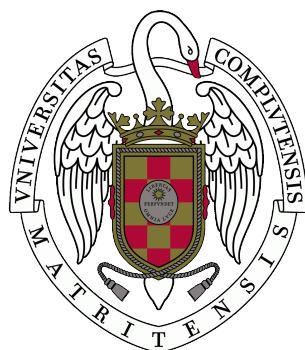

**Aprendizaje de representaciones latentes de
audio musical con Autocodificadores y
Autocodificadores Variacionales**
**Learning latent representations of musical audio
with Autoencoders and Variational
Autoencoders**



Trabajo de Fin de Grado
Curso 2024–2025

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Grado en Ingeniería Informática
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Resumen

Aprendizaje de representaciones latentes de audio musical con Autocodificadores y Autocodificadores Variacionales

La aplicación del aprendizaje profundo a la representación y generación de audio musical se investiga en este trabajo de fin de grado, considerando dos arquitecturas de aprendizaje profundo: autocodificadores (AEs) y autocodificadores variacionales (VAEs). En contraste con los métodos simbólicos convencionales que generan música en representaciones de alto nivel como MIDI, esta tesis adopta un enfoque no simbólico y trabaja directamente sobre representaciones crudas o de tiempo-frecuencia del audio. Tras una revisión de la historia de la composición algorítmica y de varios métodos de síntesis de audio, se utilizan autocodificadores convolucionales en este trabajo para comprimir y reconstruir notas musicales, y autoencoders variacionales para sintetizar audio mediante el muestreo dentro de representaciones ocultas aprendidas.

A lo largo del proyecto, se investigan diversas arquitecturas de modelos y métodos de entrenamiento con el objetivo de mejorar la calidad de reconstrucción generación. Aunque existe margen de mejora en la generación de sonido musical, los resultados demuestran el potencial de la aplicación del aprendizaje profundo en el campo musical.

Palabras clave

Aprendizaje profundo, Autocodificadores, Autocodificadores variacionales, Espacio latente, Espectrograma, Red Convolucional.

Abstract

Learning latent representations of musical audio with Autoencoders and Variational Autoencoders

The application of deep learning to audio representation and generation of music is investigated in this thesis, considering two deep learning architectures: autoencoders (AEs) and variational autoencoders (VAEs). In contrast to conventional symbolic methods that produce music in higher-level representations like MIDI, this thesis takes a non-symbolic approach and works directly on raw or time-frequency representations of audio. Following an overview of the history of algorithmic composition and several methods of audio synthesis, convolutional autoencoders are used in this study to compress and reconstruct musical notes, and variational autoencoders to synthesize audio by sampling with learned hidden representations.

Throughout the project, various model architectures and training methods are investigated in an attempt of enhancing reconstruction quality and generation performance. Although generated sound is still not yet entirely polished, results demonstrate the potential of deep learning applications in music.

Keywords

Deep Learning, Autoencoders, Variational Autoencoders, Latent space, Spectrogram, Convolutional network.

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Chapter 1

Introduction

1.1. Motivation and Objectives

In recent years, deep learning has revolutionized generative tasks in fields like image synthesis, natural language processing, and audio production. Within music, research has generally split into *symbolic* approaches (focusing on note events, pitches, and durations in formats like MIDI) and *non-symbolic* approaches (focusing on raw audio waveforms or spectrograms).

Commercial digital audio workstations (DAWs) and synthesizers already allow users to generate audio with great precision. However, these are often not driven by deep-learning-based methods. Moreover, there is a compelling interest in exploring new audio possibilities achieved by learned latent representations, e.g., timbres that might not exist in standard synthesizer libraries.

This Bachelor’s Thesis therefore focuses on the implementation of two different, through related, deep learning architectures applied to music: Autoencoders (AEs) and Variational AEs. AEs will allow us to see how well musical fragments can be mimicked, while VAEs make possible music generation through latent space sampling. While the results may not surpass the polish or versatility of commercial synthesizers, such a model can reveal new pathways for interactive sound design and serve as a starting point to other research projects.

In any way, we would like this thesis to serve as an introduction and guide for students or anyone interested in the use of deep learning in music. While we do not assume extensive knowledge from the reader, we also will not go into excessively detailed explanations in order to keep the text accessible.

1.2. Work Plan

This section describes the work plan to follow in order to achieve the objectives outlined in the previous section.

Chapter 2

State of the Art

In this chapter, we aim to first provide a brief overview of the evolution of algorithmic composition and, secondly, explore non-symbolic (i.e., low-level) music generation more in depth.

2.1. Brief history of algorithmic composition

Algorithmic composition is the process of using some formal process to make music with minimal human intervention (Alpern, 1995) and can be divided into two main categories: *non-computer-aided* and *computer-aided* methods. The reader should note the following sections are nothing but a succinct run-through of algorithmic composition and will necessarily be incomplete (in terms of its content).

2.1.1. Non-computer-aided methods

Algorithmic composition dates back thousands of years. In Ancient Greece, philosophers such as Pythagoras (500 B.C.) viewed music as fundamentally linked to mathematics, believing that musical harmony reflected universal order (Simoni, 2003). These ancient Greek “formalisms” however are rooted mostly in theory, and their strict application to musical performance itself is probably questionable (Grout and Palisca, 1996). Therefore, it can’t really be said that Ancient Greek music composition was purely algorithmic in the sense we have defined it, but it undoubtedly set the path towards important formal extra-human processes.

Ars Nova marked a pivotal shift in musical thought, where composers such as Philippe de Vitry and Guillaume de Machaut began to disentangle rhythm from pitch and text. By systematically applying rhythmic patterns—known as the *talea*—to fixed melodic cells called the *chroma*, they developed a method of composition that can be seen as an early form of algorithmic music-making (Simoni, 2003). This approach can be better understood by looking at Figures 2.1, 2.2, and 2.3, which

respectively represent the talea, chroma and the mapping between them of *De bon espoir-Puisque la douce-Speravi* by Guillaume de Machaut.



Figure 2.1: Talea of the isorhythmic motet *De bon espoir-Puisque la douce-Speravi* by Guillaume de Machaut. Retrieved from (Simoni, 2003).

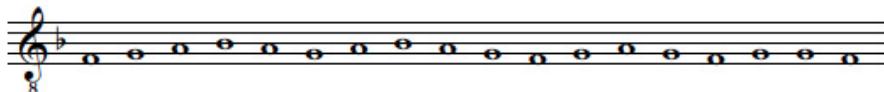


Figure 2.2: Color of the isorhythmic motet *De bon espoir-Puisque la douce-Speravi* by Guillaume de Machaut. Retrieved from (Simoni, 2003).



Figure 2.3: The tenor of *De bon espoir-Puisque la douce-Speravi* by Guillaume de Machaut. Retrieved from (Simoni, 2003).

In the Renaissance and the Baroque periods, algorithmic methods became more explicit through forms like the canon, where composers, like Johann Sebastian Bach, created strict rules dictating how single melodies are to be imitated by multiple voices at different times.

A famous Classical-era example is Mozart's *Musikalisches Würfelspiel* ("Dice Music") in which musical phrases were randomly assembled by dice rolls to allow any composer to form a waltz, explicitly employing chance-based algorithmic composition (Maurer, 1999).

The 20th century introduced more complex algorithmic techniques through serialism, where composers like Arnold Schoenberg and Alban Berg employed systematic tone-row matrices (see Figure 2.4) to structure their compositions through fixed rules. Composers such as John Cage and Karlheinz Stockhausen later incorporated chance and probabilistic methods, further extending the tradition of algorithmic music before the advent of computers (Simoni, 2003).

2.1.2. Computer-Aided Methods

The advent of computers in the mid-20th century significantly advanced algorithmic composition, introducing computational techniques that expanded creative

	I ₀	I ₁₀	I ₃	I ₄	I ₂	I ₁	I ₁₁	I ₉	I ₈	I ₇	I ₅	I ₆	
P ₀	E♭	D♭	G♭	G	F	E	D	C	B	B♭	A♭	A	R ₀
P ₂	F	E♭	A♭	A	G	G♭	E	D	D♭	C	B♭	B	R ₂
P ₉	C	B♭	E♭	E	D	D♭	B	A	A♭	G	F	G♭	R ₉
P ₈	B	A	D	E♭	D♭	C	B♭	A♭	G	G♭	E	F	R ₈
P ₁₀	D♭	B	E	F	E♭	D	C	B♭	A	A♭	G♭	G	R ₁₀
P ₁₁	D	C	F	G♭	E	E♭	D♭	B	B♭	A	G	A♭	R ₁₁
P ₁	E	D	G	A♭	G♭	F	E♭	D♭	C	B	A	B♭	R ₁
P ₃	G♭	E	A	B♭	A♭	G	F	E♭	D	D♭	B	C	R ₃
P ₄	G	F	B♭	B	A	A♭	G♭	E	E♭	D	C	D♭	R ₄
P ₅	A♭	G♭	B	C	B♭	A	G	F	E	E♭	D♭	D	R ₅
P ₇	B♭	A♭	D♭	D	C	B	A	G	G♭	F	E♭	E	R ₇
P ₆	A	G	C	D♭	B	B♭	A♭	G♭	F	E	D	E♭	R ₆
	R ₁₀	R ₁ ₁₀	R ₁ ₃	R ₁ ₄	R ₁ ₂	R ₁ ₁	R ₁ ₁₁	R ₁ ₉	R ₁ ₈	R ₁ ₇	R ₁ ₅	R ₁ ₆	

Figure 2.4: Serialism matrix. Retrieved from <https://www.musictheory.net>.

possibilities. Early pioneers like Lejaren Hiller and Leonard Isaacson composed the *Illiad Suite* (1957), one of the first pieces generated entirely by computer algorithms (Hiller and Isaacson, 1959). They utilized a generator/modifier/selector framework, where musical materials were algorithmically created, modified, and selected based on predefined rules (Maurer, 1999).

Composer Iannis Xenakis introduced *stochastic music*, employing probabilistic methods to generate musical structures. For instance, in his work *Atréees* (1962), Xenakis used probability distributions and random number generators to determine musical elements (Xenakis, 1992).

Computer-aided algorithmic composition can be categorized into three main approaches:

1. Stochastic systems: they incorporate randomness, ranging from simple random note generation to complex applications of chaos theory and nonlinear dynamics (Nierhaus, 2009).
2. Rule-Based systems: these utilize explicitly defined compositional rules or grammars, similar to earlier non-computer methods like the Renaissance canons or serialist compositions we have talked about. Notable examples include William Schottstaedt's automatic species counterpoint program and Kemal Ebcioglu's CHORAL system, which generate music based on historical compositional rules (Cope, 1991).
3. Artificial Intelligence systems: these systems extend rule-based methods by allowing a computer to develop or evolve compositional rules autonomously. David Cope's Experiments in Musical Intelligence (EMI) exemplifies this approach, analyzing existing compositions to create new music emulating specific composers' styles (Maurer, 1999).

2.2. Non-symbolic music generation

In Section 2.1 we gave an overview of historical algorithmic composition along with its two main branches: non-computer-aided and computer-aided methods, which largely focus on *symbolic* or high-level approaches. In this section, however, we turn our attention to *non-symbolic* music generation, where the emphasis is on generating and shaping audio signals directly.

We begin with an overview of foundational digital synthesis systems, which provided the bedrock for modern audio generation. We then discuss recent AI-based approaches, including various deep-learning architectures capable of producing music at the waveform (or spectrogram) level. Although this thesis aims to ultimately employ a conditional variational autoencoder for generating musical notes, understanding the broader ecosystem of audio-focused methods places our work in context.

2.2.1. Traditional Synthesis Systems

2.2.1.1. Additive Synthesis

Additive synthesis is a sound creation method based on the Fourier Theorem, which states that any sound can be decomposed into a sum of sine waves, or partials (Fourier, 1822). By controlling the frequency, amplitude, and phase of each partial, one can construct complex timbres from these elementary components. Historically, this idea finds early expression in acoustic instruments such as the pipe organ (see Figure 2.5), where multiple pipes combine to produce rich harmonic textures, and in pioneering electronic devices like the Telharmonium—often considered one of the first additive synthesizers.



Figure 2.5: Pipe organ created by Hermann von Helmholtz around 1862. Retrieved from <https://shorturl.at/VuT2w>.

The method was further advanced in the mid-20th century through the work of

Max Mathews at Bell Labs, who demonstrated the vast potential of digital additive synthesis for generating evolving and intricate soundscapes (Mathews, 1963). Although the flexibility of additive synthesis allows a precise crafting of any sound, its complexity made it less practical compared to the more cost-effective subtractive synthesis during the analog era. With the rise of digital signal processing, however, additive synthesis experienced a revival. This influenced the appearance of modern hybrid synthesizers that incorporate both additive and subtractive techniques (Roads, 1996; Tagi, 2023a).

2.2.1.2. Subtractive Synthesis

Subtractive synthesis is one of the most widely used methods in sound synthesis systems. Conceptually, this approach is not harder to understand than additive synthesis: starting with a complex waveform as the raw material, we want to shape it by filtering out unwanted frequencies, much like sculpting a figure from a block of marble. What do we shape this raw signal with? Well, a subtractive synthesizer primarily uses these components:

- Oscillators: are responsible for generating the initial complex waveforms rich in harmonics.
- Filters: which remove (or subtract) selected frequency components. This can be done with filters such as the so-called low-pass or high-pass, which respectively remove high and low frequencies.
- Amplifiers and envelope generators: amplifiers control the overall level of the sound over time while an envelope generator is a tool that shapes how a sound evolves when a note is played by controlling four different dimensions (see Figure 2.6):
 1. Attack: how quickly the sound reaches its peak.
 2. Decay: how fast it drops from the peak to a steady level.
 3. Sustain: the level at which the sound holds while the note is sustained.
 4. Release: how rapidly the sound fades after the note is released.

These simple stages allow you to craft sounds that can be sharp and percussive or smooth and evolving (Hahn, 2022).

- LFOs (Low-Frequency Oscillators): LFOs operate at very low frequencies that are below the threshold of human hearing and can create effects like vibrato or tremolo, therefore bringing the possibility of adding movement and life to a sound (Tagi, 2023b).

Historically, subtractive synthesis dates as back as 1930 with instruments such as the Trautonium and continued to be used throughout the 20th century, for example, by Robert Moog's Minimoog (Réveillac, 2024).

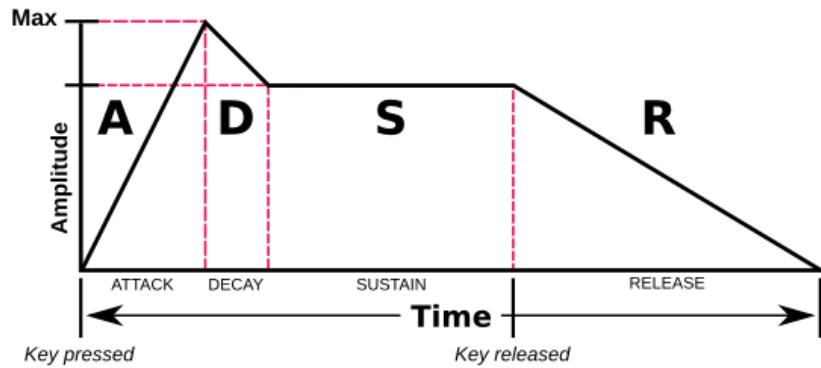


Figure 2.6: Illustration of Attack (A), Decay (D), Sustain (S) and Release (R). Retrieved from <https://shorturl.at/Zj3UQ>.

2.2.1.3. Frequency Modulation (FM) Synthesis

Frequency modulation synthesis (FM synthesis) is a method of sound design in which one oscillator, known as the *modulator*, modulates the frequency of another oscillator, called the *carrier*, which allows to create new frequency components without filters (see Figure 2.7). In simple terms, rather than “sculpting” a sound by removing frequencies (as in subtractive synthesis), FM synthesis generates complex spectra by dynamically altering the pitch of a carrier with a modulating signal (Cymatics, 2025).

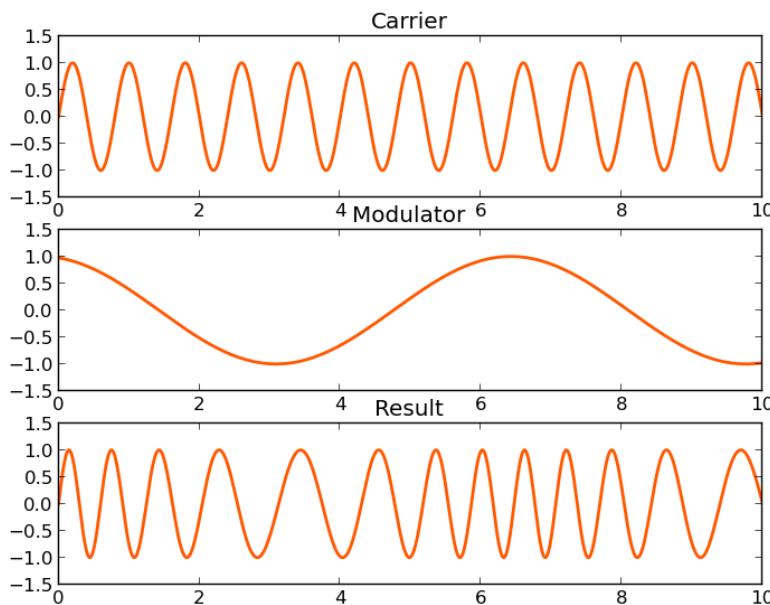


Figure 2.7: Illustration of a carrier, a modulator and the output. Retrieved from (Cymatics, 2025).

FM synthesis is the result of John Chowning's experiments in 1967 at Stanford University: by using sine waves (using one to modulate the frequency of another) Chowning discovered that a variety of new timbres could be generated (Cymatics, 2025).

FM synthesis revolves around the building block of an *operator*, that typically includes an oscillator, an amplifier, and an envelope generator (recall we have talked about these in subtractive synthesis). Operators can serve as carriers, modulators, or both, and they are arranged in various configurations or algorithms to produce different sound textures (Tagi, 2023c).

2.2.1.4. Other Approaches: Granular, Physical and Spectral Modeling

Beyond the traditional methods of additive, subtractive, and FM synthesis, there do exist other important and more modern sound generation techniques. We will very briefly talk about three of them.

Granular synthesis works by breaking a sound into tiny segments called grains. These grains can then be individually rearranged to create a rich variety of sound textures, from subtle ambiances to complex, glitch-like effects (Roads, 1996).

Physical modeling Synthesis takes a different route by simulating the behavior of real-world systems, such as vibrating strings, which allows for realistic emulations of acoustic instruments. (Smith, 1996)

Spectral modeling involves analyzing a sound's frequency content (often with Fourier techniques) and then resynthesizing it by manipulating its spectral components. This way, interpolation and morphism between sounds can be achieved in a simpler way than with other traditional synthesis methods (Serra, 1998).

2.2.2. Modern AI-Driven Non-Symbolic Music Generation

Unlike the traditional systems based on handcrafted signal-processing algorithms, deep learning methods for non-symbolic music generation learn representations directly from data. They typically produce raw audio waveforms or time-frequency representations, such as spectrograms. In recent years, several influential neural architectures have emerged, capable of generating musical audio directly at the waveform level. Our model will also follow this paradigm.

2.2.2.1. Waveform Modeling Approaches

WaveNet is a neural network initially designed for generating realistic speech audio directly from waveform samples. WaveNet operates by predicting each audio sample based on previously generated samples, using dilated causal convolutional layers. These dilations expand the receptive field, allowing the network to capture both fine-grained details and wider temporal context, which seems essential for

modeling realistic audio textures. Although initially designed for text-to-speech synthesis, WaveNet was quickly adapted for music and demonstrated its effectiveness in capturing musical features at the waveform level (van den Oord et al., 2016).

Another significant development was the introduction of **SampleRNN** (Mehri et al., 2017), a hierarchical recurrent neural network (RNN) architecture specifically created to handle the complexity of raw audio generation. SampleRNN models waveforms at multiple temporal scales by stacking RNN layers hierarchically, allowing each layer to focus on different aspects of musical structure. Higher layers manage broader temporal dependencies, capturing long-term patterns, while lower layers handle local audio details (Maurer, 1999).

Another significant breakthrough in non-symbolic music generation was achieved with Generative Adversarial Networks (GANs). An important example is *GAN-Synth*, developed by *Google Magenta*, which synthesizes audio notes using generative adversarial networks operating in the frequency domain (Engel et al., 2019). Unlike WaveNet and SampleRNN, which sequentially generate each sample, GANSynth produces entire audio clips simultaneously by generating spectrograms and instantaneous frequency components. This approach results in more realistic and coherent musical timbres. Additionally, GANSynth allows for audio synthesis control, which enables independent manipulation of pitch and timbre.

Chapter 3

Audio representation basics

In this chapter, our aim is to explain the fundamentals of audio and their most frequent representations. We see this necessary in order to be able to at least have a shallow and intuitive understanding of the deep learning model we have built and of which we will talk about in chapter 5.

First, we will give a brief introduction of what audio is and its basic components. Next, both the time and frequency domain representations will be explained, along with subtopics related to each of them.

3.1. Introduction to audio data

According to Oxford's dictionary, sound is the collection of vibrations that travel through the air or another medium and can be heard when they reach a person's or animal's ear. This is probably the definition anyone could have come up with, but in order to deeply understand sound, a closer look at its physical meaning is needed.

A common approach is to model sound as a wave that propagates through some medium. Like any other wave, it is constituted by (see Figure 3.1):

- Amplitude: it is simply the distance (measured in meters) of the wave from the resting position at a given point in time. Humans perceive amplitude as loudness. The bigger the amplitude, the louder the wave will sound.
- Frequency, period and wavelength: these three properties are closely related to the speed of the wave. Frequency is the number of oscillations of the wave during some period of time; the period and the wavelength are respectively the time and distance it takes for the wave to start repeating itself. Mathematically, if f , T , λ and v , denote the frequency, period, wavelength and speed of the wave, we have: $\lambda = v \cdot T = v/f$. Frequency is measured in Hertz (Hz), the period in seconds, and the wavelength in meters.

Since frequency, period and wavelength are all directly or inversely related, explaining how humans perceive one of them allows us to understand the rest. In particular, we perceive the frequency of a sound as its pitch, which tells us how high or low the sound is. The higher the sound, the higher its frequency will be, and therefore the more oscillations per second the wave will go through.

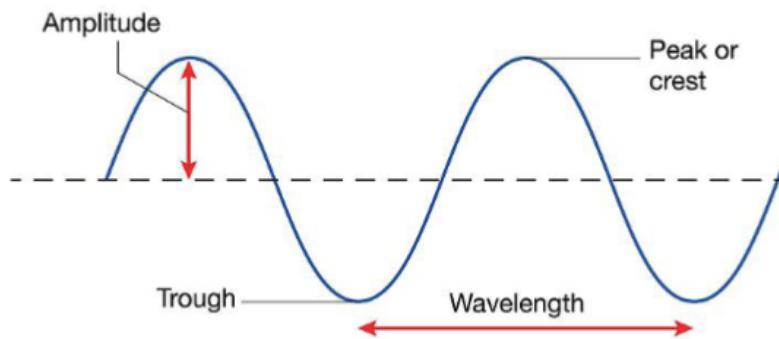


Figure 3.1: A wave can be characterized by its amplitude, frequency and wavelength. Retrieved from <https://shorturl.at/KtTHs>.

Now that we know what sound is, we can turn our attention to audio data, where audio refers to any recorded, transmitted or reproduced sound. Naturally, a sound wave is a continuous curve containing an infinite number of values over time. If we want to digitally represent this audio wave it is clear we cannot digitally store an infinite amount of information about it. Instead, the sound wave is converted into a collection of discrete values, also known as a digital representation (HuggingFace, 2023). In fact, the different types of audio files formats, such as .mp3, .wav, etc., correspond to the way the digital representation of an audio wave is encoded.

In the following sections we will discuss two of the most important digital representations of audio that are used today: time and frequency domain representations.

3.2. Time-Domain representation

A time-domain representation refers to a way of representing signals as they change over time. The usual way of representing sound in the time domain is the waveform, where the amplitude of the signal is plotted against time. In digital systems, this waveform is commonly stored using pulse code modulation encoding (PCM), which captures the amplitude values at regular intervals.

3.2.1. Sampling and sampling rate

In order to capture the wave's information through time, we apply sampling – the process of measuring the value of a continuous signal at a fixed and finite amount of time steps (see Figure 3.2).

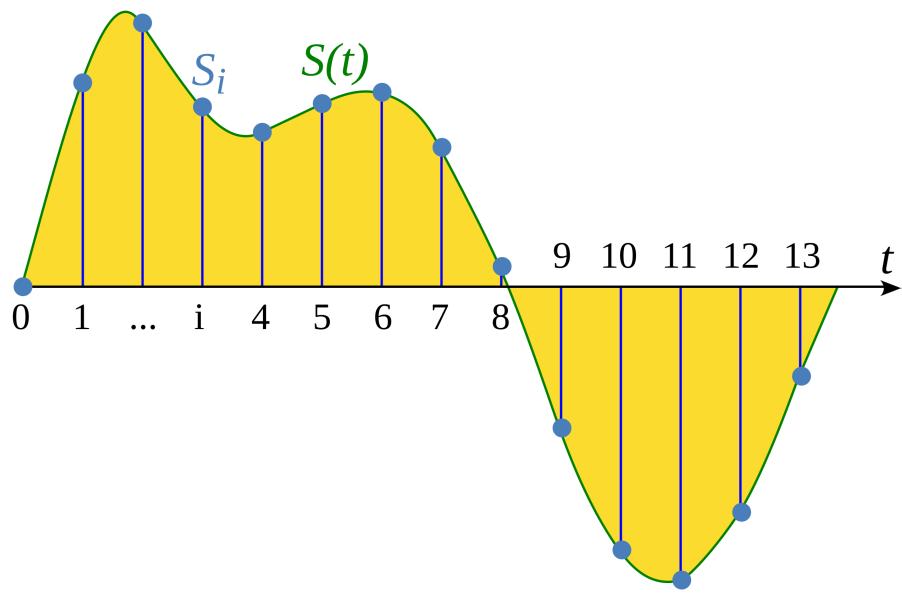


Figure 3.2: Example of sampling of an audio wave through time. Retrieved from (HuggingFace, 2023).

The sampling rate¹ is the number of samples taken in one second and is measured in Hz. For example, songs uploaded to Spotify are uploaded with a sample rate of 44.1 kHz, so each second of audio is represented by 44100 samples. For comparison, high-resolution audio has a sampling rate of 192 kHz.

The choice of the sampling rate determines the Nyquist frequency, which is the highest frequency that can be captured by the system and always equals half the sampling rate. For instance, if the sampling rate is 16kHz, then the highest frequency the audio representation will be able to model is 8 kHz.

3.2.2. Amplitude and Bit Depth

The sampling rate tells us how often samples are taken, but what exactly are we sampling? Each sampled value corresponds to the *amplitude* of the sound wave at a given instant in time. For sound waves traveling through air, the amplitude represents the deviation of air pressure from its ambient (resting) level — not the physical displacement we had previously introduced, but the change in pressure caused by the wave. This deviation is a physical quantity, usually measured in pascals (Pa).

Besides deciding *how often* we sample, there is another important aspect to take into account when converting a continuous signal into a digital form: bit depth. This metric determines the resolution of the amplitude values: the number of distinct levels into which the continuous signal is quantized. Common audio formats use 16-bit

¹A great resource for visualizing the sampling rate is <https://jvbalen.github.io/notes/waveform.html>.

or 24-bit depth, corresponding to 65,536 and 16,777,216 possible levels, respectively.

Quantization introduces rounding error, which manifests as quantization noise. The higher the bit depth, the finer the resolution and the lower the resulting noise. In practice, 16-bit audio already provides noise levels below the threshold of human hearing, making it sufficient for most use cases (HuggingFace, 2023).

To better align with how we perceive sound intensity, amplitude values are often expressed on a logarithmic scale — in *decibels* (dB). In acoustics, this is referred to as the sound pressure level (SPL), defined as:

$$\text{SPL (dB)} = 20 \cdot \log_{10} \left(\frac{p}{p_{\text{ref}}} \right),$$

where:

- p is the measured sound pressure, in pascals (Pa),
- p_{ref} is the reference pressure, typically $20 \mu\text{Pa}$.

It is worth noticing that although pressure is measured in pascals, SPL in decibels is a *dimensionless quantity*, since it represents a ratio of pressures on a logarithmic scale.

This logarithmic representation mirrors human perception: we are more sensitive to small changes at low intensities than at high ones. For example, an increase of about 6 dB is perceived as roughly twice as loud. In real-world audio, 0 dB SPL corresponds to the quietest sound the average human ear can hear, and louder sounds have positive dB values.

In digital audio systems, however, amplitude is typically expressed in decibels relative to full scale (dBFS). Here, 0 dBFS represents the amplitude of a full-scale digital signal, that is, the reference level against which all other amplitudes are measured. All other levels are negative, indicating lower amplitudes. As a rule of thumb, every decrease of 6 dB approximately halves the amplitude, and signals below -60 dBFS are generally considered inaudible.

3.2.3. Waveform

We are now in position to talk about waveforms, which are nothing but a plot of the sampled values of a sound over time which illustrates the changes in its amplitude. This visualization comes in handy for identifying specific features of audios such as its overall loudness or individual sound events (see Figure 3.3²).

It is worth noting that this waveform is not periodic (unlike the one in Figure 3.1), because the sound it represents is not a pure sinusoid. Instead, it may be a

²For consistency, the same 4 second clip from ABBA's *Lay all your love on me* will be used throughout the chapter.

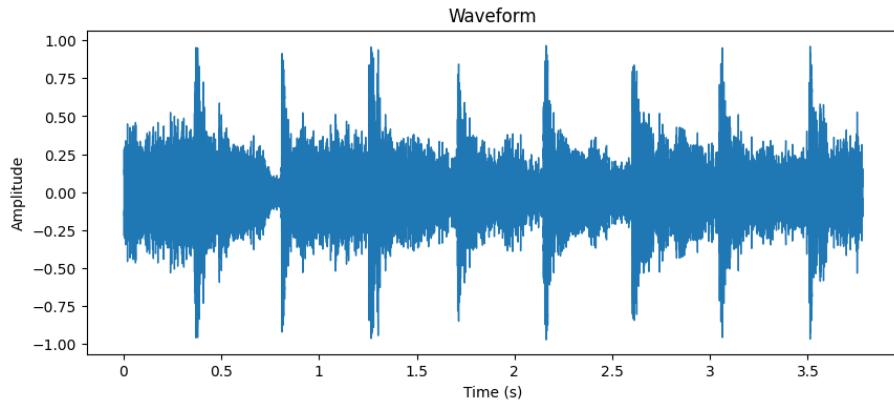


Figure 3.3: A waveform plot of a 4 second clip of ABBA's *Lay all your love on me*.

sum of multiple non-periodic sinusoids with different frequencies, or even a random signal like white noise. We will go more into depth about wave decomposition in section 3.3.

3.3. Frequency-domain representation

The frequency-domain representation provides a different way of representing sounds. While time-domain representation focuses on how a signal changes over time, frequency-domain analysis emphasizes the individual frequencies that make up the signal. This has a significant number of applications. For example, in speech separation, which aims to identify the different speakers in a conversation, different sound sources are separated based on their frequencies.

In this section, we will explore the key concepts in frequency-domain representation, starting with the frequency spectrum, then moving on to the Discrete Fourier Transform (DFT) and its applications, followed by an examination of spectrograms.

3.3.1. Fourier transform and the frequency spectrum

As we have already stated, the frequency-domain representation of an audio signal is a way to decompose it into a sum of pure periodic components, each corresponding to a specific frequency.

In order to compute a signal's individual frequencies, the Fourier transform is used. The foundation is Fourier's Theorem, which states that any periodic function can be expressed as a sum of sine and cosine functions (or equivalently, complex exponentials³) with specific amplitudes and phases. Here, the phase of each component describes its position within its cycle at a given time — that is, how much the wave is shifted horizontally relative to a reference. This mathematical transforma-

³By Euler's Identity, we have $e^{i\theta} = \cos \theta + i \sin \theta$.

tion breaks down signals into simpler sine and cosine waves, each corresponding to a specific frequency (see Figure 3.4).

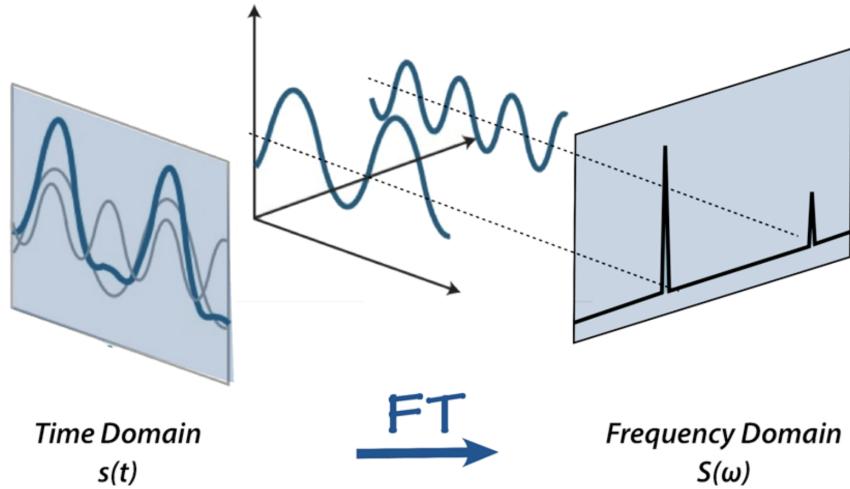


Figure 3.4: Effect of applying the Fourier transform to a signal. Retrieved from <https://shorturl.at/8D73u>.

The Fourier transform of a continuous signal $x(t)$ is given by:

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-i2\pi ft} dt$$

where:

- $X(f)$ is the Fourier transform of the signal $x(t)$ that tells us how much of each frequency f is present in the original time-domain signal.
- $x(t)$ is the original signal in the time domain.
- $e^{-i2\pi ft}$ represents oscillations at a frequency f . This expression combines sine and cosine components, and the negative sign indicates the oscillation direction.

The DFT is a version of the Fourier Transform specifically designed for discrete signals, which are composed of a finite number of samples. By applying the DFT, we can approximate the frequency content of a non-periodic, sampled signal.

If we have a sequence of N samples $\{x_k\}_{k=0}^{N-1}$, the DFT transforms them into a series of complex numbers:

$$X_k = \sum_{n=0}^{N-1} x_n \cdot e^{-i2\pi \frac{k}{N} n} \quad (3.1)$$

Esentially, it takes the signal's discrete samples and computes a frequency representation over a finite duration. In fact, taking the modulus of the output of the DFT gives us the amplitude information at a given moment, while the angle between the real and imaginary components of the output provides the so-called phase spectrum.

But what if we want to see how the frequencies of an audio change over time? The solution to this question is to compute the audio's spectrogram, a very informative audio representation that allows us to jointly visualize time, frequency and amplitude, all in the same graph.

In order to compute a spectrogram the Short Time Fourier Transform (STFT) is used. This algorithm (see Figure 3.5) first divides the signal into possibly overlapping and brief segments or windows (usually lasting a few milliseconds). Secondly, it applies the DFT to each of them in an efficient way with the Fast Fourier Transform (FFT) algorithm. Finally, all collected spectra are stacked through the time axis, forming the spectrogram. Thus, when looking at the resulting spectrogram (see Figure 3.6), each vertical stripe represents the frequency spectrum at a specific point in time, seen from the top.

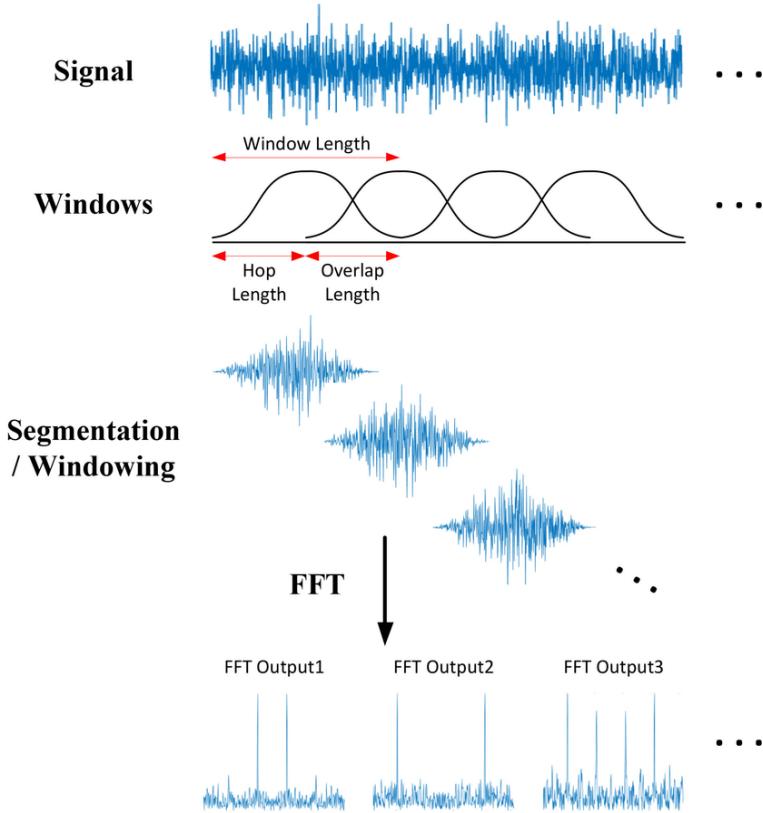


Figure 3.5: Short Time Fourier Transform diagram. Retrieved from <https://shorturl.at/JGODG>.

Note that overlapping windows are crucial in the STFT because they improve the chances of capturing short-lived events in the signal. Without this overlap, transient

features can fall between two consecutive windows and be either poorly represented or entirely missed in the time-frequency analysis. We can think of it like scanning a text with a narrow spotlight: if we move the light in non-overlapping steps, some words might fall in the dark gaps and go unread.

An example of a spectrogram can be seen in Figure 3.6. In it, the reader might have the impression that the spectrogram is either incorrect or lacks meaningful information. However, this “black void” does contain relevant data — it’s just not visually apparent due to the use of a linear amplitude scale. By re-plotting the same spectrogram with a logarithmic (dB) scale, the previously hidden details in the darker areas become clearly visible in Figure 3.7.

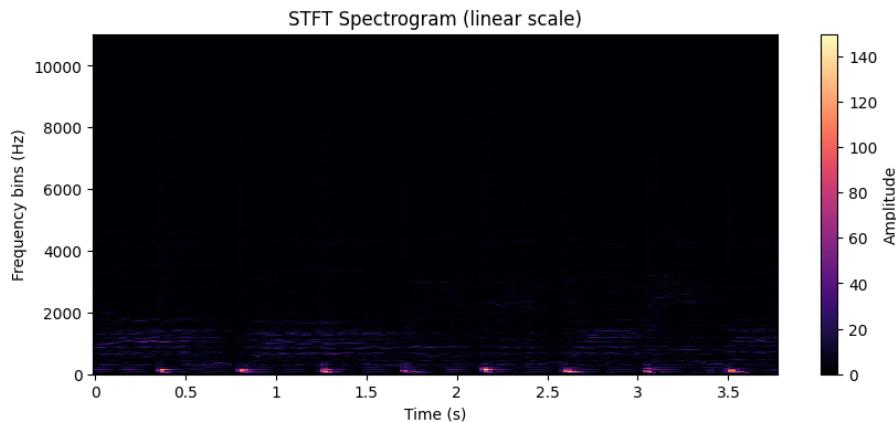


Figure 3.6: Resulting linear spectrogram after applying the STFT to a 4 second clip from ABBA’s *Lay all your love on me*.

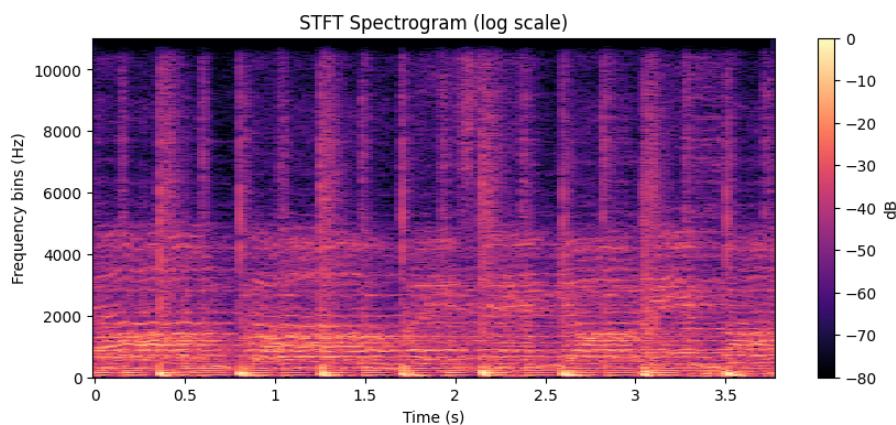


Figure 3.7: Resulting log spectrogram after applying the STFT to a 4 second clip from ABBA’s *Lay all your love on me*.

An important observation about the STFT is that it is an invertible function, so it’s possible to turn the spectrogram back into the original waveform. This means the spectrogram and the waveform are really just different views of the same data.

The discrete case inverse STFT can be computed as:

$$x_k = \frac{1}{N} \sum_{k=0}^{N-1} X_k \cdot e^{-i2\pi \frac{k}{N} n} \quad (3.2)$$

While the continuous STFT is theoretically invertible, in practice we work with sampled signals and apply a discrete STFT. This discrete version is also invertible, but it can only reconstruct the discrete signal it was computed from — not the original continuous-time signal. Since sampling inherently introduces some information loss, the inverse STFT only recovers an approximation of the original waveform.

3.3.2. Mel spectrograms

A mel spectrogram (mel is short for “melody”) is a kind of spectrogram characterized by changing the measurement of the frequency axis. In particular, while in a standard spectrogram the frequency axis is linear and is measured in Hz, a mel spectrogram applies a set of filters, also known as mel filterbank, to each spectrum, which transforms the frequencies to a logarithmic scale, commonly known as mel scale. The mel scale is non-linear and compresses higher frequencies while expanding lower ones.

But why might mel spectrograms be useful? Humans don’t perceive changes in sound in a linear manner, but a logarithmic manner instead. This can be observed empirically⁴ when listening at two pairs of sounds that differ in 50Hz but one pair has a much higher frequency value than the other: the change from 300 to 350 Hz is much more evident than the change from 8000 to 8050 Hz, which wouldn’t happen if we were able to perceive these changes linearly. Thus the mel scale, and mel spectrograms, allow to model frequency-domain representations with a higher fidelity to how humans perceive sound, a fact that should probably be taken into consideration when training a neural network on an audio task.

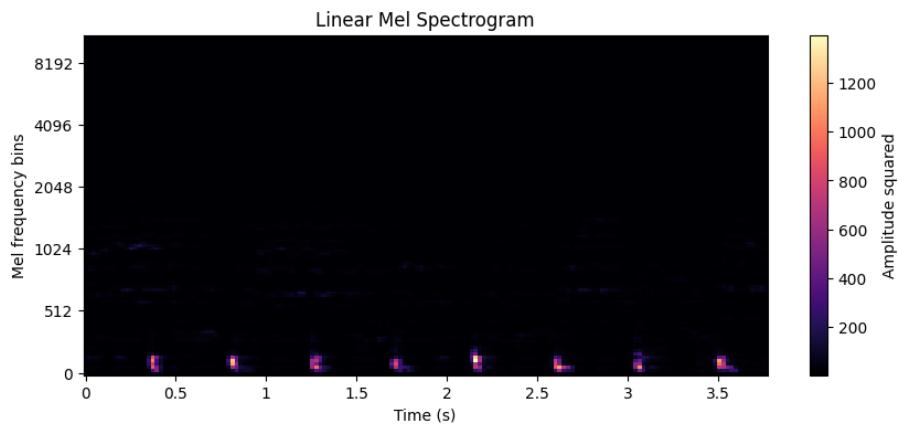


Figure 3.8: Linear mel spectrogram of a 4 second clip of ABBA’s *Lay all your love on me*.

⁴<https://onlinetonegenerator.com/>

As with Figures 3.6 and 3.7, the dark region in Figure 3.8 should not be interpreted as a loss of information. Rather, it reflects how information is visually obscured when using a linear amplitude scale. Once again, converting the amplitude to decibels reveals these hidden details, making the structure of the spectrogram more perceptible.

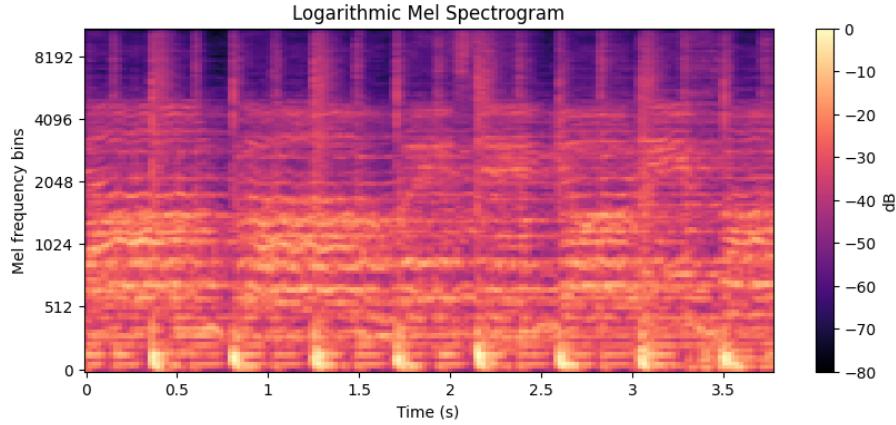


Figure 3.9: Logarithmic mel spectrogram of a 4 second clip of ABBA’s *Lay all your love on me*.

3.4. Practical Considerations in Audio Representation

In this section, we briefly discuss the trade-offs involved in selecting Short-Time Fourier Transform (STFT) parameters for converting audio signals into spectrograms. An overview of the specific parameter values used by our model is provided in Chapter 5.

3.4.1. STFT parameters and signal reconstruction

The size and quality of the input spectrograms we feed to our model depends on the parameters with which we obtain said spectrograms (Manilow et al., 2020). It is therefore important that we understand the role each of these parameters play:

- Window type: determines the shape of the short-time window applied to each audio segment. Different window shapes control how the signal is weighted, especially at the edges, to minimize spectral leakage (the spreading of frequencies into neighboring ones). Windows with smoother edges, like the Hann window, reduce leakage but come with a trade-off between time resolution and frequency resolution. A sharper window (e.g., rectangular) has better time resolution but more leakage. In our models, we use the Hann window,

as it strikes a good balance between leakage reduction and frequency clarity, making it a common choice for deep learning audio tasks.

- Window length: specifies the number of samples that each short-time window contains. The window length significantly influences the frequency resolution of the spectrogram: longer windows provide higher frequency resolution, while shorter windows provide better time resolution. The trade-off between time and frequency resolution is visible in 3.11
- Hop length: designates how many samples are skipped between two consecutive short-time windows. The shorter the hop length is, the more detailed the time axis will be (see Figure 3.12) and the larger the computational load will be. On the other hand, a longer hop length can result in a more compressed time axis, potentially causing a loss of temporal information.

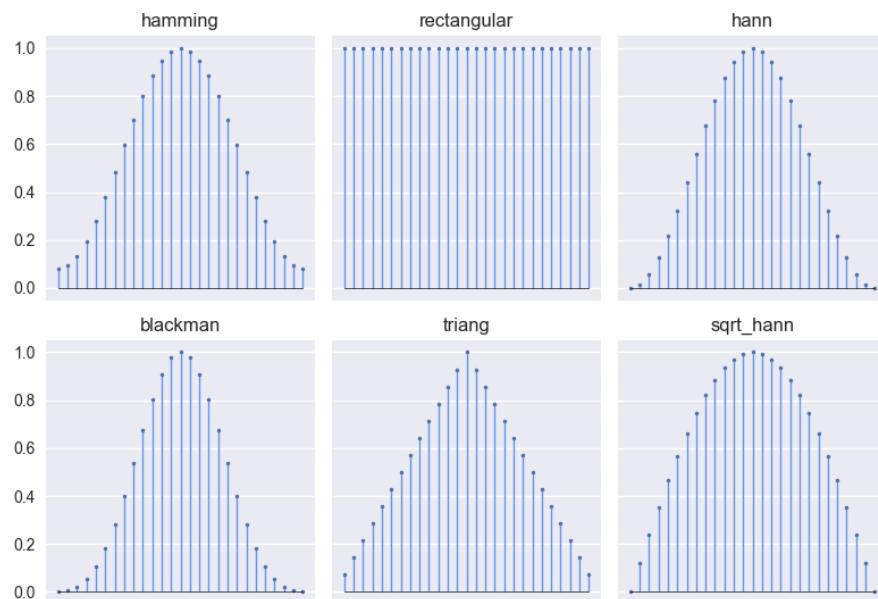


Figure 3.10: The choice of window function can significantly influence the spectral resolution and leakage effects in the spectrogram. Retrieved from <https://shorturl.at/UsEsq>.

3.5. Conclusion

In this chapter, we have explored the fundamental concepts of audio and its digital representations, which we consider essential for understanding audio tasks in deep learning. We began by discussing the basic properties of sound, such as amplitude, frequency, and wavelength, and how these relate to our perception of loudness and pitch. We then moved on to how continuous sound waves are converted into digital audio data through sampling, amplitude quantization, and bit depth, leading to time-domain representations like waveforms.

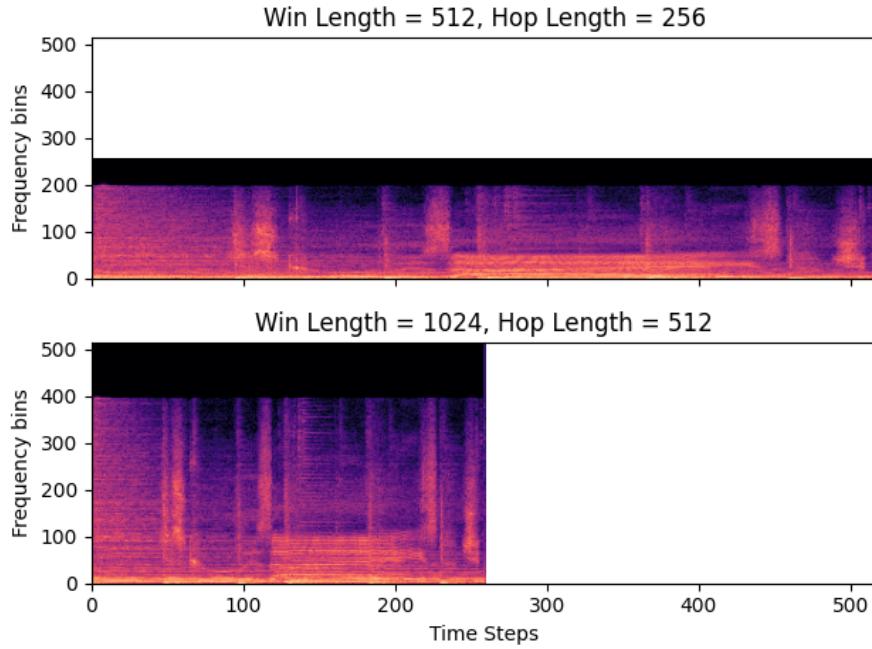


Figure 3.11: A shorter window results in finer time details but poorer frequency resolution, while a longer window captures more frequency details but at the cost of time resolution. Retrieved from <https://shorturl.at/UsEsq>.

We also examined frequency-domain representations, focusing on the Fourier transform and its discrete version (DFT), which allow us to analyze an audio signal in terms of its individual frequencies. We looked at spectrograms and mel spectrograms, which provide insights into how sound evolves over time and in the case of mel spectrograms, take into account human hearing perception. Finally, we discussed key parameters involved in creating spectrograms, such as window length and hop length.

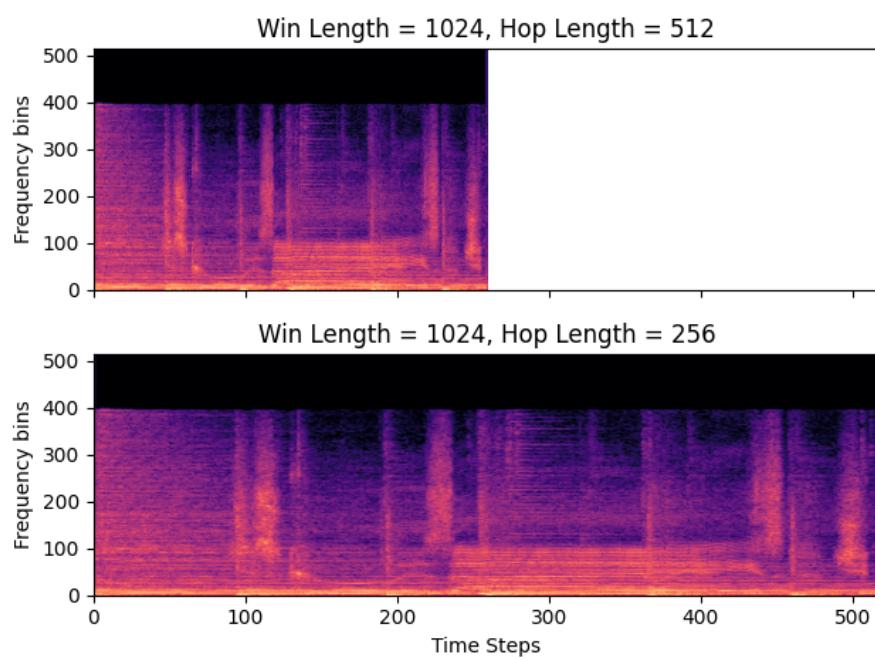


Figure 3.12: The smaller the hop length the more times a particular segment of the audio signal is represented in the STFT. Retrieved from <https://shorturl.at/UsEsq>.

Chapter 4

Introduction to deep learning, autoencoders and variational autoencoders

In this chapter, we will first provide the reader with a basic understanding of what deep learning is and its main components. The aim is not to go into detail but rather gain the necessary intuition to be able to grasp an end-to-end deep learning architecture. This will be useful for those who are introducing themselves in the topic to better comprehend our project and decisions within it.

Once we have done this, we will go a little further by explaining from both a broad and a detailed perspective the deep learning models and architectures we have used in this project: convolutional neural networks, autoencoders, variational autoencoders and conditional variational autoencoders.

4.1. Deep Learning

Deep learning is a subset of machine learning that uses big neural networks to model complex patterns in data. A neural network consists of units called neurons, organized in layers: an input layer, one or more hidden layers, and an output layer. Each neuron applies a weighted sum of its inputs, adds a bias, and passes the result through a non-linear activation function.

A typical feed-forward pass through a single neuron can be expressed as:

$$z_j = \sum_i w_{ij} x_i + b_j, \quad a_j = \sigma(z_j), \quad (4.1)$$

where x_i denotes the inputs, w_{ij} are the weights connecting input i to neuron j , b_j is the bias term, z_j is the neuron's pre-activation, $\sigma(\cdot)$ is a non-linear activation function (such as ReLU, sigmoid, or tanh), and a_j is the neuron's output. Stacking many such neurons into multiple layers allows deep networks to learn hierarchical representations of data (Goodfellow et al., 2016).

If we want a model to learn about some data, we need to train it. Training a model involves finding the best weights and biases to minimize a loss function. This function measures how far off the model’s predictions are from the actual targets. A common loss for regression problems is the Mean Squared Error (MSE):

$$\mathcal{L}_{\text{MSE}} = \frac{1}{n} \sum_{i=1}^n (y_i - \hat{y}_i)^2. \quad (4.2)$$

The optimization is usually done using gradient descent with the gradients of the weights in the network computed, using the backpropagation technique. During backpropagation, the chain rule is applied to propagate the error signal from the output layer back through the hidden layers, and by that means adjusting each weight to reduce the overall error in a direction guided by the negative gradient of the loss. Thanks to it, weights are updated iteratively using an optimizer like stochastic gradient descent (SGD) or Adam.

In order to train and evaluate a deep learning architecture we need a dataset from which to gather data. This data is usually divided into three parts: training, validation, and test sets. The training set is used to learn the model parameters, the validation set is used to tune hyperparameters and avoid overfitting (a situation where the model “memorizes” data instead of learning it), and the test set evaluates the model’s generalization ability. There exist several ways to try to avoid overfitting, such as regularization or early stopping (Goodfellow et al., 2016).

4.2. Convolutional Neural Networks (CNNs)

Convolutional Neural Networks (CNNs) are a class of deep learning models particularly well-suited for data with a grid-like topology, such as images (2D grids of pixels) or audio spectrograms (2D time-frequency grids). A CNN introduces two key concepts: *local receptive fields* and *weight sharing*. Rather than connecting every input unit to every neuron in the next layer, as in a fully-connected network, a convolutional layer uses a small *filter*, also known as *kernel*, that slides across the input to produce feature maps. This filter is a learnable matrix of weights applied to local regions of the input, detecting specific local patterns (e.g., edges, textures) wherever they might appear. The same set of filter weights is reused for every location in the input (*convolution* and weight sharing), which greatly reduces the number of parameters and makes the model more efficient (LeCun et al., 1998; Krizhevsky et al., 2012).

A typical CNN architecture consists of an input layer, followed by repeated stacks of convolutional layers, activation functions (like ReLU), and pooling layers. Pooling layers aggregate information in local neighborhoods (for example taking the maximum or average of that region), reducing the spatial dimensions of the feature map and attempting to capture the most important characteristics of the data. Repeated convolution added to pooling operations allow the network to extract increasingly

abstract features at deeper layers, while gradually reducing dimensionality. Ultimately, one or more fully connected layers consolidate the extracted features for the final prediction (LeCun et al., 1998; Krizhevsky et al., 2012). A nice visual of a typical CNN architecture can be seen in Figure 4.1.

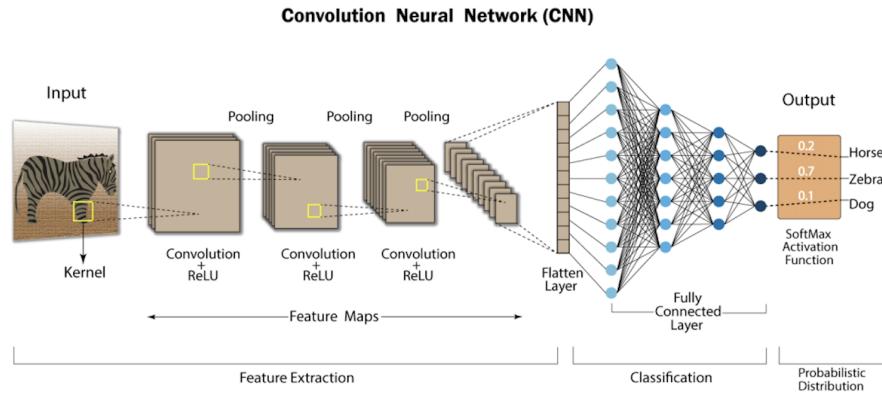


Figure 4.1: Typical CNN architecture. Retrieved from <https://shorturl.at/1uqcP>

CNNs have been extremely successful in computer vision tasks. Classic examples include LeCun’s LeNet-5 for handwritten digit recognition (LeCun et al., 1998) and the AlexNet network that won the 2012 ImageNet competition (Krizhevsky et al., 2012). Both architectures demonstrated the power of deep CNNs on large-scale image data.

In our project, we used CNN layers as the main components of the autoencoder, a type of network we will talk about in short time. In our case, CNNs were used to process spectrograms derived from the NSynth dataset (Engel et al., 2017). Spectrograms can be viewed as 2D representations of audio signals (time vs. frequency), and are therefore fit to convolutional operations.

Additionally, recent work has explored CNNs for interactive and explanatory purposes in various domains, including audio generation. For example, CNN Explainer (Wang et al., 2020)¹ demonstrates how convolutional kernels learn from image data, and similar principles extend to audio, where convolutional layers automatically discover patterns corresponding to timbral or temporal events.

4.3. Autoencoders

An autoencoder is a type of neural network made up of two main components: an encoder and a decoder. The encoder compresses the input x into a typically lower-dimensional latent representation h , and the decoder reconstructs an output \hat{x} from this representation so that \hat{x} closely matches the original input x (Michelucci, 2022; Bank et al., 2021). By minimizing a reconstruction loss between x and \hat{x} , the

¹This is a great resource to closely understand how CNNs work

autoencoder is forced to learn the most salient features of the input. For example, a simple mean squared error (MSE) reconstruction loss is:

$$\mathcal{L}_{\text{AE}} = \|x - \hat{x}\|^2, \quad (4.3)$$

where $\|\cdot\|^2$ indicates element-wise squared difference.

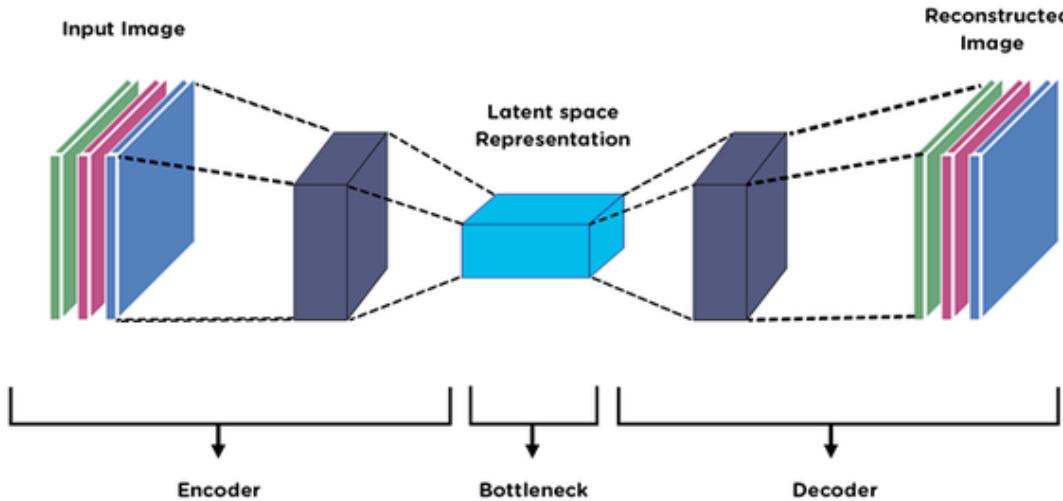


Figure 4.2: General structure of an autoencoder. The network consists of an encoder that compresses the input into a latent representation, and a decoder that reconstructs an output from it. Retrieved from <https://shorturl.at/esPtm>.

In the simplest form of autoencoder, both encoder and decoder are neural networks (often mirrored architectures) and h is a fixed-size vector (or tensor) of lower dimension than x . The hope is that h is an *informative* representation, meaning it captures the essential factors of variation in the data while discarding noise or irrelevant details. This learned latent space can then be useful for tasks like dimensionality reduction, visualization or anomaly detection (where high reconstruction error highlights anomalies).

There exists different types of autoencoders, and each of them serves a different functionality:

- **Denoising autoencoders** add noise to the input and train the model to reconstruct the original clean input, which encourages the network to learn robust features rather than simply memorizing the data (Michelucci, 2022).
- **Sparse autoencoders** impose a sparsity penalty on the latent representation, encouraging the network to use only a small number of active neurons for any given input. This often leads to the discovery of meaningful, disentangled features.

- **Convolutional autoencoders** apply convolutional layers in the encoder and decoder, which are especially effective for spatial or temporal data like images or spectrograms, since they preserve local structure. In our project, we use a convolutional autoencoder to learn compact representations of musical sounds through spectrograms.

Ultimately, an autoencoder can learn an informative and compressed representation of data in an unsupervised manner. However, this deep learning architecture is not enough for the purposes of our thesis, since our aim is to be able to generate data from the learned distribution of samples. For this reason, we now introduce variational autoencoders.

4.4. Variational Autoencoders (VAEs)

Latent variable models are a common tool in probabilistic modeling, particularly when dealing with data that is assumed to be generated by hidden or unobservable factors. These latent variables z are not directly accessible, but they influence the observed data x . For example, in audio signals, latent variables could correspond to features such as pitch, rhythm, or timbre. In medical domains, health is often treated as a latent quantity inferred from observable tests, such as blood tests or pressure. We assume a generative process where z is sampled from a prior distribution $p(z)$, and then x is sampled from a likelihood $p_\theta(x|z)$. This leads to a joint distribution:

$$p_\theta(x, z) = p_\theta(x|z)p(z)$$

The marginal likelihood of the observed variable is then given by:

$$p_\theta(x) = \int p_\theta(x|z)p(z)dz$$

In most realistic models, the integral in the expression above is not tractable. As a result, computing the exact posterior $p_\theta(z|x)$ is infeasible:

$$p_\theta(z|x) = \frac{p_\theta(x, z)}{p_\theta(x)} = \frac{p_\theta(x|z)p(z)}{\int p_\theta(x|z)p(z)dz}$$

To overcome this issue, variational inference introduces an approximate posterior $q_\phi(z|x)$ drawn from a simpler, tractable family of distributions (typically, a diagonal Gaussian). The problem then is to find a q_ϕ that is as close as possible to the true posterior. To measure this closeness, we use the Kullback-Leibler (KL) divergence, which quantifies how one probability distribution differs from another. More formally, the KL divergence between two distributions $q(z)$ and $p(z)$ ² is defined as:

$$\text{KL}(q(z)\|p(z)) = \int q(z) \log \frac{q(z)}{p(z)} dz,$$

²Note that $\text{KL}(q(z)\|p(z)) \neq \text{KL}(p(z)\|q(z))$.

a quantity that is always non-negative and equals zero only when $q(z) = p(z)$ almost everywhere.

In our case, the KL divergence expresses how much information is lost when using the approximate posterior $q_\phi(z|x)$ instead of the true posterior $p_\theta(z|x)$. The optimal approximate posterior is therefore obtained by minimizing:

$$q_\phi^*(z|x) = \arg \min_{q_\phi} \text{KL}(q_\phi(z|x) \| p_\theta(z|x))$$

To analyze this objective more concretely, we fix a datapoint x and derive a decomposition of the KL divergence:

$$\begin{aligned} \text{KL}(q(z|x) \| p(z|x)) &= \int q(z|x) \log \frac{q(z|x)}{p(z|x)} dz \\ &= \int q(z|x) (\log q(z|x) - \log p(z|x)) dz \\ &= \mathbb{E}_{q(z|x)}[\log q(z|x)] - \mathbb{E}_{q(z|x)}[\log p(z|x)] \\ &= \mathbb{E}_{q(z|x)}[\log q(z|x)] - \mathbb{E}_{q(z|x)}\left[\log \frac{p(x, z)}{p(x)}\right] \\ &= \mathbb{E}_{q(z|x)}[\log q(z|x)] - \mathbb{E}_{q(z|x)}[\log p(x, z)] + \log p(x) \end{aligned}$$

This gives us the identity:

$$\text{KL}(q(z|x) \| p(z|x)) = \mathbb{E}_{q(z|x)}[\log q(z|x) - \log p(x, z)] + \log p(x)$$

Rearranging terms we obtain:

$$\mathbb{E}_{q(z|x)}[\log p(x, z) - \log q(z|x)] = \log p(x) - \text{KL}(q(z|x) \| p(z|x))$$

Since the KL divergence is non-negative, the left-hand side forms a lower bound on the marginal log-likelihood. This quantity is known as the **Evidence Lower Bound (ELBO)** (Patacchiola, 2021):

$$\text{ELBO}(q) = \mathbb{E}_{q(z|x)}[\log p(x, z) - \log q(z|x)] = \mathbb{E}_{q(z|x)}\left[\log \frac{p(x, z)}{q(z|x)}\right]$$

We now unpack this expression further to obtain a more interpretable form:

$$\begin{aligned} \text{ELBO}(q) &= \mathbb{E}_{q(z|x)}[\log p(x, z) - \log q(z|x)] \\ &= \mathbb{E}_{q(z|x)}[\log p(x|z) + \log p(z)] - \mathbb{E}_{q(z|x)}[\log q(z|x)] \\ &= \mathbb{E}_{q(z|x)}[\log p(x|z)] + \mathbb{E}_{q(z|x)}[\log p(z)] - \mathbb{E}_{q(z|x)}[\log q(z|x)] \\ &= \mathbb{E}_{q(z|x)}[\log p(x|z)] - \text{KL}(q(z|x) \| p(z)) \end{aligned}$$

This final form of the ELBO highlights two competing forces during training: the first term pushes the decoder to reconstruct the input well given the latent

variable, while the second term encourages the approximate posterior to stay close to the prior. The second term, because of the negative sign, drives $q(z|x)$ to match $p(z)$, hence regularizing the latent space and making sampling and interpolation meaningful.

We are now in position to introduce variational autoencoders.

A Variational Autoencoder (VAE) (Kingma and Welling, 2022) is a generative model that implements variational inference using neural networks. The encoder network outputs parameters of a diagonal Gaussian $q_\phi(z|x)$ —that is, a mean $\mu(x)$ and standard deviation $\sigma(x)$ —representing the distribution over the latent space. This is in contrast to a standard autoencoder, which instead outputs a reconstruction attempt of the input.

The VAE is trained by maximizing the ELBO, now written in its parameterized form:

$$\mathcal{L}(x; \theta, \phi) = \mathbb{E}_{q_\phi}[\log p_\theta(x|z)] - \text{KL}(q_\phi(z|x) \| p(z))$$

The first term encourages the decoder to reconstruct the input accurately from the sampled latent variable. The second term regularizes the latent space by penalizing deviations of $q_\phi(z|x)$ from the prior $p(z)$ (Kingma and Welling, 2022).

To make backpropagation through the stochastic sampling operation possible, VAEs employ the **reparameterization trick** (Kingma and Welling, 2022). Instead of sampling z directly, we sample $\epsilon \sim \mathcal{N}(0, I)$ and compute:

$$z = \mu(x) + \sigma(x) \odot \epsilon$$

where \odot denotes element-wise multiplication. This formulation allows gradients to flow through μ and σ during optimization.

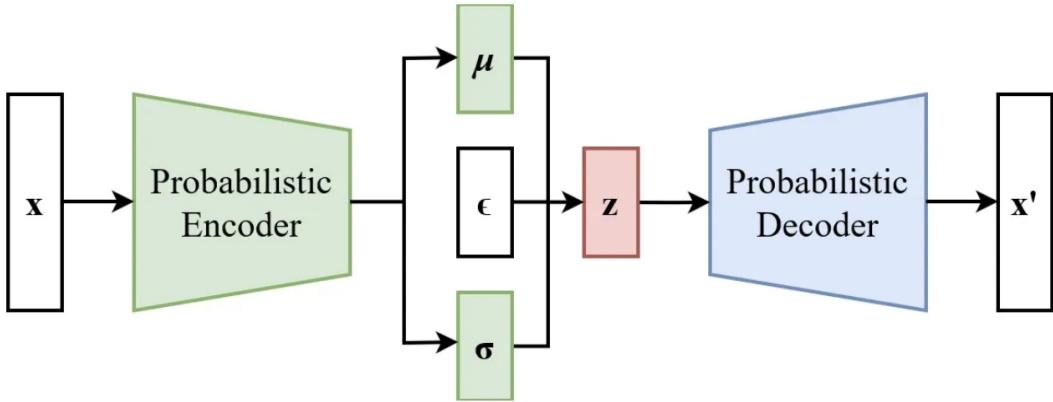


Figure 4.3: Variational autoencoder architecture (with reparameterization trick). The encoder (green) maps an input x to parameters $\mu(x)$ and $\sigma(x)$ of a Gaussian distribution $q(z|x)$ over the latent variable z . A latent sample z is drawn (by combining μ, σ with a random noise ϵ) and passed through the decoder (blue) to produce a reconstruction x' . Retrieved from <https://shorturl.at/uBd3M>.

Once trained, the decoder can generate new data by sampling $z \sim p(z)$ (in our case from a $\mathcal{N}(0, I)$) and computing $x' = p_\theta(x|z)$. The KL regularization ensures that the latent space is well-behaved, so samples drawn from the prior typically result in realistic outputs. Moreover, interpolation between latent codes leads to smooth transitions in the data space. These properties make VAEs powerful tools for generative modeling in domains such as images, text, and audio (Michelucci, 2022).

4.5. Conditional Variational Autoencoders (CVAEs)

This section extends the discussion on variational autoencoders by briefly introducing Conditional Variational Autoencoders (CVAEs), which, although not implemented or evaluated in this work, provide a natural conceptual extension relevant to the task at hand. The goal here is to give the reader a foundational understanding of the CVAE framework and its relevance to our task. A Conditional Variational Autoencoder (CVAE) is an extension of the VAE that allows us to introduce conditioning information into the encoding and decoding process (Sohn et al., 2015). While the standard VAE is useful for modeling the underlying distribution of data, the CVAE provides additional flexibility by incorporating external information, allowing the generation process to be controlled or guided in a structured way. For example, in our case, we might be interested in generating musical notes of a specific instrument or pitch. The CVAE framework enables this by introducing a condition variable c , which is supplied to both the encoder and the decoder networks.

More concretely, the encoder models the conditional distribution $q_\phi(z | x, c)$, while the decoder models $p_\theta(x | z, c)$. The condition c can be any auxiliary information relevant to the data, such as a class label or one-hot vector. Providing c to the encoder allows the encoding of x to depend explicitly on the context, while passing c to the decoder allows the model to generate outputs that are consistent with that condition.

The training objective for a CVAE mirrors that of a VAE but includes conditioning on c throughout:

$$\mathcal{L}(x, c; \theta, \phi) = \mathbb{E}_{q_\phi(z|x,c)} [\log p_\theta(x|z,c)] - \text{KL}(q_\phi(z|x,c) \| p(z|c))$$

In practice, the prior $p(z|c)$ is often chosen to be the standard normal distribution $\mathcal{N}(0, I)$ for all c , simplifying optimization while still enabling effective conditioning.

One of the main advantages of applying a CVAE in our use case would be that it uses label information to structure the latent space more efficiently. In a standard VAE, the model might dedicate part of the latent representation to implicitly encode categorical distinctions; for instance, separating pianos from violins if those instrument types lead to major differences in the input. The CVAE, however, is explicitly told what the instrument is via the condition c , which frees up the latent dimensions to capture other relevant variations such as articulation, dynamics,

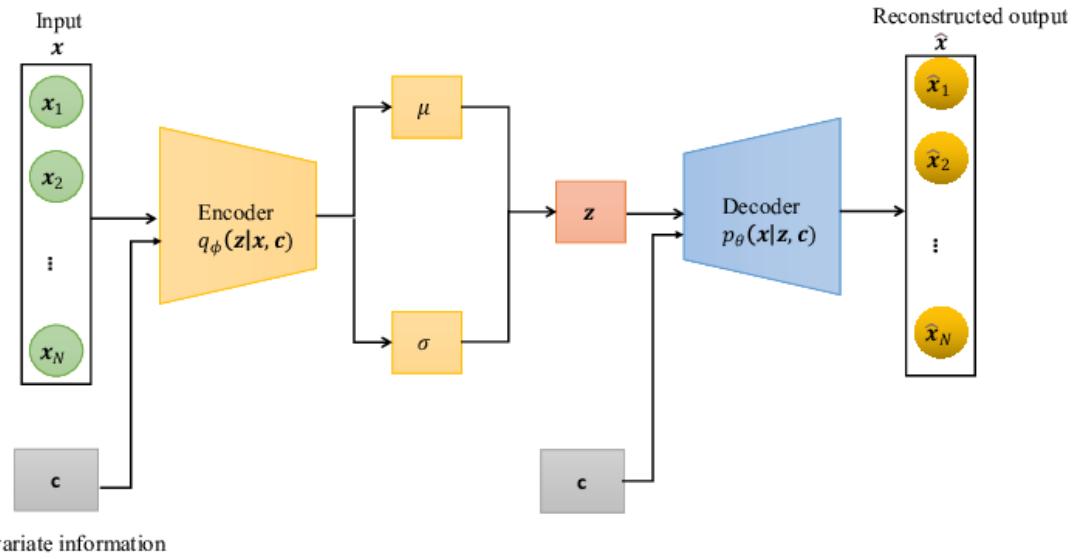


Figure 4.4: Conditional variational autoencoder architecture. The encoder (green) maps an input x and condition c to a latent distribution $q(z|x, c)$, and the decoder (blue) defines a conditional distribution $p(x|z, c)$ from which a sample x' can be drawn. Retrieved from <https://arxiv.org/abs/2211.02847>.

or timbre. This often leads to better use of the latent space and more targeted generation quality for each category (Sohn et al., 2015).

Chapter 5

Experiments and results

In this final chapter, we present the findings obtained from our experiments with the two deep learning architectures employed in this work: autoencoders and variational autoencoders.

Although our primary objective was to generate music notes using the latter, we consider it essential to also analyze and discuss the results achieved with the former. For each of the models, we dive into its design choices, training parameters and quantitative and/or qualitative results.

Before proceeding with the chapter, we would like to note that while some architectural decisions were informed by experimental results, others were more exploratory in nature, mainly drawing inspiration from existing architectures without extensive empirical justification.

5.1. Autoencoder

This section details the architecture, training setup, and results of the convolutional autoencoder used in our experiments. We begin by describing the model design and the reasoning behind key architectural choices. Next, we outline the training configuration and dataset preparation. Finally, we present both quantitative and qualitative evaluations of the model’s performance.

5.1.1. Model design and training setup

The autoencoder model employed in this work is a convolutional autoencoder consisting of three hidden layers as well as flattening/unflattening and linear layers in both the encoder and the decoder. The choice of three layers was guided by a trade-off between empirical performance and computational efficiency. In preliminary experiments, deeper models offered only marginal improvements at significantly

higher computational cost.

Internally, each convolutional layer in the encoder consists (in this order) of the convolutional functional block and ReLU activation and BatchNorm layers. These last ones weren't originally considered but helped the model stabilize training and generalization. The number of channels in convolutional block was varied across different configurations; in our final setup, we used 16, 32 and 64 channels respectively in each of the encoder's convolutional layers. The channel configuration was reversed in the decoder to mirror the encoder's architecture. Similarly, we experimented with different latent space dimensionalities and found that values in the range of 128 to 256 offered a good balance between compression and reconstruction fidelity, fixing this parameter in the end at a value of 200.

To connect the convolutional encoder to the latent vector representation, we introduced a flattening layer followed by a fully connected (linear) layer which allowed us to transform the final convolutional feature maps into a latent vector of dimension one. Symmetrically, in the decoder, we used a linear layer followed by an unflattening operation to reshape the latent vector back into a suitable set of feature maps that could be processed by the transposed convolutional layers.

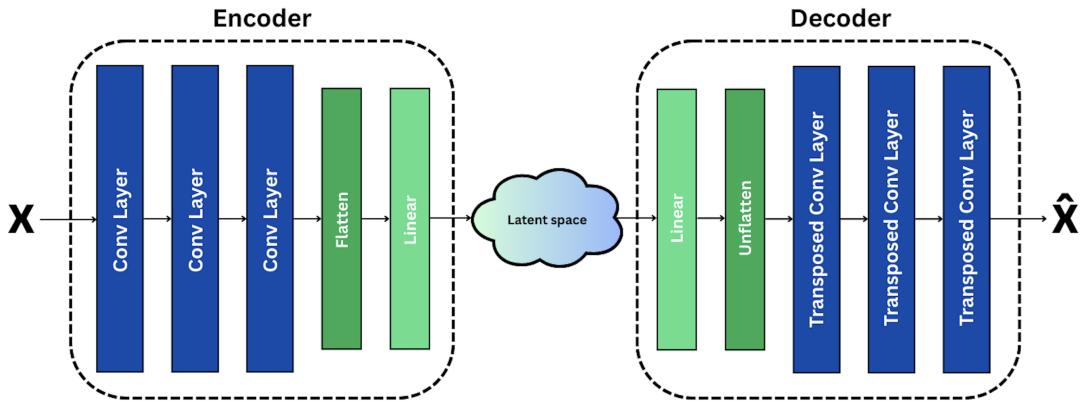


Figure 5.1: Autoencoder high level architecture. On the left, \mathbf{x} is a 4D tensor input compressed by the encoder onto the latent space and brought back by the decoder to a 4D tensor $\hat{\mathbf{x}}$ of the same shape as \mathbf{x} . Note the encoder's and decoder's symmetry.

The input to the model is a 4D tensor of shape $[B, C, H, W]$, where B is the batch size, C the number of channels, and H and W the height and width of the time-frequency representation of the audio. Initially, we used mel spectrograms as input — treating them as 2D grayscale images with a single channel, i.e., $C = 1$. However, this approach resulted in poor reconstructions, likely due to the loss of phase information inherent in mel spectrograms.

To address this, we switched to a richer representation by computing the STFT of the signal and explicitly separating its magnitude and phase components. These two components were then treated as separate input channels, yielding an input tensor of shape $[B, 2, H, W]$. Although reconstructions were still not perfect, this

representation retained more of the original signal’s structure and led to noticeably better performance compared to using mel spectrograms alone.

For downsampling in the encoder, we compared the use of strided convolutions and pooling layers. We observed that using a stride greater than one in the convolutional layers yielded better audio quality in the reconstructions. Our interpretation is that this improvement stems from the learnability of the stride mechanism, in contrast to fixed pooling operations. Moreover, the additional computation introduced by strided convolutions was not prohibitively high.

In the decoder, we used transposed convolutions to upsample the latent representation. These layers employed a stride of 2 to progressively reconstruct the original input shape. To ensure the final output matched the original input dimensions, we applied zero-padding at the end of the decoding process. While this introduces a small irregularity in the decoding pipeline, we noticed minimal impact on the output quality and significantly simplified the implementation. We also explored using the `output_padding` parameter in the transposed convolutions to achieve precise output sizing. However, doing so imposed strict constraints on the input dimensions: only certain input sizes would result in valid outputs without violating `output_padding` restrictions. This made experimentation less flexible, as we would have had to carefully tailor the input dimensions to the architecture.

The loss function we used is the Mean Squared Error (MSE) with the mean reduction option. MSE has been widely used in autoencoder architectures and it computes the average squared difference between the input and its reconstructed output. By using the mean reduction, we ensured that the loss was averaged over all data points in the batch. For the optimization of our autoencoder, we employed the Adam optimizer, which is popular for its efficiency and adaptive learning rate.

We also used a `ReduceLROnPlateau`¹ learning rate scheduler that monitored the validation loss during training. If the validation loss increased after a certain number of epochs, the scheduler would reduce the learning rate to mitigate this issue.

Regarding overfitting, we initially had set the model to train for 50 epochs, but we found that the validation loss usually began to increase continuously around epoch 30 (see Figure 5.2). To mitigate this problem, we decided to limit the training to 30 epochs.

We also experimented with data normalization at two different stages: directly on the raw waveform and on the STFT spectrogram. In both cases, we observed a drop in performance. Normalizing the waveform aimed to standardize the input amplitudes early on, while normalization at the spectrogram level aimed to make use of the structured frequency-time representation to try to achieve a more consistent scaling. However, neither approach led to improved results; in fact, models trained on normalized data exhibited significantly worse accuracy, with some test performances dropping as low as 1.42% (the metric used is explained in subsection

¹Unlike other schedulers that adjust the learning rate based on a fixed schedule or after a set number of epochs, `ReduceLROnPlateau` adjusts the learning rate only when the validation loss plateaus.

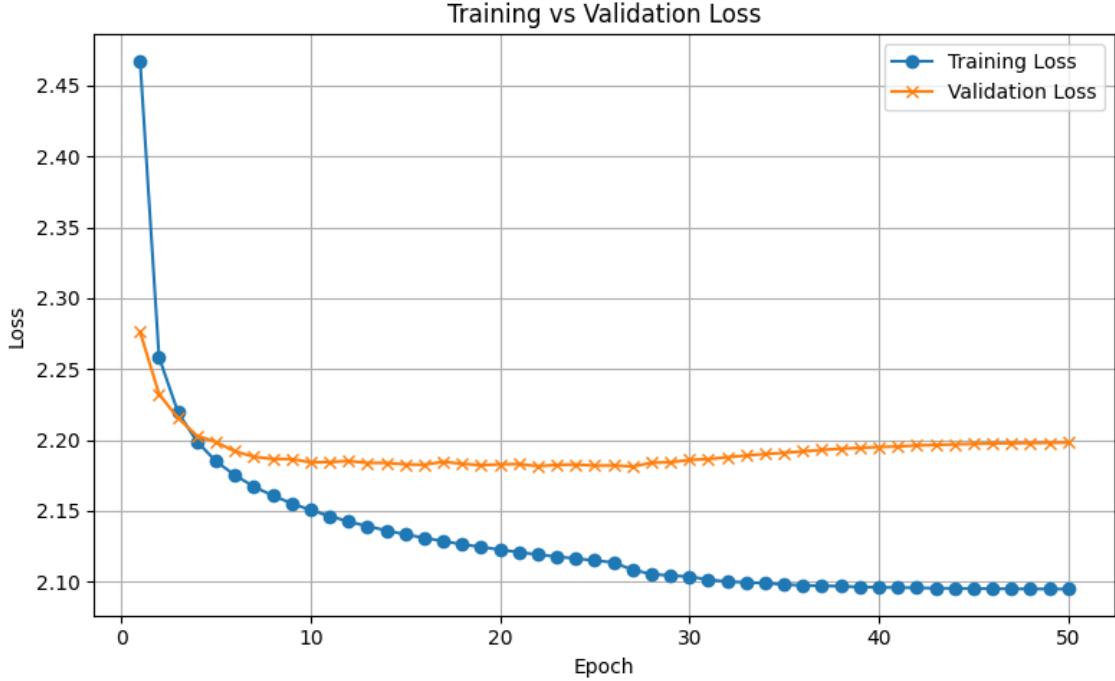


Figure 5.2: After epoch 30, the validation loss kept increasing, likely indicating model overfitting.

5.1.2).

The training was performed on CUDA-enabled GPUs, which significantly accelerated the computations and enabled us to handle large batches and complex calculations more efficiently. Specifically, the machine in which we trained the model had the following characteristics: a 12th Gen Intel Core i7-12700K processor (3.60 GHz), 64.0 GB of RAM, and an NVIDIA RTX 3090 GPU (CUDA-enabled).

The final training parameters used for the final version of the autoencoder are shown in Table 5.1.

5.1.2. Quantitative Evaluation

In this subsection, we present the evaluation of the autoencoders's performance using a specific metric applied to the test set, which we will first describe. Following that, we will include a table that displays the testing accuracies for different configurations of model parameters.

The following per-sample accuracy formula was applied to the test set:

$$\text{accuracy}_i = \max \left(0, 1 - \frac{\|x_i - \hat{x}_i\|_2^2}{\|x_i\|_2^2 + \varepsilon} \right) \times 100\% \quad (5.1)$$

where:

Parameter	Value
Train Loss	
Validation Loss	
Number of Epochs	30
Average Epoch Time (s)	
Learning Rate	
Batch Size	64
Sample Rate (Hz)	16000
FFT Size	
Hop Length	
Input Height	
Input Width	
Latent Dimension	

Table 5.1: Training Parameters

- x_i is the input for sample i ,
- \hat{x}_i is the reconstructed output from the model,
- $\|\cdot\|_2^2$ denotes the squared L2 norm,
- ε is a small constant added to prevent division by zero.

The denominator $\|x_i\|_2^2 + \varepsilon$ plays a crucial role in normalizing the reconstruction error relative to the energy of the original input. This normalization makes the metric scale-invariant, meaning that inputs with different magnitudes are evaluated fairly.

Observe that when $x_i = \hat{x}_i$, the reconstruction is perfect, and the accuracy for that sample reaches 100%. On the other hand, when the input and output are significantly different — that is, when the reconstruction error $\|x_i - \hat{x}_i\|_2^2$ is larger than the normalization term $\|x_i\|_2^2$ — the fraction in the formula can exceed 1. In such cases, the resulting accuracy becomes negative. To prevent this and keep the interpretation consistent (as a percentage between 0% and 100%), we clip negative accuracies and define the minimum accuracy as 0.

Once the per-sample accuracy formula is defined, we can simply define the total accuracy as the mean over all accuracies:

$$\text{accuracy} = \frac{1}{N} \sum_{i=1}^N \text{accuracy}_i \quad (5.2)$$

where N is the total number of samples evaluated.

A comparison of testing accuracies based on the model's configuration can be observed in Table 5.2.

Row	T loss	V loss	Epc	T	lr	FFT	Hop	Ht	Wd	Lat.	Acc.
1	0.25	0.30	50	12.3	44100	1024	256	64	64	128	87.5%
2	0.20	0.28	100	11.5	44100	2048	512	64	64	128	89.2%
3	0.15	0.22	150	10.8	48000	2048	512	128	128	256	91.1%
4	0.30	0.35	50	15.0	44100	1024	256	64	64	128	85.3%
5	0.18	0.24	200	9.5	44100	2048	512	64	64	128	90.4%

Table 5.2: Relevant training and transformation parameters and corresponding testing accuracy. The columns represent the following: T loss: Training loss, V loss: Validation loss, Epc: Number of epochs, T: Average time per epoch in hours, lr: Learning rate, FFT: FFT size, Hop: Hop length, Ht: Input height, Wd: Input width, Lat.: Latent dimension, Acc.: Testing accuracy.

Two noteworthy implementation details emerged during experimentation. The first relates to PyTorch’s `DataLoader` and its `num_workers` argument. When set to 8, training times increased drastically—up to 40 minutes per epoch. Upon investigation, this slowdown appeared to stem from the overhead of spawning multiple worker processes, likely due to the specific configuration of our development environment. Setting `num_workers` to 0 resolved the issue and restored reasonable training speeds.

The second observation concerns the use of batch normalization. Prior to introducing BatchNorm layers, the autoencoder achieved a test accuracy of approximately 51%. After incorporating BatchNorm and training on larger input sizes, performance improved significantly, with testing accuracy reaching up to 71%.

FILL TABLE CORRECTLY!!

5.1.3. Qualitative evaluation

Listening to the reconstructed audio samples revealed one consistent artifact worth noting: many reconstructions, particularly those corresponding to low-frequency inputs, tend to contain noticeable background noise, while higher-pitched sounds tended to be reconstructed with greater clarity and less audible noise. A possible explanation for this behavior is that these higher frequency sounds are distributed across a broader set of STFT frequency bins and can therefore be detected better by the model.

Perhaps a more insightful evaluation of the model’s performance can be gained by examining the actual audio reconstructions. Unfortunately, we can not directly do so on this document. However, we have provided several examples, which can be found in the project’s Github repository². Each folder number corresponds to the row number in Table 5.2 and contains:

1. A subdirectory named `examples` that includes pairs of original and reconstructed audio files. Each `Xr.wav` file represents the reconstruction of the

²<https://github.com/pabgarcialopez/TFG-info/tree/main/examples/AE>

original file `X.wav`.

2. The trained autoencoder model saved as `autoencoder.pth`.
3. A plot file `losses.png` showing the evolution of training and validation losses over epochs.
4. A `configs.txt` file containing the model's configuration parameters in JSON format.

TODO: input files into repo

5.2. Variational Autoencoders

This section reuses many of the architectural and training details described in Section 5.1, so they are not repeated here. Instead, we focus on the specific modifications required to implement a variational autoencoder (VAE), particularly regarding the model design, training strategy, and qualitative observations based on prior sampling. Note quantitative evaluation for VAEs wasn't performed because the model is generative in nature, and its outputs are stochastic samples from a learned distribution rather than deterministic reconstructions. As such, standard reconstruction metrics are less meaningful and not the primary way to assess VAE performance.

5.2.1. Model design and training setup

The VAE model builds on the autoencoder architecture with a few essential changes. The encoder now outputs the parameters of a diagonal Gaussian distribution over the latent space: the mean and log-variance of $q_\phi(z|x)$. This is accomplished by modifying the final layer of the encoder to produce two separate linear projections: one for the mean $\mu(x)$ and another for the log-variance $\log \sigma^2(x)$. During training, the reparameterization trick is used to draw samples from this approximate posterior in a differentiable manner:

$$z = \mu(x) + \sigma(x) \odot \epsilon, \quad \epsilon \sim \mathcal{N}(0, I).$$

The decoder remains structurally similar to that of the autoencoder, mapping the latent sample z back to the reconstructed input space.

A major departure from the autoencoder lies in the loss function. Instead of a pure mean squared error (MSE), the VAE is trained using a variational objective derived from the evidence lower bound (ELBO), as introduced in Section 4.4. The ELBO can be written as:

$$\text{ELBO} = \mathbb{E}_{q_\phi(z|x)}[\log p_\theta(x|z)] - \text{KL}(q_\phi(z|x)\|p(z)),$$

which involves two terms: a reconstruction loss and a regularization loss via the KL divergence. In our implementation, the reconstruction loss is instantiated using the MSE:

$$\mathcal{L}_{\text{recon}} = \frac{1}{n} \sum_{i=1}^n (x_i - \hat{x}_i)^2,$$

while the KL term was computed as:

$$\text{KL}(q_\phi(z|x) \| p(z)) = -\frac{1}{2} \sum_{j=1}^d (1 + \log \sigma_j^2 - \mu_j^2 - \sigma_j^2),$$

since this is the analytical form of the KL divergence between two diagonal multivariate Gaussians: the approximate posterior $q_\phi(z|x) = \mathcal{N}(\mu, \text{diag}(\sigma^2))$ and the standard normal prior $p(z) = \mathcal{N}(0, I)$.

Initially, we wrongly experimented with a weighted combination of the reconstruction and KL divergence terms:

$$\text{Loss} = \alpha \cdot \mathcal{L}_{\text{recon}} + (1 - \alpha) \cdot \text{KL},$$

where $\alpha \in [0, 1]$ controls the relative importance of the two components. This approach was later replaced based on advisor feedback, since it lacks a direct variational justification.

Instead, we moved to the standard ELBO form:

$$\text{Loss} = \mathcal{L}_{\text{recon}} - \text{KL},$$

which directly reflects the negative ELBO (as it is being minimized).

Despite this theoretical grounding, we observed that during early stages of training, the KL term would often collapse to near zero, a phenomenon known as posterior collapse. To address this, we introduced KL annealing, a technique in which the KL term is gradually introduced using a dynamic weighting factor β . This results in a β -VAE objective:

$$\text{Loss} = \mathcal{L}_{\text{recon}} - \beta \cdot \text{KL}.$$

KL annealing mitigates posterior collapse by allowing the model to focus first on accurate reconstruction before gradually enforcing latent regularization. In our experiments, β was linearly increased over a warm-up period of several epochs.

Apart from this change in the loss formulation, the rest of the training loop remained largely consistent with the autoencoder setup, including the use of the Adam optimizer, learning rate scheduling, and early stopping based on validation performance.

TALK ABOUT REMOVING THE LAST RELU ACTIVATION LAYER

To qualitatively evaluate the generative ability of the VAE, we focused on prior sampling. That is, we drew random samples $z \sim \mathcal{N}(0, I)$ from the latent prior and decoded them using the trained decoder to generate new audio outputs.

The results were mixed. While the decoder successfully generated valid audio waveforms, most of the outputs lacked significant diversity or structure—many sounded like variations of the same note or pitch. This suggests that the latent space was underutilized, or that the decoder did not fully learn a disentangled or expressive mapping from latent variables to audio space.

To gain further insight into how the latent space was being structured, we applied Principal Component Analysis (PCA) to the latent means μ produced by the encoder. PCA is a dimensionality reduction technique that projects high-dimensional data into a lower-dimensional subspace, preserving as much variance as possible in the process. In this case, we encoded 1000 real samples from the NSynth test set and extracted their latent means, which are vectors in \mathbb{R}^{200} in our setup. We then applied PCA to reduce these 200-dimensional vectors to 2D, enabling visualization of the latent structure.

The resulting plot (see Figure ??) reveals how the VAE distributes real data points in the latent space. Each point corresponds to a sample, and colors indicate different instruments. Surprisingly, instead of forming distinct clusters, the latent means appeared to collapse into a single dense region with no clear separation by instrument class. This suggests that the encoder may not have learned to represent diverse semantic features effectively, leading to an underutilized or collapsed latent space. This observation aligns with our findings from prior sampling, where generated outputs tended to lack diversity and mostly resembled minor variations of the same base sound.

INPUT IMAGE PCA

While posterior sampling (i.e., reconstructing from latent codes derived from real data) would yield more faithful reconstructions, we intentionally avoided this route in qualitative evaluation to focus instead on the VAE’s ability to generate novel data from pure noise.

Several examples of prior sampling results can be found in the project’s GitHub repository³. Each folder corresponds to a different training configuration and contains the following:

1. A directory named `examples`, which contains audio samples generated from random latent vectors.
2. The trained VAE model saved as `vae.pth`.
3. A configuration file `configs.txt` listing all relevant training and model parameters in plain-text format.
4. A file `losses.png` showing the training and validation loss curves over time.
5. A plot `PCA.png` visualizing the distribution of latent means from real data using Principal Component Analysis.

³<https://github.com/pabgarcialopez/TFG-info/tree/main/examples/VAE>

5.3. Conclusion

Conclusions and Future Work

The aim of this thesis was to explore the capabilities of three closely related deep learning models — autoencoders (AEs), variational autoencoders (VAEs), and conditional variational autoencoders (CVAEs) — in the context of musical audio data.

Conclusions

- Our autoencoder experiments achieved a testing accuracy of around 71%. While not optimal, these results were obtained using a custom architecture built from scratch, demonstrating a solid baseline for future improvements.
- The VAE model introduced probabilistic sampling, allowing for generation of new audio samples. However, prior sampling produced limited diversity: most outputs shared similar textures and pitch, suggesting an underutilized latent space.
- The CVAE model, designed to condition generation on instrument type, remains underexplored at this stage. Results are still preliminary and require further testing and tuning to assess its ability to produce class-controlled outputs.

Future Work

- Try alternative model architectures: experiment with deeper networks, different kernel sizes, channel counts, or latent dimensions.
- Investigate the latent space in more depth: for instance why VAE-generated samples tend to have the same pitch or how interpolation behaves between points.
- Use pretrained decoders.

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