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An Automated Approach to the Application of Reverberation

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ABSTRACT

The field of intelligent music production has been growing over recent years. There have been several different approaches to automated reverberation. In this paper, we automate the parameters of an algorithmic reverb, based on analysis of the input signals. Literature is used to produce a set of rules for the application of reverberation, and these rules are then represented directly as direct audio feature. This audio feature representation is then used to control the reverberation parameters, from the audio signal in real time.

1 Introduction

The application of reverberation is a vital aspect of producing a mix. Reverberation can be used to give the tracks space, tie them together and add ambience and depth to the audio mix [1]. The creative intention of an engineer is paramount to the ability to produce a pleasant and balanced drum space and image. An overview of automatic mixing approaches is presented in [2].

[3] performed a study of reverb matching a reference source, and identified that shorter reverbs are more difficult to hear and need to be louder to be clearly identified. [4] investigated the reverb parameters and the impact of pre-delay on the perceived loudness of the reverb. [5] demonstrated that more reverb can influence the perception of a sound, making it seem more natural and spatially wider. [6] studied user preferences of artificial reverb parameters, with varying musical content, while [7] analysed a set of multitrack mixes to determine preference of reverb level.

[8] developed an approach for automatic subgrouping of audio tracks as in [9] a survey was performed with ten distinguished professional mix engineers, in which the use of subgrouping and reverb. It was identified reverb is commonly applied to a subgroup or mix bus [9]. There are a number of approaches to automatic mixing and automatic parameter setting of audio effects [10]

There have been several approaches to producing an automated reverb. A probabilistic soft logic approach is taken by [11], where rules are defined, in terms of audio feature representation with a logical probability, and then a constraint optimisation approach is taken to identify suitable reverb settings. Furthermore, the authors identify that the reverberation applied was not ideal, and that considerable further work was required. [12] produced a system for training a system to apply a given set of reverb parameters, based on audio feature extraction [13], and on a user interaction. This was further extended in [14], to include a more general approach to parameter mapping.

It is clear that the automated application of reverberation is little studied field. Despite reverberation being one of the oldest and most important audio effects [15, 7]. Reverberation can be used in a number of cases outside of music production, such as making sound effects more realistic [16].

[17] reported that a musician performing in a space with a different reverb time (RT60), will naturally modify their performance speed to time with the reverb. This principal will be developed further. This paper presents an approach to the automation of reverberation parameters, specifically for percussive instruments. Through the use of the tempo of a percussive track, we select and will select and apply an appropriate reverberation RT60. This work was developed further, with a considerably larger dataset in [18], and the results agree with the initial paper.

2 Reverberation Mixing Rules

In order to construct a mapping between audio effect parameters and attributes of the audio, a set of audio rules were taken from literature, as presented in Table 1.

A typical and well known reverb designed by Dattorro will be taken [22], and the parameters are automated, based on a set of rules drawn from literature [23]. The Dattorro reverberation unit was used with the following control parameters:

- Gain
- Pre-delay
- Diffusion
- Tail Decay
- High Frequency Damping
- High Frequency Cut

3 Controlling Reverberation Parameters

From these rules, a number of links were made between parameters and low level attributes. It was identified, through experimentation and analysis on test signals, it was identified that the *diffusion* and *tail decay* parameters are the two parameters that impact the RT60. The RT60 is defined as the period of time taken for an input impulse to decay by $-60dB$.

Rule 1 and 2 identify that the tempo can be used to parameterise the RT60 of the reverberation. The *diffusion* and *tail decay* parameters can be used to control the RT60 directly. Rule 7, in Table 1, identifies that the *pre-delay* parameter can be controlled from the track tempo. Furthermore, a constraint that the Since Rule 9 and 10 identify that low frequencies and transients are less tolerant to reverberation, the *HF damping* and *Gain* parameters can be controlled by a measure of the transient nature of a track. In this case, we will use the Crest Factor [24]. A MATLAB VST implementation [25] of the full project, along with some example pieces of audio content are all available online.[1]

3.1 Mapping Tempo to preferred RT60

[17] demonstrated that the RT60 of a space will influence the appropriate tempo of a musical performance. Using the data, provided by [17], a Pearson correlation was performed. The mean tempo of the performance and the RT60 of the space, at 1kHz, are strongly correlated $r = -0.89$ with a $p = 2.5 \times 10^{-3}$.

As such a linear regression was performed to identify the link between tempo and RT60, such that

$$F(x) = p1 * x + p2 \tag{1}$$

Where, $p1 = -1.1$ and $p2 = 136.1$

3.2 RT60 to Diffusion and Tail Decay

Within this reverberator, it was identified that the *diffusion* and *tail decay* parameters are the two parameters that impact the RT60. As such, we perform an optimisation approach on these two parameters, to select the appropriate parameters, based on the intended RT60, and the track tempo. A number of test signals were passed through the reverberator, and the results sampled and analysed to identify the impact that each of

[1] https://github.com/djmoffat/AutoReverb

Mixing Rule	Ref.
Mixing Rule 1. *There is a strong correlation between tempo of a song and RT60.*	[17]
Mixing Rule 2. *A slower song will require a longer reverb.*	[17]
Mixing Rule 3. *Typical reverb times (RT60) will not be longer than around 3s.*	[15]
Mixing Rule 4. *It is better to err on the side of too little reverb, rather than too much.*	[19]
Mixing Rule 5. *For a higher perceived amount of reverberation, increase the reverb loudness and/or reverb time.*	[20]
Mixing Rule 6. *Reverb time is strongly dependent to an autocorrelation measure.*	[20]
Mixing Rule 7. *The pre-delay is timed as a multiple of the subdivided song tempo.*	[20, 21]
Mixing Rule 8. *The pre-delay should be over the Haas fusion point*	[20]
Mixing Rule 9. *Low-end frequencies are less tolerant of reverb and delay.*	[20]
Mixing Rule 10. *Transients are less tolerant of reverb and delay.*	[20]
Mixing Rule 11. *The sends into the reverbs should be equalised.*	[20]
Mixing Rule 12. *The level of the reverb returns is on average set to a specific amount of True loudness lower than the direct sound.*	[20]

Table 1: Set of mixing rules used to automate reverberation

the parameters on the output RT60. The equations were solved, using a MATLAB package to produce the parameter mapping as follows

$$\begin{aligned} RT60 =& 942.4 + 0.00002e^{(20354*decay)} \\ &+ 16170e^{(-6.446*diffusion)} \end{aligned} \tag{2}$$

This means that the parameters can be iteratively set to the given value, through the use of an optimisation approach, with the following equations 3 and 4.

$$\text{decay} = \frac{\log(\frac{\alpha + \sigma e^{\kappa \text{diffusion}} - RT60}{\beta})}{\gamma} \tag{3}$$

$$\text{diffusion} = \frac{\log(\frac{(\alpha + \beta e^{\gamma \text{decay}} - RT60}{\sigma}}{\kappa} \tag{4}$$

where

$$\begin{aligned} \alpha &= 942.4 \\ \beta &= 2.004 \times 10^{-5} \\ \gamma &= 20.54 \\ \sigma &= 1.617 \times 10^{4} \\ \kappa &= -6.446 \end{aligned}$$

3.3 Pre-Delay

Given the Rules 7 and 8, the pre-delay parameter is controlled directly by the smallest beat interval that is considered to be within the threshold of the haas fusion point. The haas point, otherwise known as the precedence effect is the limit as to which two repetitions of the same audio source is distinguishable as two separate events [26]. This is typically considered to be around 30ms, for most types of audio content. As such the pre-delay parameter was set as the largest fraction of the best interval that is greater than 30ms.

$$\text{Pre-Delay} = \frac{\frac{60}{bpm}}{2^{x}} \tag{5}$$

Such that $x \in \mathbb{N}$, and $Pre-delay > 0.03$.

3.4 Gain

The reverb *gain* parameters are dictated by the level of reverb already existing in the audio track, with a specific target reverb rate to the entire track. As such, the is inversely proportional to the crest factor.

$$gain = \frac{1}{crest^{3}} - DRR \tag{6}$$

where *DRR* is the direct to reverberant ratio of the incoming audio signal, as calculated with the method presented in [27]. This is an important part of the process, as if an audio track is already somewhat reverberant, it would not be preferable to add additional reverb to the audio signal.

3.5 HF Damping

High frequency damping is controlled by the transient nature of the audio track, particularly due to the length of the reverb tail having a large impact on the perception the reverb loudness [19]. As such, the high frequency damping is controlled by the following equation

$$HighFrequencyDamping = \frac{1}{crest^{2}} \tag{7}$$

3.6 High Frequency Cut

Informal testing, along with Rule 11, dictated that *high frequency cut* parameter should be set at a fixed value of 5000Hz.

4 Discussion

An novel approach to the automation of reverberation parameters, based on analysis of the input audio signal, is presented. The signal processing approach will take the content of the audio signal and determine an appropriate set of reverberation parameters for the incoming audio content. Where an incoming audio signal contains considerable reverberation, less reverb will be applied, and where the signal is more transient, less reverberation is applied. Transient content will also contain more high frequency damping, this is vital, as in some informal tests, it was discovered that high frequency reverberation on broadband transient sounds, such as a snare drum, was fairly unpleasant.

The mapping of two reverberation parameters, diffusion and tail decay, to the RT60 of the reverberation was an important aspect of this reverberator, as it allows for a more conceptual understanding as to exactly what properties of the reverb are being controlled. In parallel with the RT60, there is also the scope to control the perceived brightness of the reverb at this time, which will allow for the optimisation of these two parameters to be performed to a specific target. This was investigated, as part of this work, but no target parameters could clearly identified as suitable for generic audio content, and as such this has been left available in the implementation, but a detailed discussion is out of the scope of this paper.

There are a number of approaches or methods that could be used to considerably improve this automatic reverberator. A data drivemachine learning approach could be taken, such as that described in [28].

It is clear that further development of the timbral aspects of the reverberator are necessary, to produce the preferred tonality, allowing for a semi-autonomous control approach to a reverberator. Furthermore, a formal subjective listening experiment, rather than the informal approach as part of implementation, should be taken to verify the results of this work, and to confirm that this approach is both useful and can produce suitable application of reverberation.

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References

[1] Izhaki, R., *Mixing audio*, Focal Press, 2017.

[2] De Man, B., Reiss, J. D., and Stables, R., "Ten Years of Automatic Mixing," in *3rd AES Workshop on Intelligent Music Production*, 2017.

[3] Gardner, W. and Griesinger, D., "Reverberation level matching experiments," in *Proceedings of the Sabine Centennial Symposium*, 1994.

[4] Griesinger, D., "How loud is my reverberation?" in *Audio Engineering Society Convention 98*, Audio Engineering Society, 1995.

[5] Kaczmarek, M., Szmal, C., and Tomczyk, R., "Influence of the Sound Effects on the Sound Quality," in *Audio Engineering Society Convention 106*, Audio Engineering Society, 1999.

[6] Pestana, P. D., Reiss, J. D., and Barbosa, Á., "User preference on artificial reverberation and delay time parameters," *Journal of the Audio Engineering Society*, 65(1/2), pp. 100–107, 2017.

[7] De Man, B., McNally, K., and Reiss, J. D., "Perceptual evaluation and analysis of reverberation in multitrack music production," *Journal of the Audio Engineering Society*, 65(1/2), pp. 108–116, 2017.

[8] Ronan, D., Moffat, D., Gunes, H., and Reiss, J. D., "Automatic Subgrouping of Multitrack Audio," in *Proc. 18th International Conference on Digital Audio Effects (DAFx-15)*, DAFx-15, 2015.

[9] Ronan, D., Gunes, H., and Reiss, J. D., "Analysis of the subgrouping practices of professional mix engineers," in *Audio Engineering Society Convention 142*, 2017.

[10] Moffat, D. and Sandler, M. B., "Approaches in Intelligent Music Production (In Review)," *Arts*, 2019.

[11] Benito, A. L. and Reiss, J. D., "Intelligent multitrack reverberation based on hinge-loss markov random fields," in *Audio Engineering Society Conference: 2017 AES International Conference on Semantic Audio*, 2017.

[12] Chourdakis, E. T. and Reiss, J. D., "Automatic control of a digital reverberation effect using hybrid models," in *Audio Engineering Society Conference: 60th International Conference: DREAMS (Dereverberation and Reverberation of Audio, Music, and Speech)*, Audio Engineering Society, 2016.

[13] Moffat, D., Ronan, D., and Reiss, J. D., "An Evaluation of Audio Feature Extraction Toolboxes," in *Proc. 18th International Conference on Digital Audio Effects (DAFx-15)*, 2015.

[14] Chourdakis, E. T. and Reiss, J. D., "A Machine-Learning Approach to Application of Intelligent Artificial Reverberation," *Journal of the Audio Engineering Society*, 65(1/2), pp. 56–65, 2017.

[15] Reiss, J. D. and McPherson, A., *Audio effects: theory, implementation and application*, CRC Press, 2014.

[16] Mengual, L., Moffat, D., and Reiss, J. D., "Modal Synthesis of Weapon Sounds," in *Proc. Audio Engineering Society Conference: 61st International Conference: Audio for Games*, Audio Engineering Society, London, 2016.

[17] Weaver, J., Barthet, M., and Chew, E., "Analysis of Piano Duo Tempo Changes in Varying Convolution Reverberation Conditions," in *Audio Engineering Society Convention 145*, Audio Engineering Society, 2018.

[18] Weaver, J., Barthet, M., and Chew, E., "Filling The Space: The Impact of Convolution Reverberation Time on Note Duration and Velocity in Duet Performance," in *Audio Engineering Society Convention 147*, Audio Engineering Society, 2019.

[19] Paulus, J., Uhle, C., and Herre, J., "Perceived level of late reverberation in speech and music," in *Audio Engineering Society Convention 130*, Audio Engineering Society, 2011.

[20] Pestana, P. D. L. G., *Automatic mixing systems using adaptive digital audio effects*, Ph.D. thesis, Universidade Católica Portuguesa, 2013.

[21] Owsinski, B., *The Mixing Engineer's Handbook*, Nelson Education, 2013.

[22] Dattorro, J., "Effect design, part 1: Reverberator and other filters," *Journal of the Audio Engineering Society*, 45(9), pp. 660–684, 1997.

[23] Moffat, D., Thalmann, F., and Sandler, M. B., "Towards a Semantic Web Representation and Application of Audio Mixing Rules," in *Proceedings of the 4th Workshop on Intelligent Music Production (WIMP)*, Huddersfield, Uk, 2018.

[24] Peeters, G., "A large set of audio features for sound description (similarity and classification) in the CUIDADO project," Technical report, IRCAM, 2004.

[25] DeVane, C. and Bunkheila, G., "Automatically Generating VST Plugins from MATLAB Code," in *Audio Engineering Society Convention 140*, Audio Engineering Society, 2016.

[26] Litovsky, R. Y., Colburn, H. S., Yost, W. A., and Guzman, S. J., "The precedence effect," *The Journal of the Acoustical Society of America*, 106(4), pp. 1633–1654, 1999.

[27] Jeub, M., Nelke, C., Beaugeant, C., and Vary, P., "Blind estimation of the coherent-to-diffuse energy ratio from noisy speech signals," in *2011 19th European Signal Processing Conference*, pp. 1347–1351, IEEE, 2011.

[28] Moffat, D. and Sandler, M. B., "Machine Learning Multitrack Gain Mixing of Drums," in *Audio Engineering Society Convention 147*, 2019.