

Emulating LA-2A Optical Compressor With a Feed-Forward Digital Compressor Using the Newton-Raphson Method

Chin-Yun Yu* and György Fazekas

Centre for Digital Music, Queen Mary University of London, UK, chin-yun.yu@qmul.ac.uk

Abstract— This paper presents a method to emulate the LA-2A optical compressor using a feed-forward digital compressor. The Newton-Raphson algorithm is used to find the optimal parameters that minimise the differences between the actual recordings and the compressor’s output. We build an LA-2A emulation plugin using the learnt parameters, serving as a creative tool for music producers.

Index Terms— Virtual analog modelling, Newton-Raphson, feed-forward compressor, differentiable DSP

I. INTRODUCTION

Compressors control audio dynamic range and are often used in music production and broadcasting. Early analog compressors, such as the popular LA-2A by Universal Audio, have iconic sounds that have attracted considerable interest in replicating their characteristics in the digital world, particularly as an audio plugin inside a DAW.

This paper explores modelling the LA-2A (its peak reduction parameter specifically) by matching its sound using a simple feed-forward digital compressor. Although the expressiveness is constrained by the standard five parameters (threshold, ratio, make-up gain, attack, and release), the learnt mapping ($\mathbb{R} \rightarrow \mathbb{R}^5$) can give us a further understanding of this classic unit and enables more creative control than just replicating the sound.

II. METHODOLOGY

Given an input signal \mathbf{x} , we aim to match the sound of a feed-forward compressor $\hat{\mathbf{y}} = f_{\mathbf{x}}(\theta)$ to the real recordings \mathbf{y} of LA-2A by finding the optimal parameter values that minimise the L_2 distance between them as $\theta^* = \min_{\theta} \mathcal{L}(\theta)$ where $\mathcal{L}(\theta) = (\mathbf{y} - \hat{\mathbf{y}})^T (\mathbf{y} - \hat{\mathbf{y}})$. Both $\mathbf{x}, \mathbf{y} \in \mathbb{R}^N$ are from the SignalTrain dataset [1], and $\theta \in \mathbb{R}^5$ are the parameters.

Assuming \mathcal{L} is convex, we can get θ^* using the Newton-Raphson method, which performs the following step repeatedly until convergence:

$$\theta \leftarrow \theta - [\nabla^2 \mathcal{L}(\theta)]^{-1} \nabla \mathcal{L}(\theta). \quad (1)$$

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We use the differentiable implementation of $f_{\mathbf{x}}$ by Yu et al. [2] to calculate both the gradients ($\nabla \mathcal{L}(\theta) \in \mathbb{R}^5$) and the Hessian matrix ($\nabla^2 \mathcal{L}(\theta) \in \mathbb{R}^{5 \times 5}$) in PyTorch.

III. RESULTS

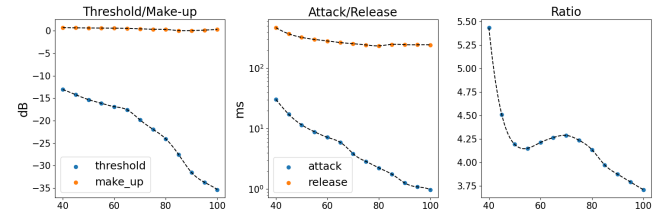


Figure 1: The resulting mapping from LA-2A peak reduction (x-axis) to the compressor’s threshold, attack/release time, ratio, and make-up gain.

We show the learnt parameters with peak reduction ranging from 40 to 100 with a spacing of 5 in Fig. 1. The five parameters generally follow some smooth trajectories that can be approximated easily. For example, the threshold is approximately linear, and the attack/release varies exponentially to peak reduction.

We build an LA-2A emulation plugin using APE [3] that runs our feed-forward compressor under the hood based on the above results¹. We linearly interpolate the parameters for peak values not included in the training set.

IV. CONCLUSION AND FUTURE WORK

This paper demonstrates that the behaviour of LA-2A can be described and controlled by the parameters of a feed-forward compressor using the Newton-Raphson method. We plan to publish the resulting model as a standalone plugin using JUCE and evaluate different interpolation methods for the LA-2A’s peak reduction parameter.

V. REFERENCES

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- [3] J. L. Thorborg, “Audio programming environment,” 2021, <https://www.jthorborg.com/apex.html> [Accessed: (2024-11-20)].

¹<https://github.com/aim-qmul/4a2a>