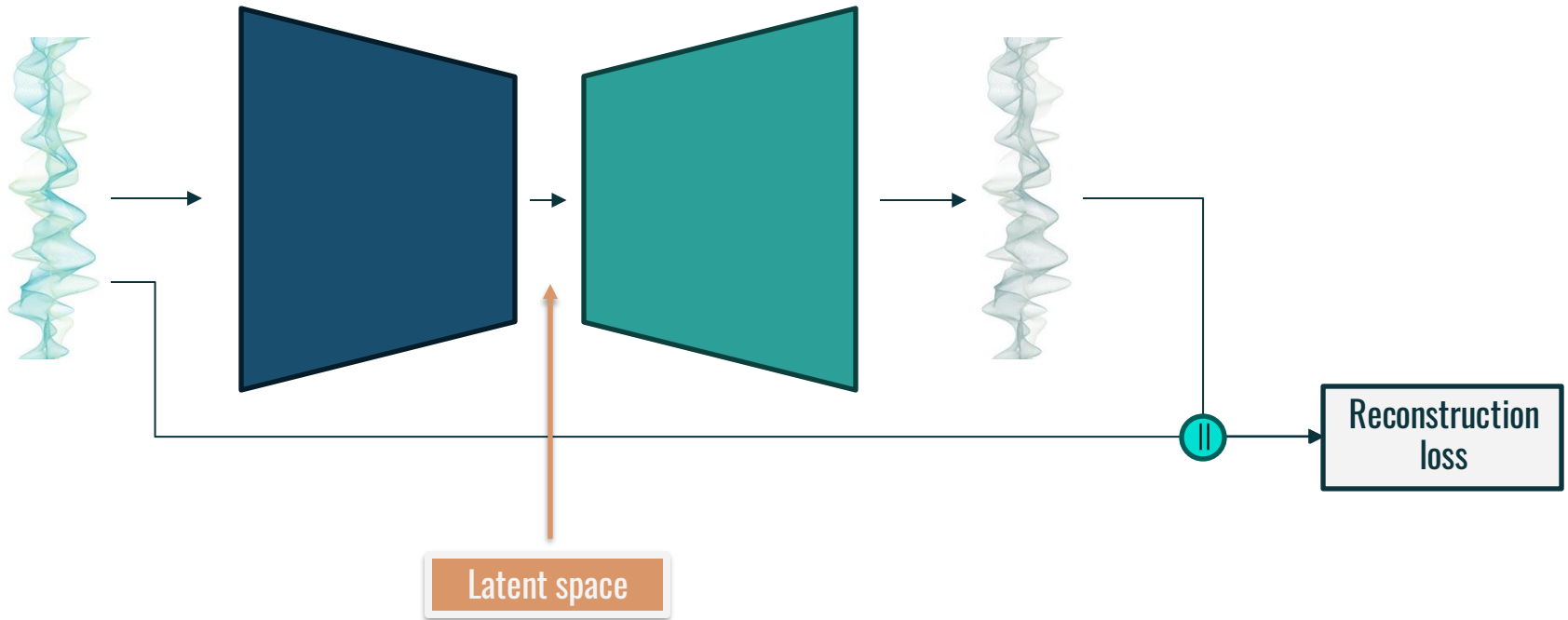
1. Efficient compression
2. Tokenizing audio
3. Generating audio



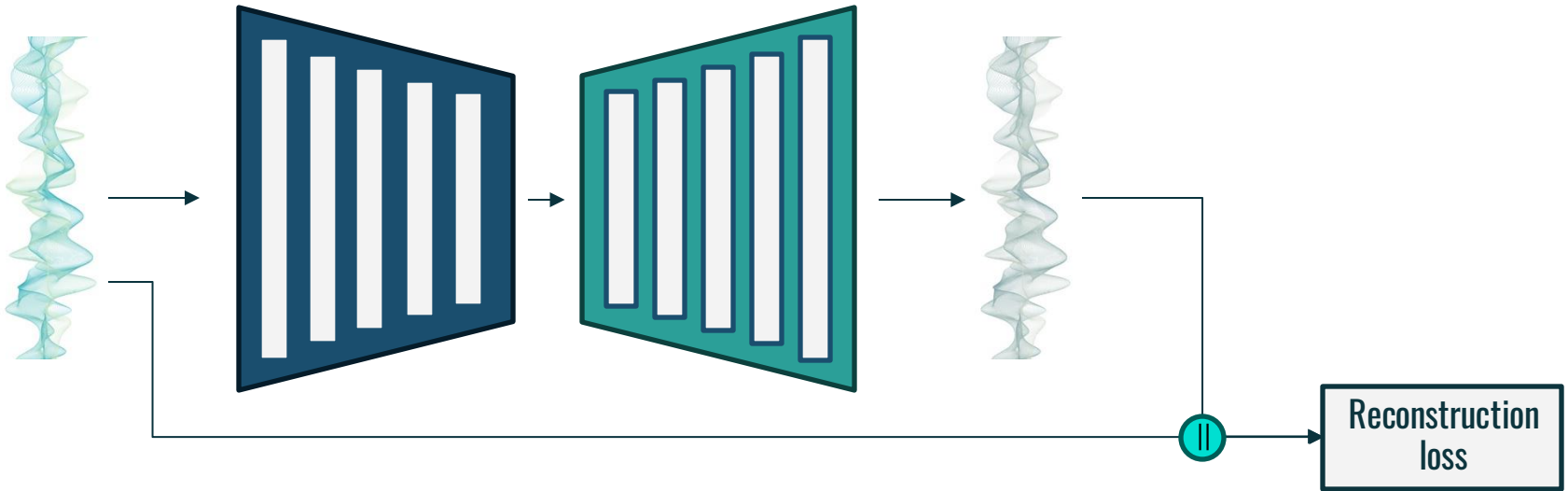
# Model architecture



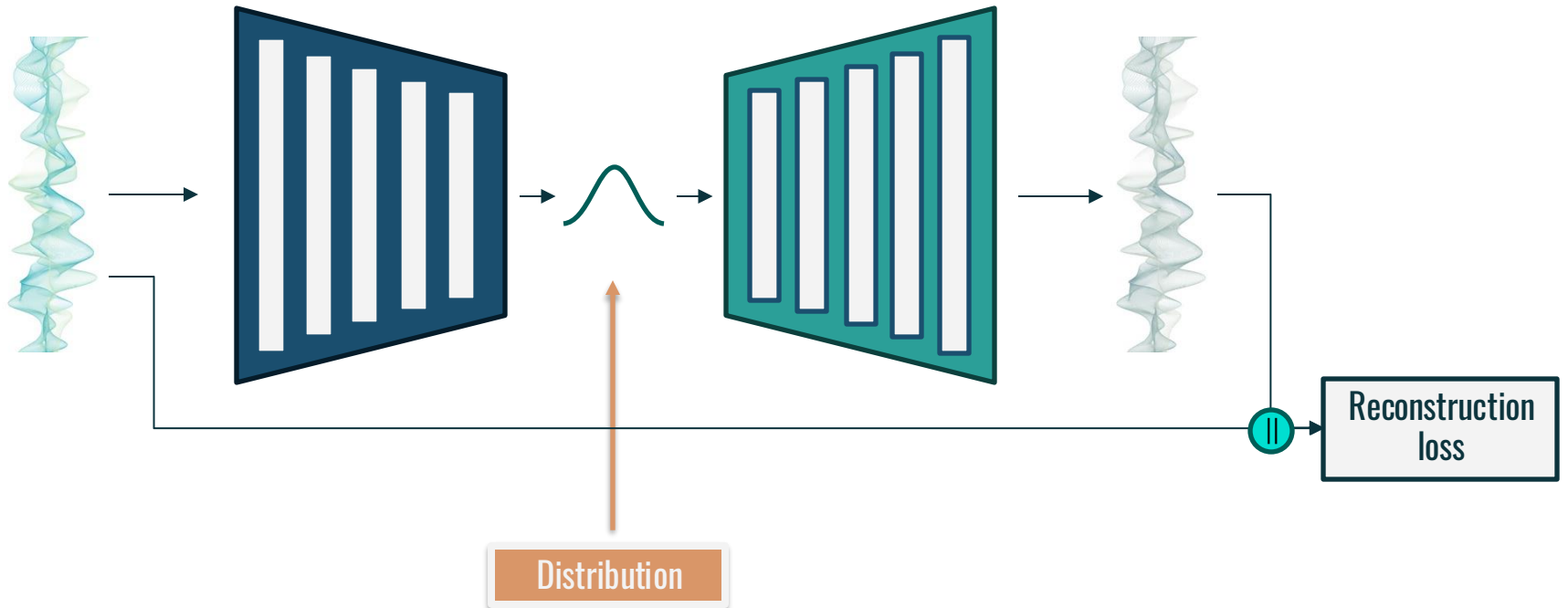
# Autoencoder



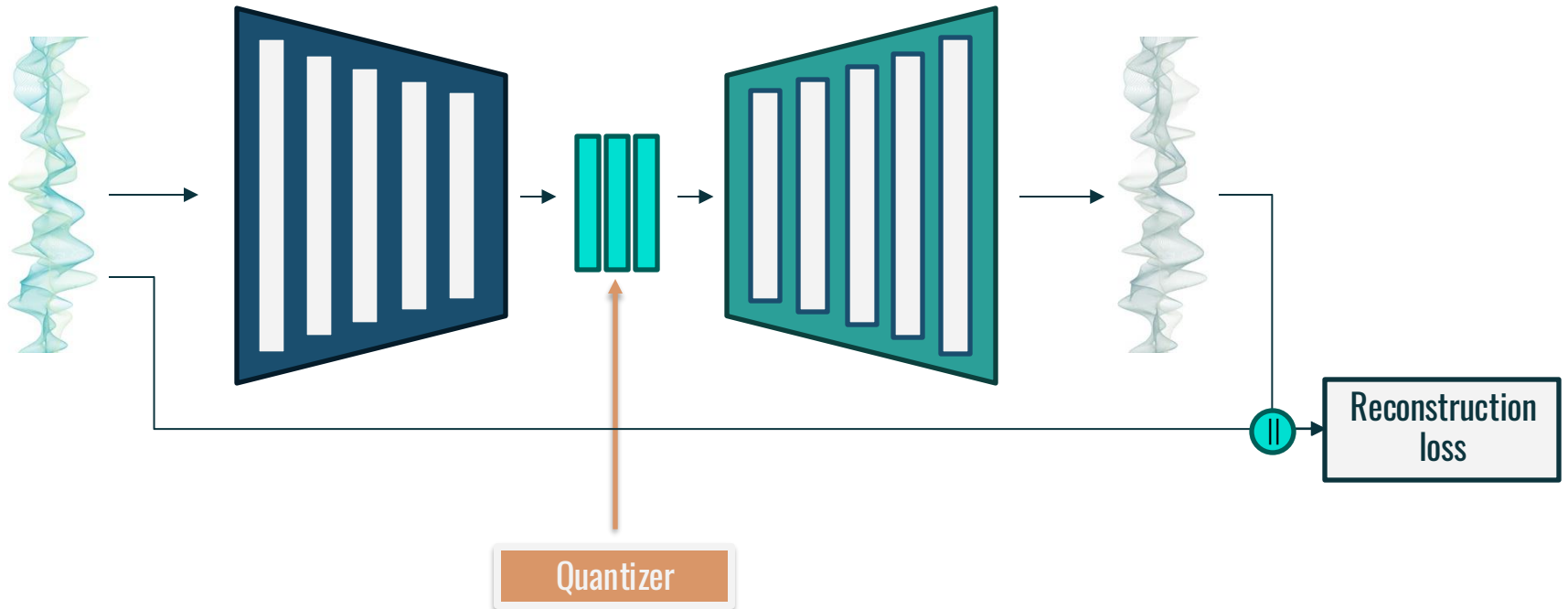
# Autoencoder



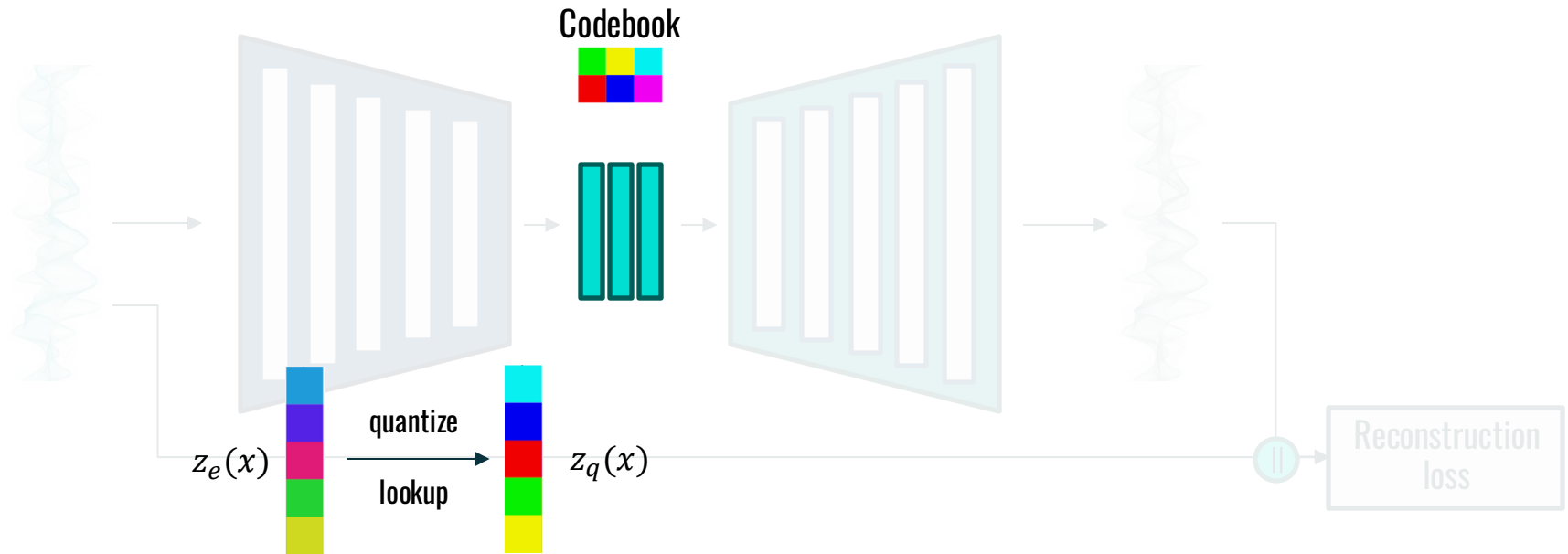
# Variational autoencoder



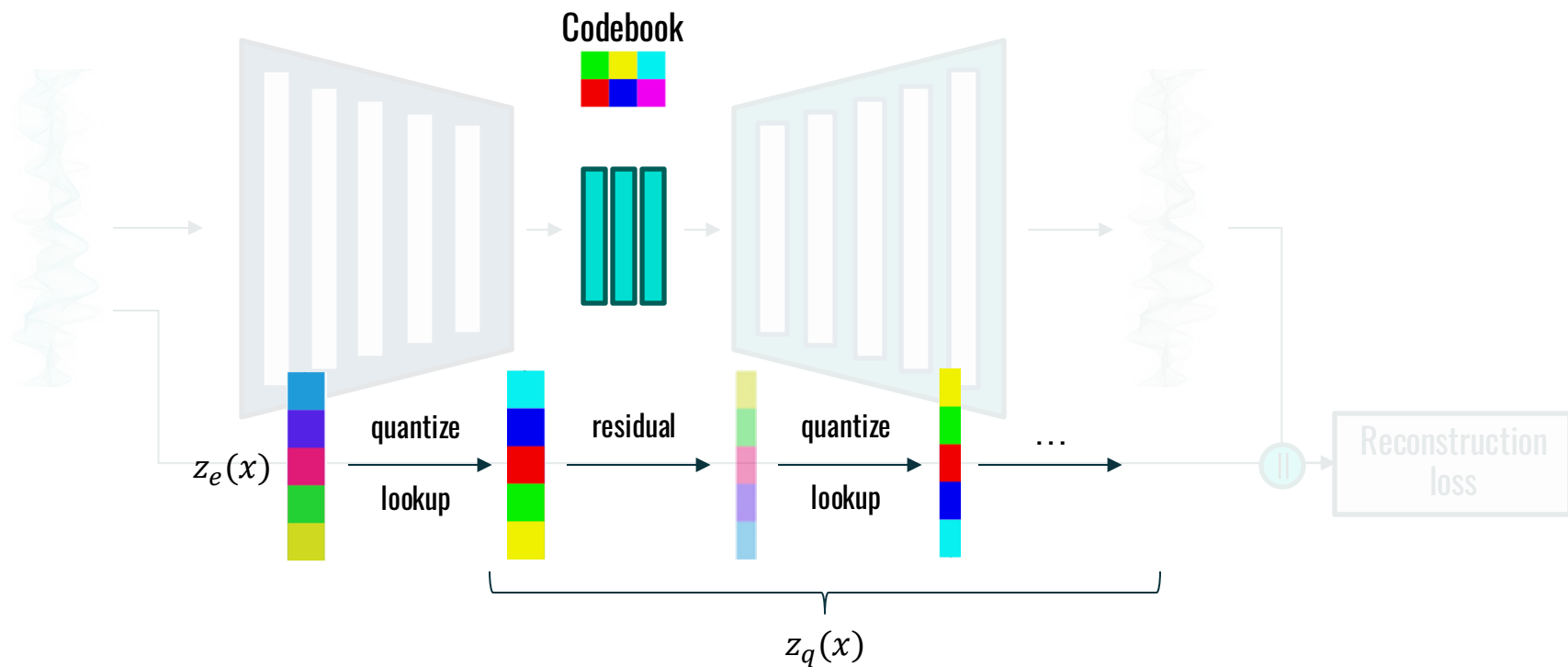
# Vector quantized VAE



# Vector quantized VAE



# Residual vector quantized VAE



# Bitrate of RVQVAE

Codebook size: 1024  $\rightarrow$  10 bits  
Number of quantizers: 1 – 9



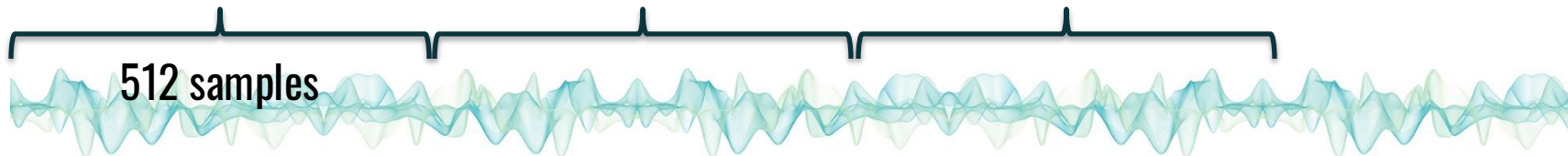
$\sim$ 86 windows / s  
90 bit / window



$\sim$  8 kbps



...

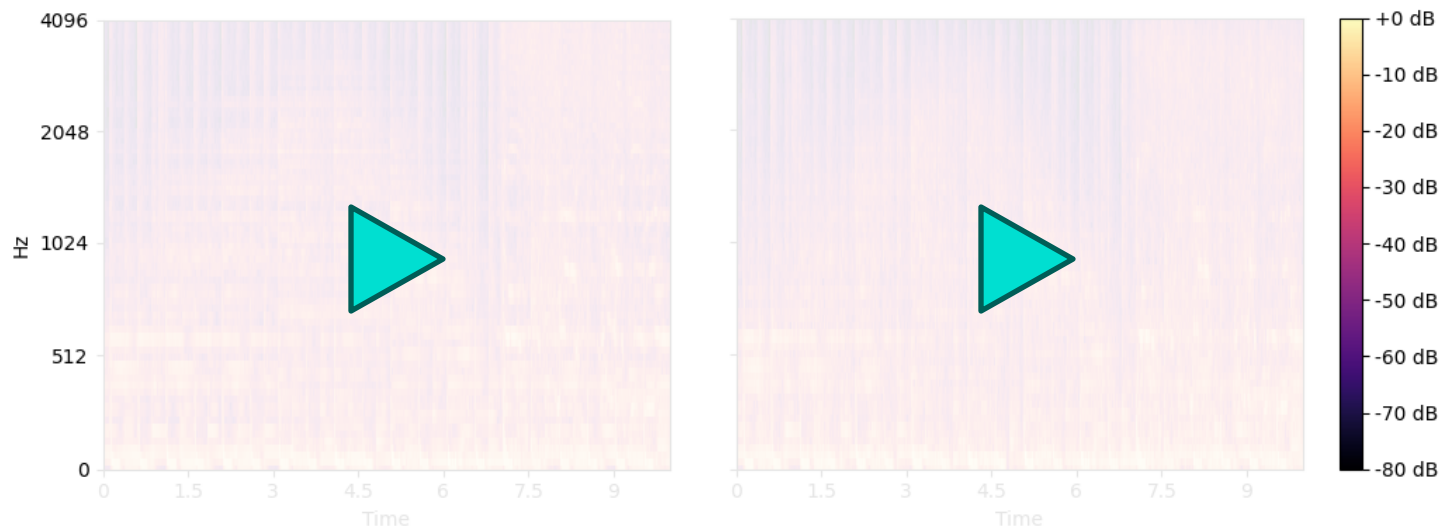


# Residual vector quantized VAE

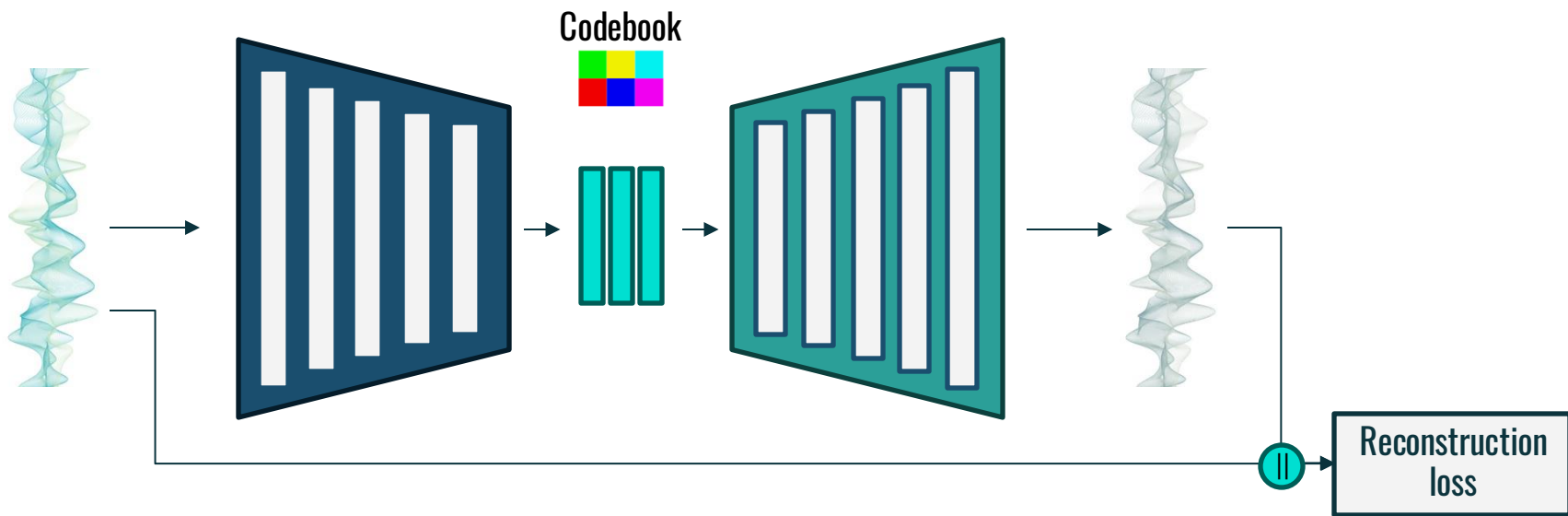




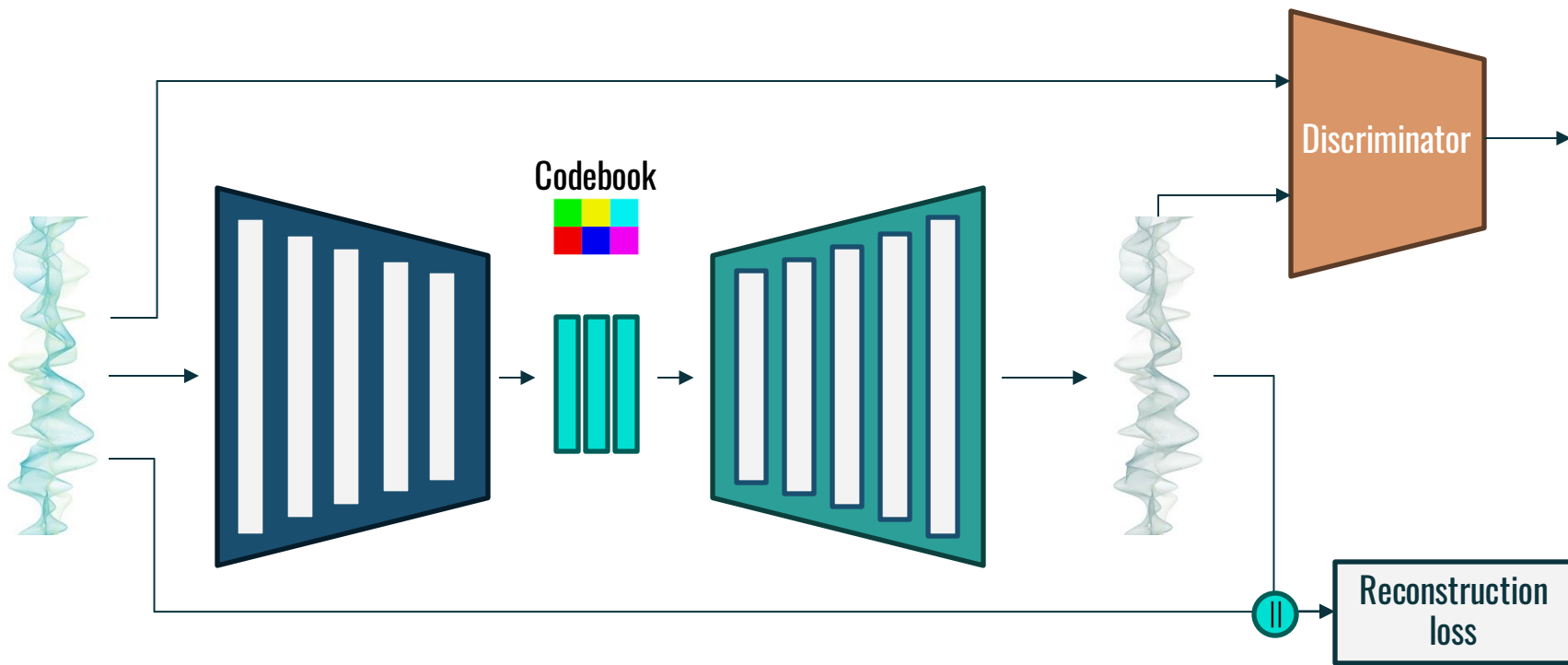
# Problem with simple loss



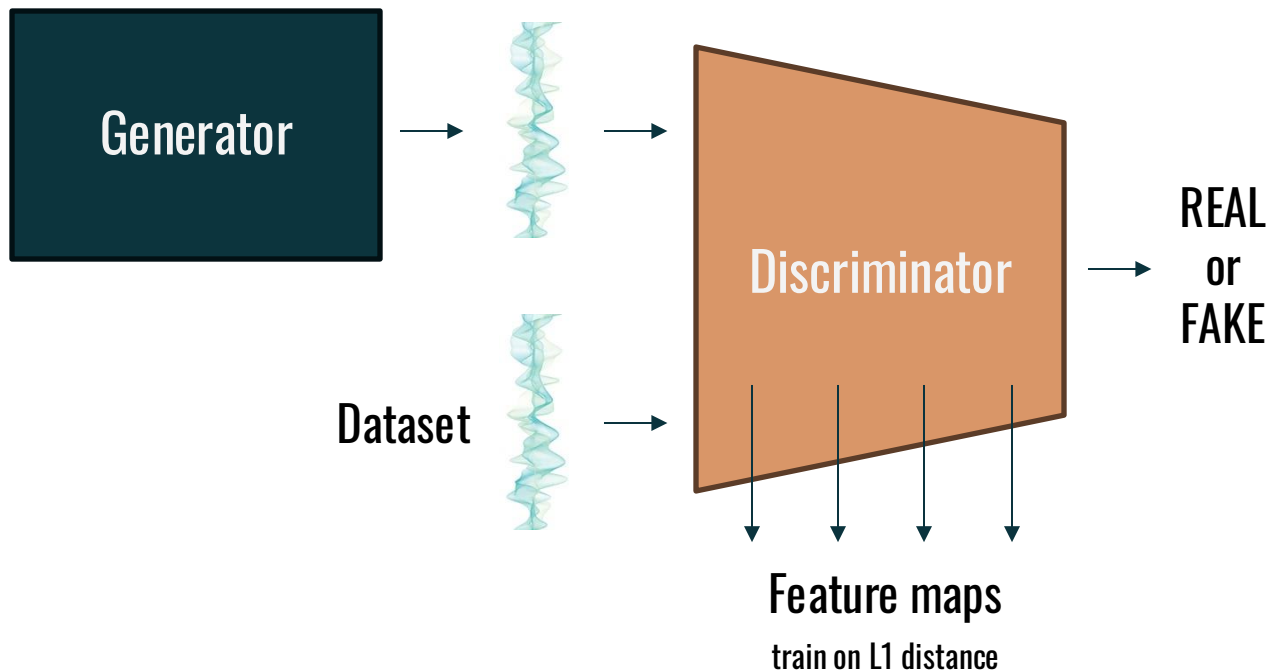
# Residual vector quantized VAE



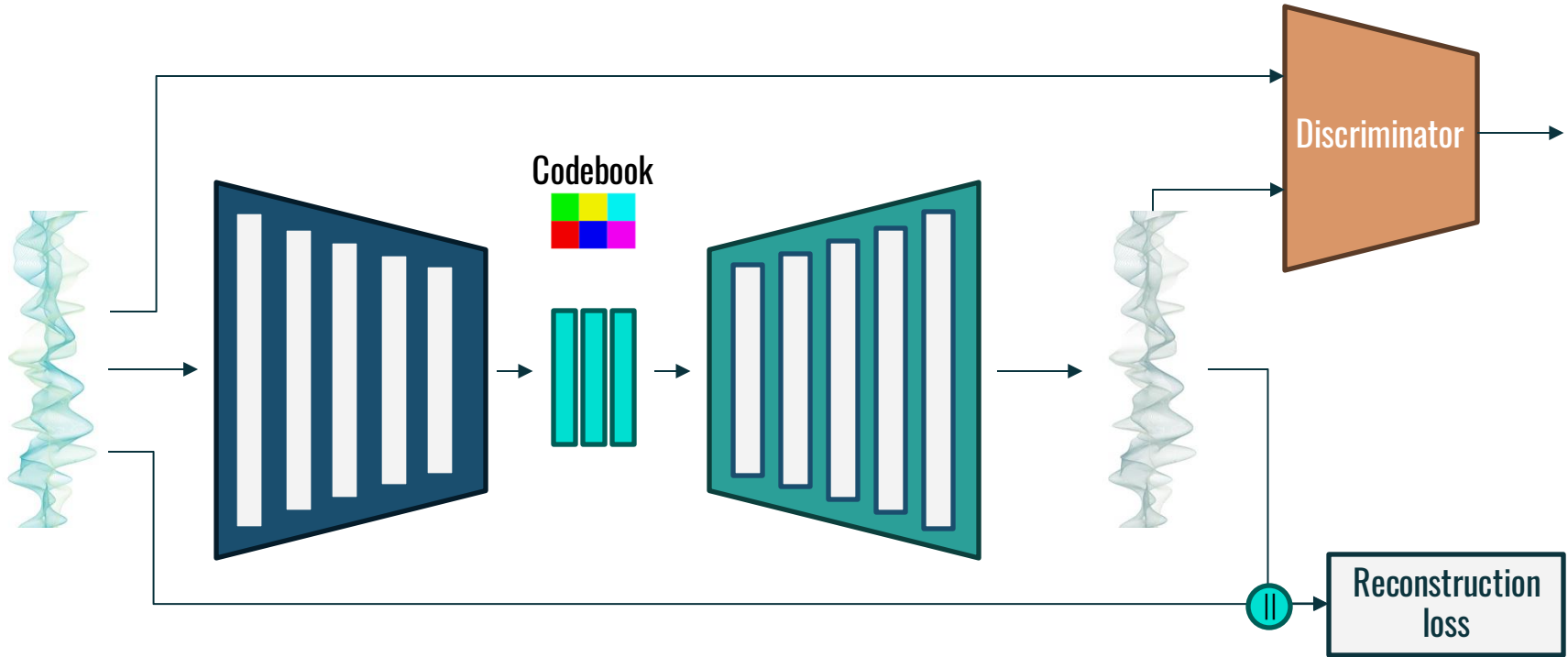
# Residual vector quantized VAE



# Generative adversarial networks



# EnCodec



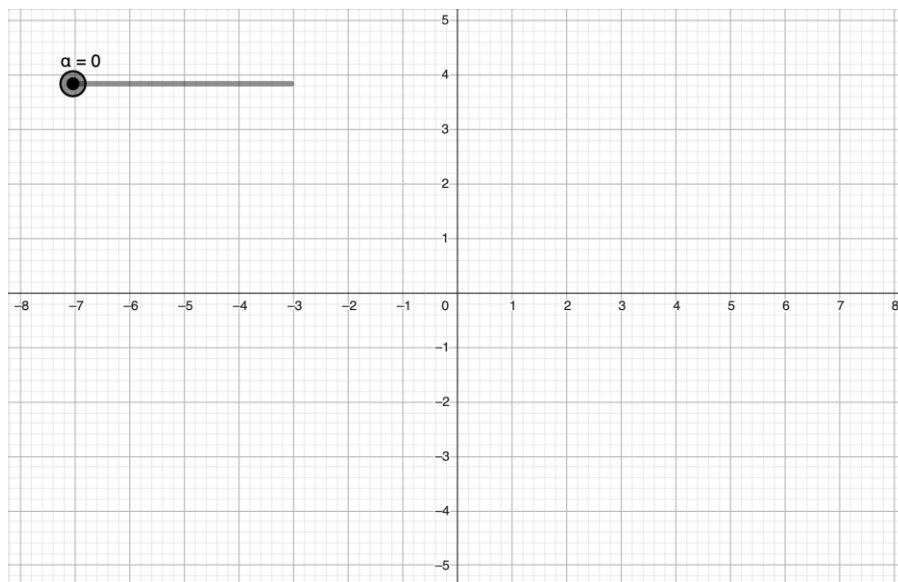


# Improved RVQGAN

# 1. Periodic activation function

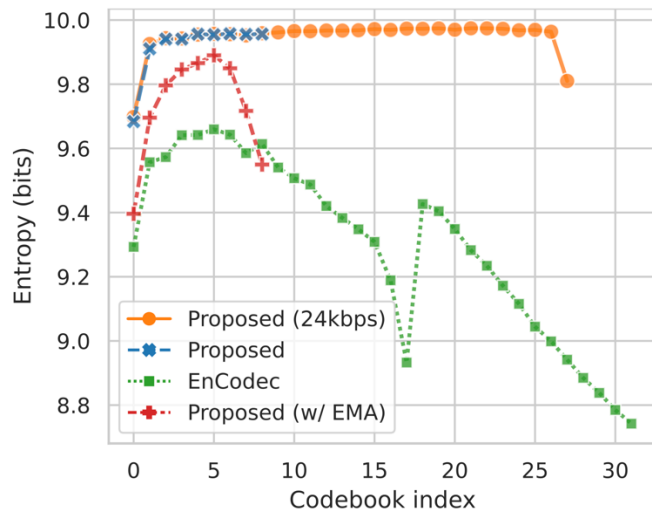
Snake activation function:

$$\text{snake}(x) = x + \frac{1}{\alpha} \sin^2(\alpha x)$$



# 2. Improved residual vector quantization

## Low codebook utilization



→ Inefficient encoding  
Lower quality reconstructions



## 2. Improved residual vector quantization



K-means clustering to initialize codebook

Randomized restart for underutilization

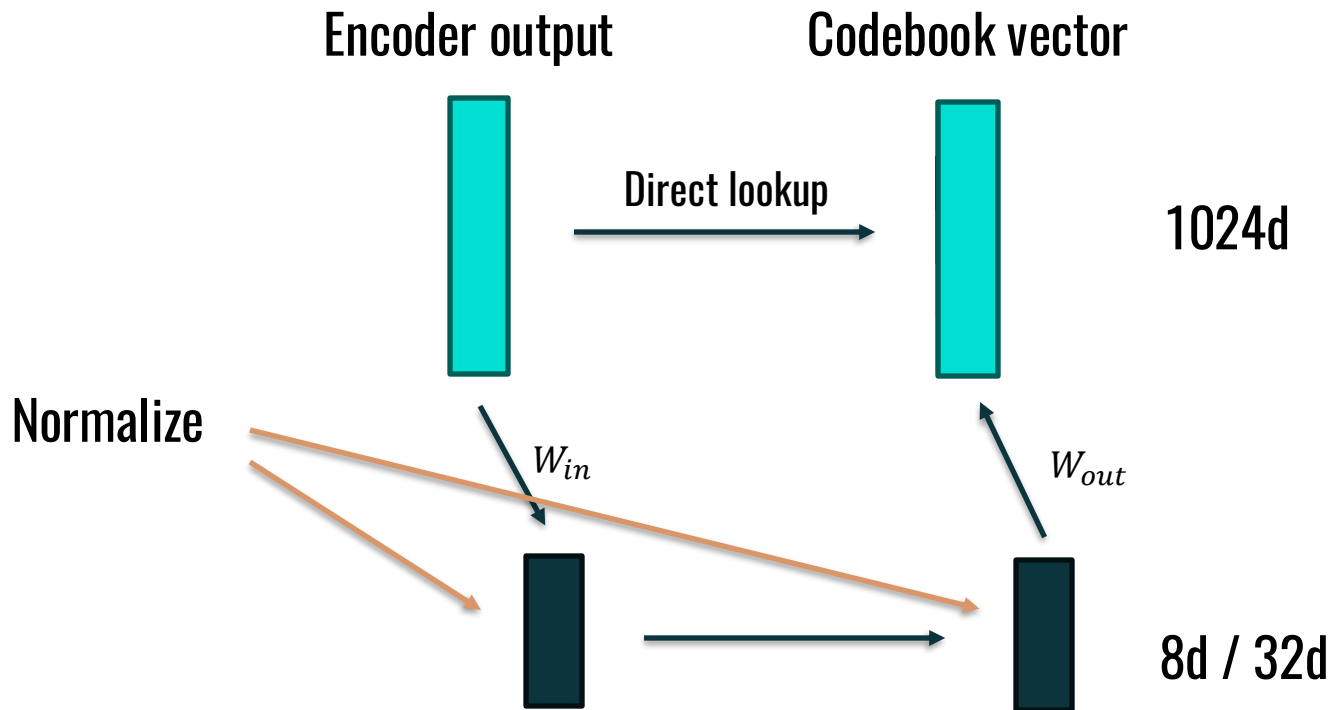
Exponential moving average (EMA)



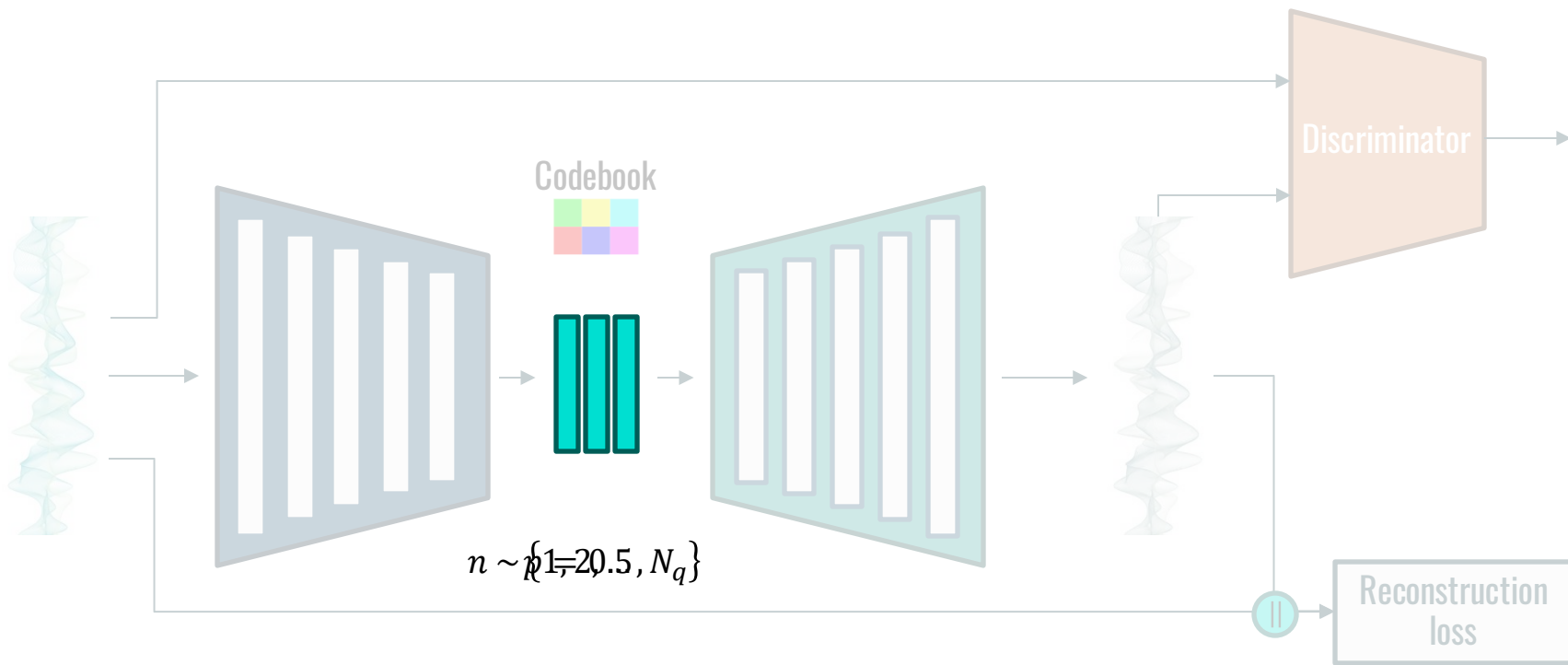
Factorized codes

L2-normalized codes

## 2. Improved residual vector quantization



# 3. Quantizer dropout rate





# 4. Discriminator design

1. Multi-scale discriminator (MSD) -> waveform
2. Multi-period discriminator (MPD) -> waveform
3. Complex short-time Fourier transform (STFT) discriminator at multiple time-scales -> frequency

# Multi-scale discriminator



# Multi-period waveform discriminator



# Complex STFT at multiple time-scales



Fourier transform

Real part: **frequency**

Imaginary part: **phase**

# 5. Loss functions

## Reconstruction

- Mel-reconstruction loss
- Multi-scale spectral losses

## Codebook learning

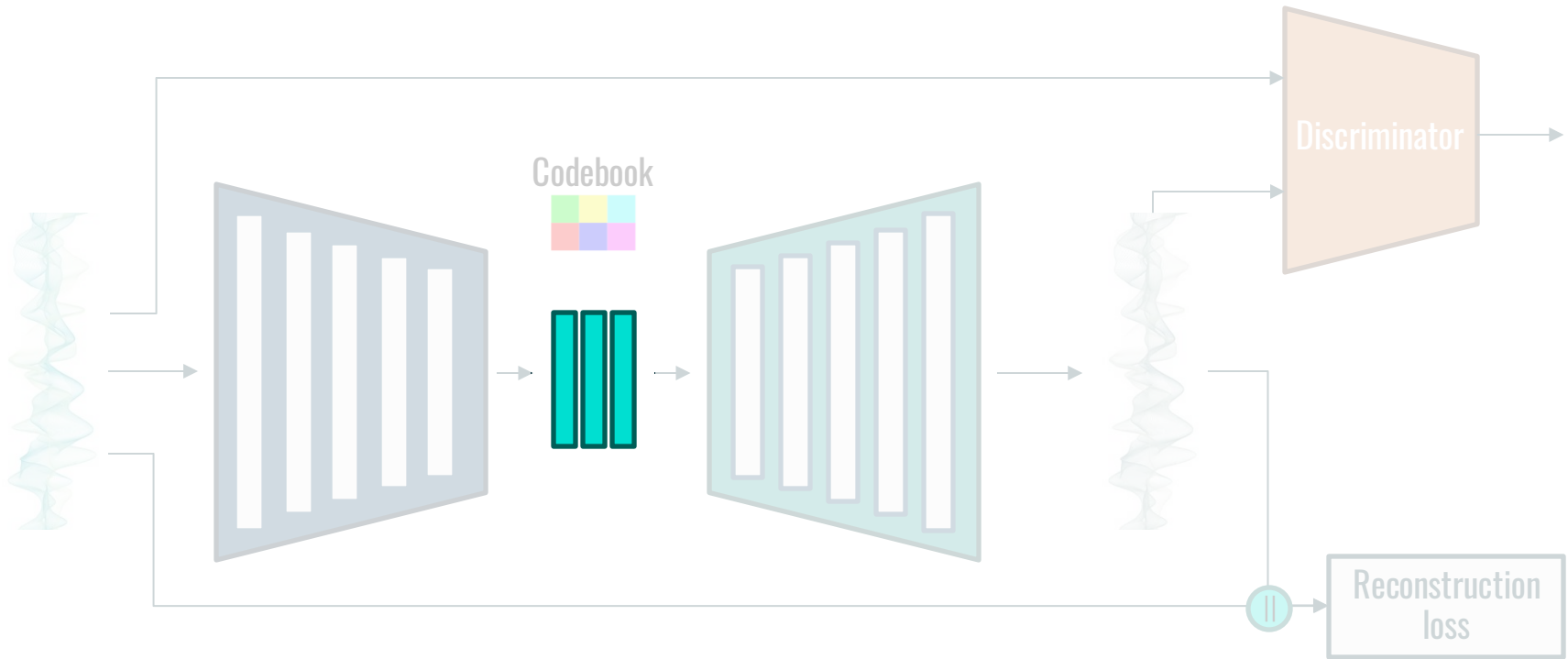
- Codebook loss
- Commitment loss

## Adversarial

- Multi-scale discriminator
- Multi-period discriminator
- Multi-band multi-scale STFT discriminator



# RVQGAN





# Training



# Training data

- **Speech, music, and environmental sounds**
- **Balanced data sampling (full-band)**



# Experiments



# Ablation study

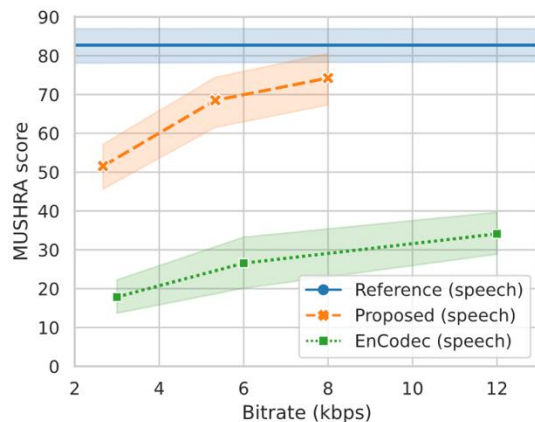
- Discriminators
- Mel reconstruction loss
- Latent dimension of codebook
- Quantization setup
- Balanced data sampling

# Objective metrics

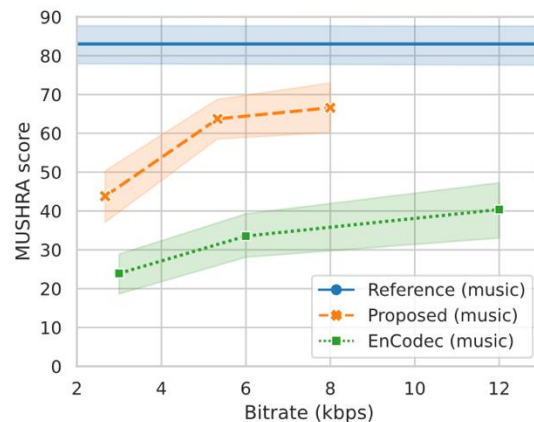
Codec	Bitrate (kbps)	Bandwidth (kHz)	MeI distance ↓	STFT distance ↓	ViSQOL ↑	SI-SDR ↑
Proposed	1.78	22.05	1.39	1.95	3.76	2.16
	2.67	22.05	1.28	1.85	3.90	4.41
	5.33	22.05	1.07	1.69	4.09	8.13
	8	22.05	0.93	1.60	4.18	10.75
EnCodec	1.5	12	2.11	4.30	2.82	-0.02
	3	12	1.97	4.19	2.94	2.94
	6	12	1.83	4.10	3.05	5.99
	12	12	1.70	4.02	3.13	8.36
	24	12	1.61	3.97	3.16	9.59
Lyra	9.2	8	2.71	4.86	2.19	-14.52
Opus	8	4	3.60	5.72	2.06	5.68
	14	16	1.23	2.14	4.02	8.02
	24	16	0.88	1.90	4.15	11.65

# Subjective metrics

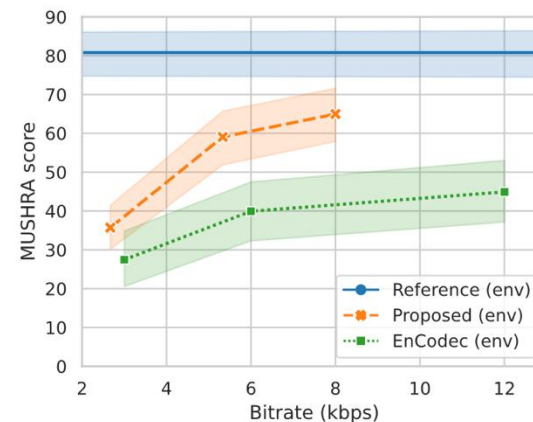
## Speech



## Music



## Environmental s.



**MUSHRA = MULTIPLE Stimuli with Hidden Reference and Anchor**



# Opinion





# Opinion

- + Impressive results, very clearly presented
- + Clean codebase, 1-line usage from command line
- + Focus on new applications (encoding and audio generation)
- Lacking speed test (is it real-time?)
- Streamability
- Hard to compare sampling rates
- EnCodec uses entropy coding -> low codebook utilization is OK
- Reviewers criticize novelty (OpenReview)



# Discussion time

CREDITS: This presentation template was created by **Slidesgo**, including icons by **Flaticon**, and infographics & images by **Freepik**



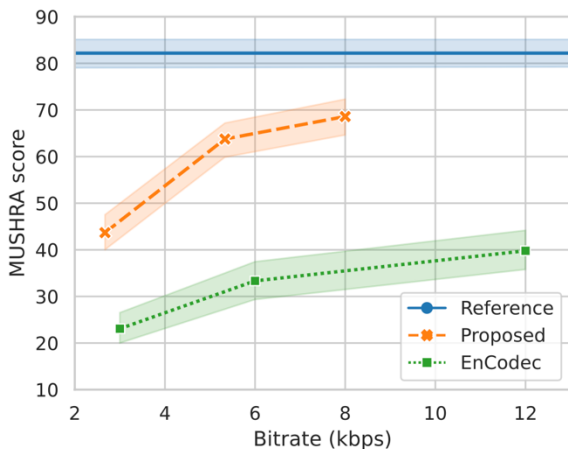


# Evaluation metrics

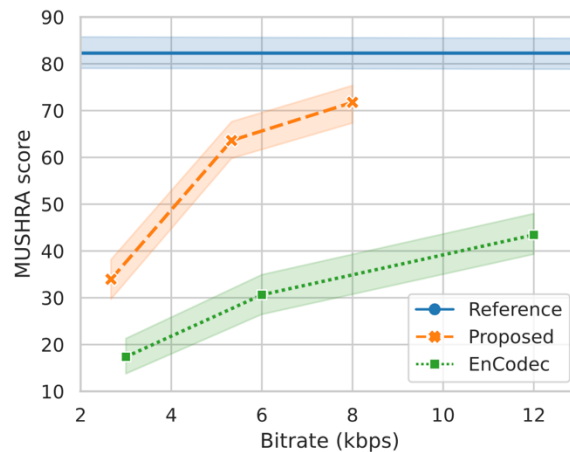
- 1. Mel Distance**
- 2. STFT Distance**
- 3. ViSQOL (Virtual Speech Quality Objective Listener)**  
-> deep learning model trained on human hearing data to predict Mean Opinion Score
- 4. SI-SDR (Scale-Invariant Signal-to-Distortion Ratio)**  
-> similar to signal-to-noise ratio, with modifications so that it is invariant to scale differences, indicates the quality of the phase reconstruction of the audio

# Sample rate comparisons

44 kHz



24 kHz



# Loss function

$$\mathcal{L}_{VQ} = \|\text{sg}[\ell_2(z_{\text{proj}}(x))] - \ell_2(e_k)\|_2^2 + \beta \|\ell_2(z_{\text{proj}}(x)) - \text{sg}[\ell_2(e_k)]\|_2^2$$

**Reconstruction loss (multi-scale mel, multi-scale spectral): 15.0**

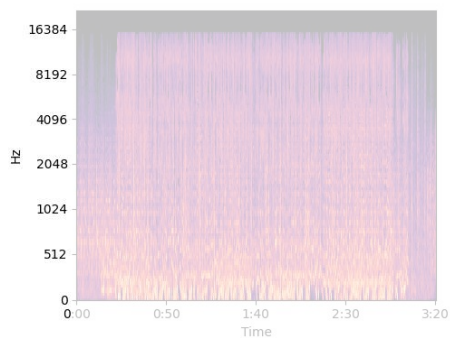
**Feature matching loss: 2.0**

**Adversarial loss: 1.0**

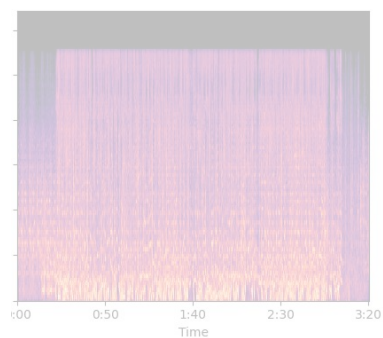
**Codebook loss: 1.0**

**Commitment losses: 0.25**

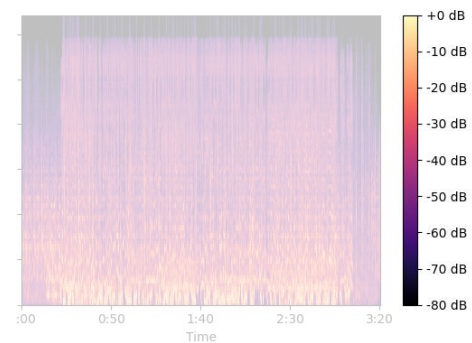
# EnCodec demo



**Original**



**EnCodec – 12 kbps**



**RVQGAN – 8 kbps**