

Interpolation Based Neural Audio Synthesis using Convolutional Autoencoders

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Declaration

I hereby declare and confirm that this thesis is entirely the result of my own original work. Where other sources of information have been used, they have been indicated as such and properly acknowledged. I further declare that this or similar work has not been submitted for credit elsewhere. This printed copy is identical to the submitted electronic version.

Hagenberg, June 27, 2023

Benedikt Langer, BSc

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Preface

Abstract

This should be a 1-page (maximum) summary of your work in English.

Kurzfassung

An dieser Stelle steht eine Zusammenfassung der Arbeit, Umfang max. 1 Seite. ...

Chapter 1

Introduction

Chapter 2

Related Works

Despite this technology is not that well explored and popular as in the image domain, there exist a few proposed approaches that have developed a rather good solution. Some of these approaches have proven, that with Neural Networks it is possible to generate synthesized audio up to a certain quality.

2.1 Audio Synthesis

Probably one of the most prominent solution is from Engel et al. [1]. With their work they have proposed a system that is capable of synthesizing audio as well as interpolating/morphing encoded audio data of two instruments to create new audio. Not only they have proposed a system, but also a public available Dataset called “NSynth” that contains a large scale of high quality musical notes. The latter has been used for training of this specific project. In their work regarding the synthesis, Engel et. al. developed and compared two different approaches with two different kind of networks. Nevertheless they have a similar structure, as they are both designed as Autoencoders but accept different kinds of data and thus have different components. While the one kind of network operates on time domain data the other one is trained on the spectral representation of audio samples. Throughout their work the second technology using spectrograms is referenced/used as Baseline Model as they focus on the use of so called “WaveNet Autoencoders” that are trained on continuous time signal. With using the Autoencoder-Structure they make use of its ability in learning efficient encodings of the music data. These encodings are representing essential features from the original audio. To create new sounds they take the encoded data from the embedding space of two instruments and interpolate them linearly. In addition they used the decoder part to reconstruct it back to audio data. Using this mechanisms they were able to create some new sounds which contain the characteristics of two different audio signals. The result can be explored via their online AI-Experiment called “Sound Maker”.¹ Furthermore can be mentioned at this point, that for their purpose they also created a huge dataset of audio samples (>306 000) that is open for public use.

Work by [7]

¹“Sound Maker” <https://magenta.tensorflow.org/nsynth-instrument>

2.2 Audio Style Transfer

Another approach is proposed in a paper by Ramani et al. [6]. In their approach they developed a Neural Network that is also, like the previous paper, based on an (convolutional) Autoencoder Architecture. As also the title says, they speak officially about their system as “Audio Style Transfer Algorithm”. The process of generating an audio containing characteristics of two audio signals is here slightly different as in the work of Engel et al.. as they use in order two networks, namely a transformation network and a loss network. Both networks have the same structure and composition of layers. The loss network is trained to compress input spectrograms to lower dimensions which means that the encoder part learns to preserve the high level features of the input. In addition the decoder learns to reconstruct from the compressed data a spectrogram similar to the input of the network. For the training of the transformation network, the pre-trained weights of the loss network are used which speeds up training (just optimization towards low level features/style). Having the trained transformation network, it then is able to transform an input spectrogram into a stylised spectrogram. The loss network is subsequently used to calculate the style-loss but also content-loss between the respective spectrograms and the output from the transformation network. This loss gets minimized by back- propagation to the transformation network. By this procedure it is possible to pass a single spectrogram through the transformation network which in order outputs a new spectrogram containing the characteristics of itself (content) but also of one other style audio. To be also mentioned due to its architecture it also performs really fast and could be used for real-time use.

Verma et al. presented in their paper in 2018 a new machine learning technique for the purpose of generating novel sounds [8]. In this approach they tried to apply the method for artistic image style transfer to audio where they specifically mentioned the approach proposed by Gatys et al.[2] (see section 2.3). Therefore they adapted and trained an AlexNet architecture on the classification of audio-samples. This kind of network is a so called convolutional neural network, whereas the audio therefore gets converted into spectrograms, as those can be seen as grey-scale images. An important note here is that in this work they used the log-magnitude data of the STFT output. Also to mention, they adapted the network to use a smaller receptive size (kernel) of 3x3 instead of the larger ones in the original network, as they claim that it retains the resolution of the audio. As in the image domain the stylised output image gets initialized with random noise, they also use here an input spectrogram consisting of a gaussian noise signal. The random noise spectrogram afterwards gets iteratively optimised by minimizing the content- but also style loss via back-propagation. In the end this process creates a spectrogram containing the content of one audio with the style of one other audio sample. They also found out that including additional loss terms for temporal and frequency energy envelopes, helped to improve the quality, as otherwise temporal dynamics would not get incorporated. For their experiments they imposed the style of a tuning fork onto a harp sound and also transferred the style of a violin sound onto a sample of a singing voice. In this way they developed a novel method for achieving cross-synthesis by using image style transfer methods.

More work in that field is coming from *Liu et al.* [5] which also explored the application of technologies given from the image domain for “mixing audio”. This also means, that this approach focuses on using audio as spectrograms. As the previous work solely investigated on the one technique by Gatys et al. this one explores two more approaches in addition to compare the results. While one of those two additional is inspired by Johnson et al. which is a convolutional autoencoder coupled with a VGG classification network the other one uses an approach with GAN (Generative Adversarial Network). In their work they called Gatys’ approach specifically slow transfer, as the iterative computation from gaussian noise was proven really slow. In contrast to the previous work by Verma et al., they used for the “slow transfer” method an adapted VGG network (1 input channel in first layer instead of 3) which has also been used in Gatys’ image style transfer. The transfer process is also similar to the previous work, as they use a spectrogram initialized as gaussian noise to iteratively minimize the content loss (in the higher layers) and the style loss (lower layers). Using this one as baseline model, they also adapted a faster style transfer method as coupling the VGG network with a convolutional autencoder network. The purpose of this network is to take as input the content spectrogram and outputting a spectrogram containing also the style features of a style spectrogram. Comparing it to other approaches this is very similar to the one of Ramani et al. having a transformation network. The only difference is the second network as here they are using a VGG classification network and no second autoencoder. Having the output of the autoencoder network (also called generative network) this one is the initial spectrogram on which the content and style loss gets computed in the VGG network (just like previously with gaussian noise). The gradient descent then will get applied to the autoencoder network, resulting after few iterations, in a stylised spectrogram. They have proven that this approach is way faster than the one with gaussian noise. As already mentioned before, for the third experiment they adapted a cycleGAN to accept audio spectrograms instead of images. In the image domain this kind of network is able to apply style transfer to only a portion of the input images. Also when using this method, two new sounds are calculated (in both directions). They also mentioned, that this approach generates the results in a shorter amount of time. For comparison, they listen to the outcome but also apply objective mechanisms like visual assessment of spectrograms, consistency tests with classification and examination of signal clusters. With the baseline approach e.g. the harmonic is not clear and high frequencies are discarded, also the faster transfer emphasizes on lower frequencies but is missing out on beginnings of the notes. With cycleGAN also the lower frequencies get emphasized while higher ones get discarded. The listenable results of each approach are provided online.²

As the already mentioned approaches are working on single notes/sounds, the work of *Grinstein et al.* has been implemented for whole audio samples [3]. Within their work they were adapting several other approaches with neural networks from the image domain for his idea. Besides of neural networks, they also implemented a handcrafted sound texture model which got compared to the neural approaches. The latter one is composed of three sound processing steps, that in combination emulates the human

²<https://www.xuehaoliu.com/audio-show>

auditory system. Taking a closer look on their approach, especially with the neural networks, it can be said that it differs in several ways. On the one hand they do not use a random noise spectrogram, moreover they already use the content spectrogram which then gets stylised through their methods.

2.3 Image Style Transfer

<In this section briefly mention the approaches of Gatys and Johnson for image style transfer and it's relation to audio style transfer [2, 4]>

Chapter 3

Approach

Chapter 4

Experiment

Chapter 5

Results

Chapter 6

Discussion/Evaluation

Chapter 7

Conclusion

Chapter 8

Future Work

Appendix A

Technical Details

Appendix B

Supplementary Materials

List of supplementary data submitted to the degree-granting institution for archival storage (in ZIP format).

B.1 PDF Files

Path: /

thesis.pdf Master/Bachelor thesis (complete document)

B.2 Media Files

Path: /media

*.ai, *.pdf Adobe Illustrator files

*.jpg, *.png raster images

*.mp3 audio files

*.mp4 video files

B.3 Online Sources (PDF Captures)

Path: /online-sources

Reliquienschrein-Wikipedia.pdf **WikiReliquienschrein2022**

Appendix C

Questionnaire

Appendix D

LaTeX Source Code

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