

# Timbre Latent Space: Exploration and Creative aspects

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## Models

We propose two models based on the Variational Auto-Encoder (VAE) [1], which learn a latent representation of an audio dataset by jointly optimizing two functions used to analyse (encoding) and synthesize (decoding) audio.

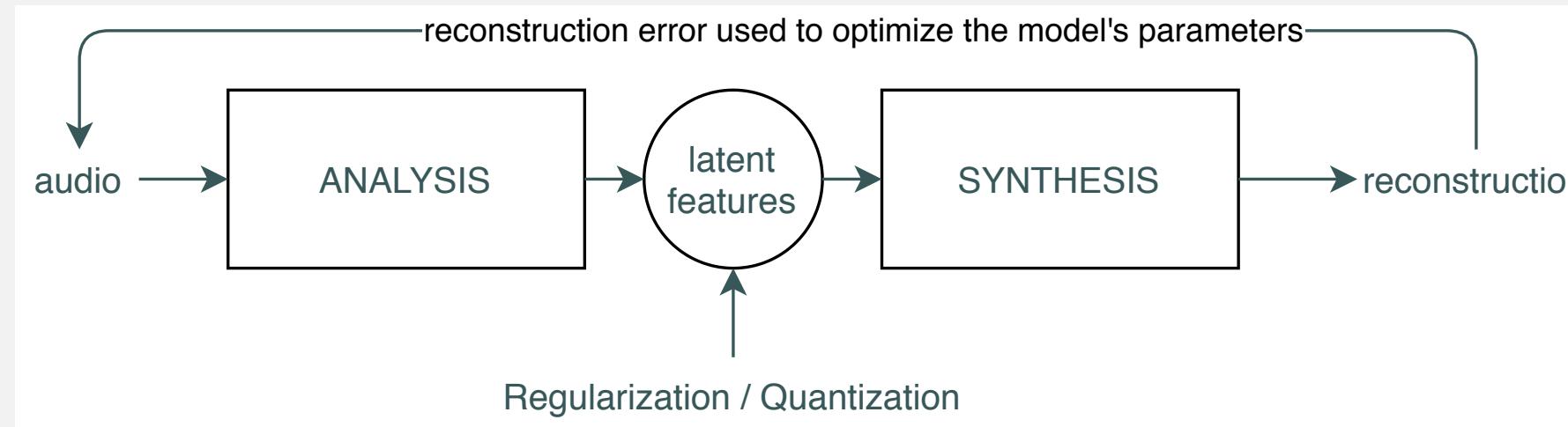


Figure: Overall architecture shared by both models

This invertible representation is generally of lower dimensionality than audio, but its use as a synthesis tool in a creative process remains complicated. In this work we explore interactions either based on a continuous latent representation or a discrete set of latent features.

## References

- [1] Diederik P. Kingma et al. "Auto-Encoding Variational Bayes". In: *2nd International Conference on Learning Representations*. 2014.
- [2] Kundan Kumar et al. "MelGAN: Generative Adversarial Networks for Conditional Waveform Synthesis". In: *Advances in Neural Information Processing Systems* 32. 2019.
- [3] Yaroslav Ganin et al. "Unsupervised Domain Adaptation by Back-propagation". In: vol. 37. *Proceedings of Machine Learning Research*. 2015.
- [4] Aaron van den Oord et al. "Neural Discrete Representation Learning". In: *Advances in Neural Information Processing Systems* 30. 2017.

## Contact Information

- Web: [https://acids-ircam.github.io/timbre\\_exploration/](https://acids-ircam.github.io/timbre_exploration/)
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## Continuous latent space

The continuous model is composed of two main blocks: a mel-spectrogram VAE and a mel-spectrogram to waveform model [2]. We train the VAE using an objective composed of a **reconstruction loss** and a **regularization loss**, itself being the addition of a prior regularization and a domain adaptation loss [3]. The obtained regularization loss ensures that the latent space is smooth and loudness invariant.

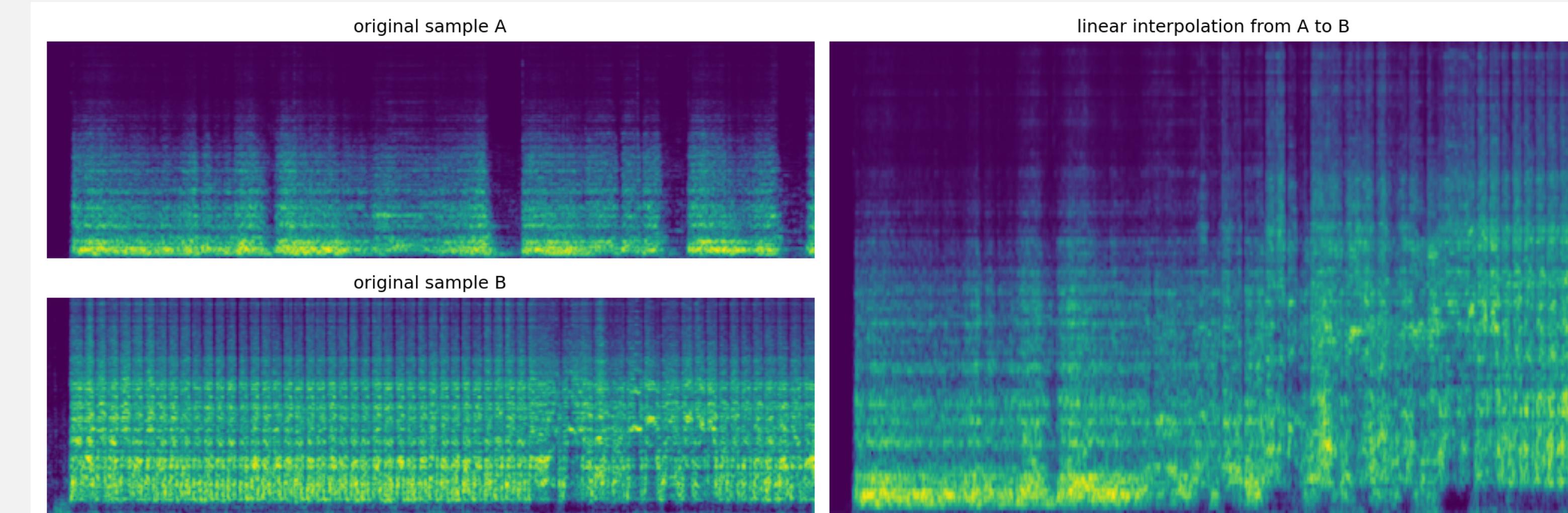


Figure: Time linear interpolation between two audio samples

## Discrete latent space

The discrete model is based on a Vector-Quantized VAE [4] for frame-wise processing of raw waveform. Each signal window is analysed and quantized with the nearest latent vector, also invariant to audio levels. Once trained, we can analyse each individual latent feature and compute some corresponding acoustic descriptor values. This provides a mapping that allows direct **descriptor-based** synthesis, by matching a given descriptor target with the series of nearest latent features.

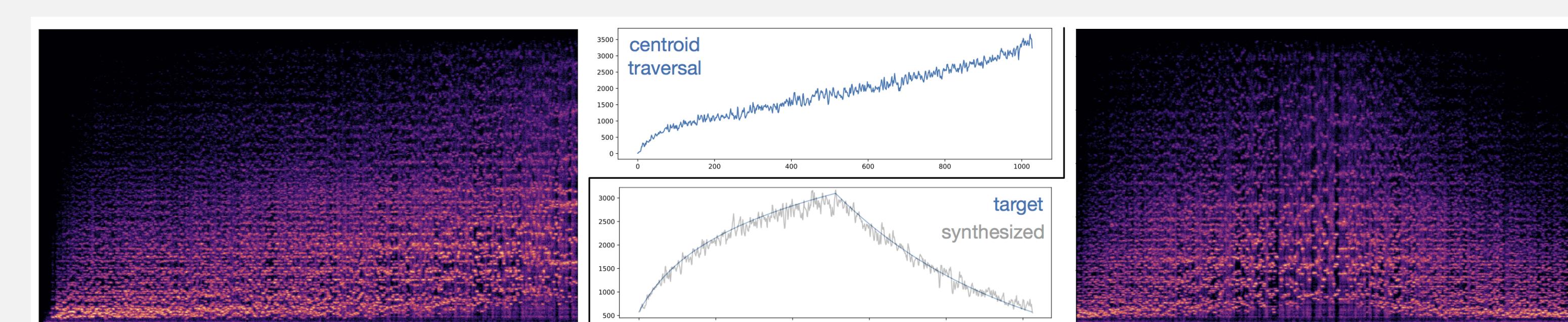


Figure: Left: traversal of the discrete representation in the increasing order of the spectral centroid. Right: Example of descriptor-based synthesis.

## Offline generation

Max/MSP interface designed to help the process of encoding and decoding audio. We added several tools like **manual deformation of latent series** and an **interpolation plane**.

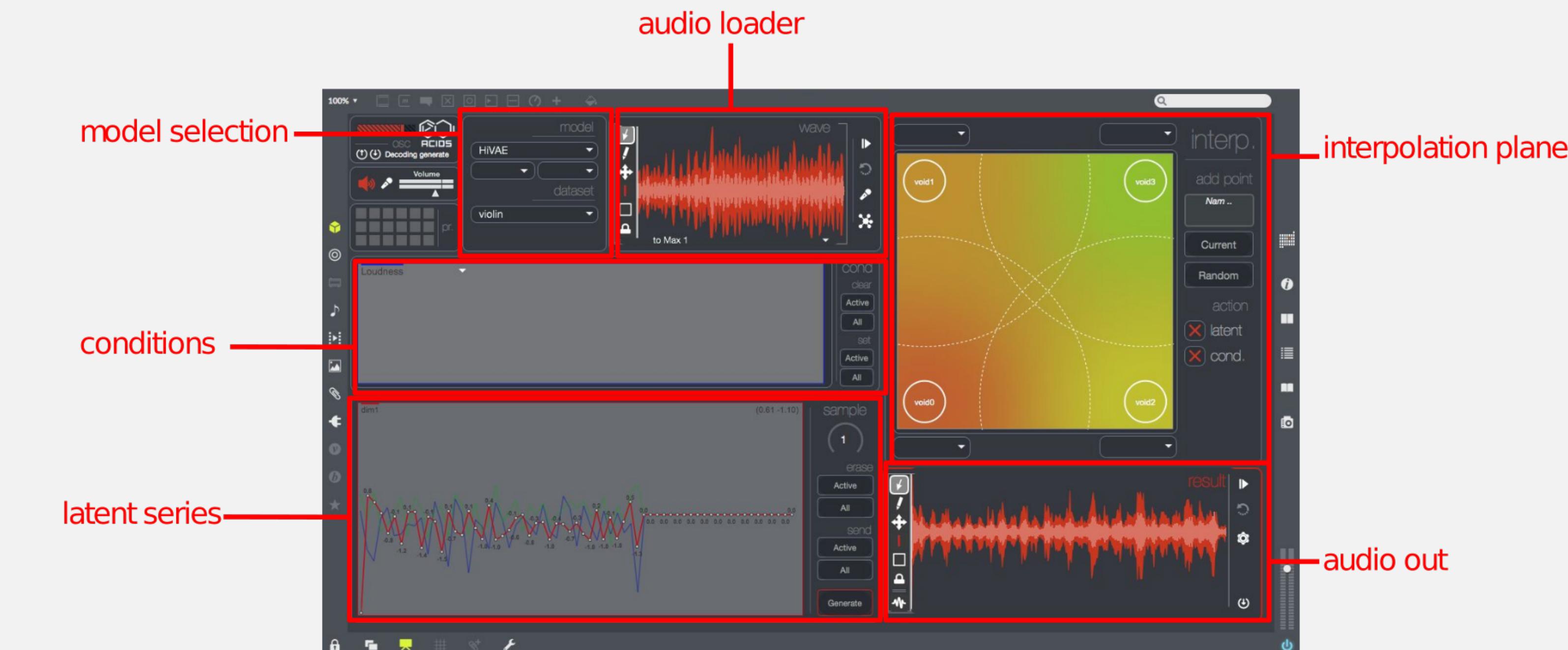


Figure: Max / MSP interface for offline generation

## Online generation

In order to allow a **realtime interaction** with the model, we abstracted the encoder-decoder pair as PureData signal objects, allowing their use inside a complex composition workflow.

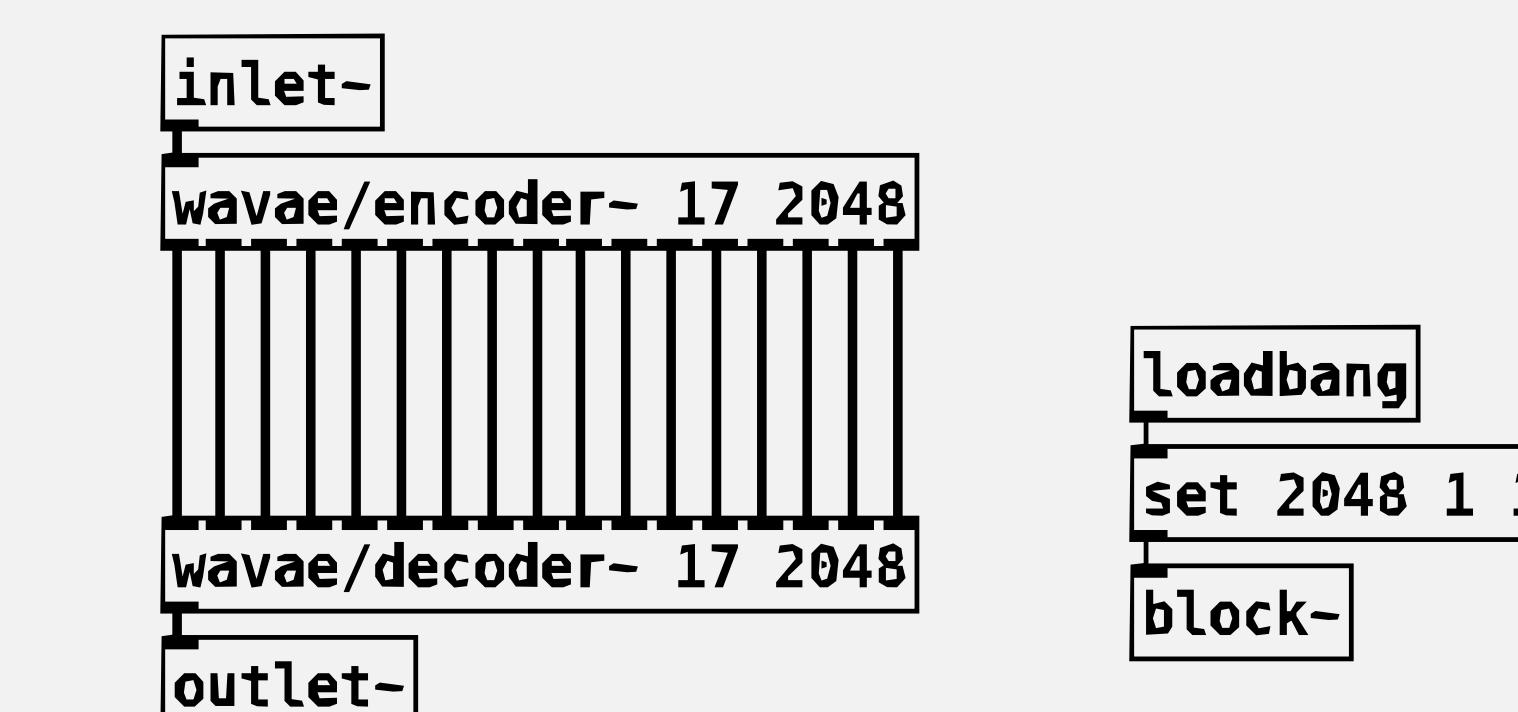


Figure: PureData encoder / decoder objects



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