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Settings of Reverb Processors from the Perspective of Room Acoustics

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The issues of reverberation in acoustic architecture and music production share the same theoretical core; nevertheless the first one has been scientifically researched in depth while the second one remains at technical and experimental crossroads. It could be stated that the ISO 3382 parameters were proposed in the "analog" era for room acoustics (actual halls) whereas the "digital" parameters introduced in software, artificial reverbs, are not standardized in any way but help to create desired reverberation for music or audio effects. The interest herein is to bind these two disciplines together and analyze some of the significant descriptors of room acoustics (RT, C50, C80, BR, ER, CT) applied in plug-in reverberant processors to observe how the virtual space is affected by changing the values of different parameters. Psychoacoustic ranges of JND were applied to conclude their relevance (or rather influence) and whether it is possible to perceive alteration. Five of the selected popular and commercial VST reverbs are juxtaposed with five similar settings and the results of analysis might be useful for sound mixers and automated mixing algorithms.

1. INTRODUCTION

Reverberating processors (reverbs) are among the most essential tools in music production. First and foremost, they are responsible for adding the sensation of room or abstract space to the recording. With close microphone techniques, high separation between instruments, and high acoustic absorption of the recording studios, there is hardly any natural reverberation registered. That very basic function initially opened up a wide range of possibilities with creative potential, such as adding depth, decay, and timbre manipulation, special effects and so on. Nowadays, there are countless reverberating processors: built into Digital Audio Workstations (DAWs), Digital Signal Processing (DSP) or native plug-ins, dedicated systems, microprocessors, and more [1]. Additionally, there are multiple settings and presets that come with every tool and are subject to change. The matter of reverberation in music production is usually treated more intuitively with experimental rather than technical approaches. Some research has been conducted on the perceptual evaluation of loudness/length of reverb and common tendencies in preferences can be found [2], but this kind of investigation is more likely to be found in room acoustics rather than sound engineering. The main aim herein is a detailed analysis of selected, popular reverberation processors together with their configuration possibilities and the relation to the parameters of architectural acoustics. Although the art of mixing is considered as balancing between technical and experimental areas, there are a number of psychoacoustical phenomena and being aware of them may help users to get on the right track quicker and save time. Preferences concerning reverberation time and other acoustical architecture parameters provided for specific music genres (especially classical) have been precisely evaluated through numerous tests [3–6]. Some, but not many, research projects were also conducted on other parameters that are more intuitive for sound engineers—such as reverb relation to the tempo of the music or preferred wet/dry ratio [2]. This can provide suggestions for mixing engineers and may lead to an improvement in their work. The wet/dry ratio is an essential parameter in the reverb perception and is included in the A-value by Ando [7] in the Theory of Subjective Preference. In room acoustics, the value of this parameter depends on the source-receiver distance and room absorption. In audio engineering, implementation of the wet/dry parameter is obvious and expressed as a relative level between the dry signal and reverberation (wet) in dB or as a fraction (sometimes also as a percentage). Therefore, the wet/dry ratio is outside the scope of this work.

Reverberation is such a complex phenomenon that it is often implemented by plug-ins or devices with a multitude of knobs, pots, sliders, and additional options and presets that often bring more confusion than help for aspiring engineers.

Architectural acoustics and sound engineering are separate scientific and technique fields, but both have one very common factor: reverberation. In the authors' experience, these two worlds function almost completely in separation. On one hand, sound engineers during the mixing process follow their own intuition and experience, especially while applying reverberation [8], [9]. On the other hand, in room acoustics, numerous parameters connected with reverberation have been defined and examined and, as already stated, there are acoustical guidelines for almost every kind of room [3], [5]. Over and above that, the excellent auditory capabilities of sound engineers are rarely employed in room acoustic design and, moreover, sound engineers (or their tools) do not use room acoustics achievements (in the mixing process) almost at all.

However, in the history of music and architecture a very strong link to interiors can be observed, especially the influence of the reverberation on music compositions and sound texture. Examples could be acoustics of traditional houses of worship, churches, cathedrals, etc.—they have all formed several trends in the evolution of choral, organ, and polyphony music in general [9], [10], [11]. The authors of the current paper in previous investigations [12] show the radically different acoustic features of wooden and brick or marble orthodox churches in Poland. Because of that, the musical traditions differ between these two types of churches within the same congregation. Also, the listener's preference of choral music in these rooms is a complex and not straightforward phenomena. There are several other cases of composers creating the pieces dedicated for particular interior or type of room [13], [14], [15], the prominent example being Bach's heritage that has formed modern theory of harmony [16]. From this angle there is a definite indirect connection between architectural acoustics and reverb processors and their settings. After all, almost every reverb processor has presets such as concert hall, dark cathedral, or live room.

The main goal of this work is to find out a contemporary relation between sound engineering and room acoustic approaches to reverberation. From a technical point of view, it is interesting to find out:

- how the reverberation processor settings influence the room acoustic parameters of produced reverbs,
- if it is possible to shape and control room acoustic parameters of reverberation using reverb processors, and
- if reverberation processors could be used to educate room acoustic designers.

Similar problems can be found in [17], where the main assumption was the automatic recreation of specific room acoustics by reverberation processors. Another work [18] examined reverb processors for listening and quality evaluation.

The second section describes reverb processor settings in general and the third focuses on processors examined in the current work. The fourth section describes the experiment and calculations that were conducted on selected reverb processor settings. The fifth section shows the results of the experiment and the last section contains the conclusions.

2. REVERB PROCESSORS AND THEIR TYPICAL SETTINGS

Reverb processors have been developed for more than half a century [1]. One can distinguish two main types of reverbs: convolution-based and algorithm-based reverbs, which include the following concepts: Schroeder's and Moorer's, Feedback Delay Network, Velvet noise based, Scattering delay networks, Gardner's, Dattorro's, and many others [1], [18]. There is also the possibility of using the simulation of sound propagation in room acoustic software, but this kind of approach is outside the scope of the current research.

The settings of the reverb processors depend mostly on the manufacturer's implementation or used algorithm, both for convolution and other algorithm-based reverbs. Most commonly they include Reverberation Time (also called Reverb Time or Decay Time), Pre-Delay, and Frequency modifications. Reverberation time (RT60) is standardized and defined as the time it takes for the sound pressure level to reduce by 60 dB, measured after the test signal is stopped. Pre-delay in most sources is defined as the time difference between the direct sound and very first reflection. Sometimes it is also explained as a gap between the dry signal and the beginning of reverberation. Reverb frequency equalization can be achieved as a pre-reverb or post-reverb or by damping within the algorithm, which relates to the physical properties of absorptive materials.

There are also other common settings that refer to the physical properties of reverberation, such as density and diffusion. While diffusion emulates rough irregular reflective surfaces and is critical to match the source material, density is dedicated to making the tails of the reverb smoother (Fig. 1).

Theoretically, the first one operates on early reflections and the second on the tail, but in practice there are plugins that split those parameters and offer separate control over 'early' and 'late' reflections of density / diffusion.

Low density leaves more space between the first and subsequent reflections, resulting in replication of a large room (together with long decay time). High density percentage emulates smaller spaces (quick and thick reflections). The time division between early discrete reflections and stochastic reverb tail could be related to the mixing time and calculated for measured or modeled room response using basic statistical estimators [19]. In mixing, decreasing density is very often desirable on vocals and sources sustained

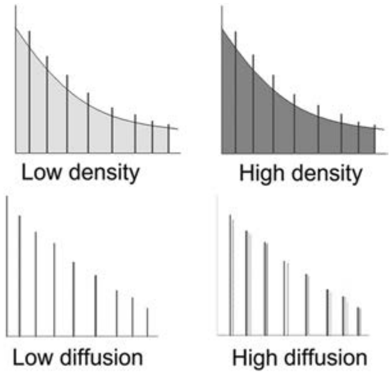

Fig. 1. Concepts of diffusion and density—comparison.

for a long time as it improves clarity, but it is not pleasant with transients, such as percussion, as it creates an impression similar to fluttering. In addition, increasing that parameter may result in a more natural sounding reverb but together with unwanted masking [7], [8].

Additionally, there are very often some other settings, such as Size, Color, Resolution, etc., that can modify and mean anything and that refer to synergy of the senses.

3. EXAMINED REVERBS AND SETTINGS

For this paper, 5 popular reverb processors were examined: IR, HR, TV, VH, and LX. This choice was made to include different types and manufacturers. Additionally, these processors were available to the authors.

3.1. IR

This is a parametric convolution processor designed to imitate real acoustical spaces. It contains a large library of over 100 various impulse responses (IRs) registered by, among others, Angelo Farina [20], [21], and includes Concert Halls such as Uhara and Sydney Opera House, Auditoriums (Atherton, Parma), and Theaters (Noh Drama, Wells Fargo). There is also the option of implementing self-measured IRs. Examined settings:

Size (range 0.25–4)—should influence dimension and volume of the room as well as relation (time-wise) between reflections,

Pre-Delay (range 0–500 ms)—defined as delay of the entire IR,

Reverb Density (range 0.25–4:00)—ratio of modified to original IR. The higher this parameter, the thicker and smoother-sounding the reverb. The result is dependent on the room's original reflections,

Reverb Resonance (range 0.25–4)—ratio of boosting or reducing any existing resonances in the original IR, and

Decorrelation Control (range 0–100%)—controls of decorrelation between channels resulting in greater or lesser spaciousness. It gives the capability of creating stereo output out of mono IR.

3.2. HR

HR is an Impulse Response-based algorithm reverb plug-in, combining the convolution of impulse response with a synthetic feedback-based reverb [22]. It comes with a library of different acoustic spaces. Maximum reverb time is 6 s and 12 s with the "long" plug-in variant. It operates in mono, stereo, and multichannel formats up to 5.1.

Examined settings:

Pre-Delay (range 0–500 ms, default 24 ms or sync to tempo by default to 3/256 of BMP)—delay of the reverb; options free,

Buildup Time (range 0–2 seconds, default 0.025)—time from early energy arrival to peak level of the reverb (the larger the space, the longer the buildup time),

Reverb Size (range 0.5–2, default 1)—controls the density of early reflections and slope of the previous parameter. The higher the value, the larger and less condensed the considered space,

ER/Select (range 1–10, default 5)—it is possible to pick one out of ten different early reflection distributions represented by signal envelopes charts, and

ER/Tail Balance (range 0–100, default 73.2)—ratio of ER to tail gain. For the value of 50, the tail gain equals the gain of ER.

3.3. TV

TV links the generator of early reflections with an algorithm-based reverb. The TV manual [23] informs that the processor, through its configuration, allows users to freely create and shape the desired room response. Early reflections are generated up to 100 ms and affected by Decay Time, Balance, and other settings. Reverb herein is the output signal of variable density of reflections to create a virtual space, affected mostly by Decay Time and Balance.

Examined settings:

Dimension (1–4, default 3)—controls the pattern of EarlyRef for simulation of the 1 to 4 dimensional space,

Room Size (50–20,000, default 5,516)—cubic meters of created space,

Balance (−48–12, default 3)—maintains a constant perceived level while changing the relative level between Reverb and Direct/EarlyRef,

Pre-Delay (range 10–300, default 88.9)—time between direct sound and Reverb; does not affect early reflections, and

Density (0–1, default 0.85)—controls the density of the initial reflections in the Reverb; the higher values have higher density.

3.4. VH

VH is a popular and typical algorithmic reverb processor [24]. It is designed to produce large spatial images with modification of density that can be adjusted from very sparse to very dense. The software has no documentation so the setting description is based on the software interface.

Examined settings:

Pre-Delay (0–70 s, default 20 ms)—delays the reverb before signal direct,
Diffusion Early (0–1, default 1)—the initial echo density,
Late Size (0–1, default 1)—defined as relative room size,
Mode (multi-choice from presets such as default Large Room, Dark Space, Nostromo, Sulfaco, Dense Room, etc.)—reverb algorithm, and
Bass Mult. (0.5x–2x, default 1)—scales reverberation time in low frequencies.

3.5. LX

LX is one of the pioneers in algorithm-based reverbs. LX is a continuation of classical hardware reverb devices [25]. It is described as a large space with low initial reflection density and low energy before 60–100 ms and lower frequencies reverberating longer.

Examined settings:

Pre-Delay (range 0–1 s, default 16 ms)—described as the time between dry and wet signals,
Diffusion (range 0–100%, default 93%)—setting responsible for smear of the signal,
Early Level – (-inf–0 dB, default -7.5 dB)—controls the level of early reflections,
Color (Dark1... – Neutral – Light...3)—controls the timbre over time, where darker values mean that high frequencies decay faster, and
Reflection (0 – 8 – 15)—allows users to select from several presets of discrete reflection combinations.

All the settings selected for in-depth examination are listed in Table 1. The range rows show the examined values and, in most cases, these were minimum, middle, and maximum of the parameter range. Only the pre-delay setting was common for all the examined processors, so for all cases it was fixed with the same values for objective comparison.

4. METHOD

In order to relate the reverb processor parameters to room acoustic estimators, a simple experiment / calculation was performed. For each selected reverb processor setting, the static impulse response h_{RP} was calculated by deconvolution of the test signal s, which was a 10-second logarithmic sine sweep (Eq. 1).

$$h_{RP}(n) = \mathcal{F}^{-1}\left[\frac{s_{RP}(f)}{s(f)}\right], \qquad (1)$$

Where s_{RP} is the output test signal of a particular reverb processor with one of the examined settings set to the desired value. Fig. 2 shows the workflow of the reverb processor's impulse response measurement. In order to calculate the room acoustic parameters, the Exponential Sine Sweep (ESS) method was used to capture the impulse response of every examined reverb processor and its different settings.

First of all, the initial values listed in Table 2 were set and measured (impulse response was captured). Only reverberation time was set to 1.8 seconds for all the reverbs as a common setting—the other initial settings were left as they were by default. After that, basic room acoustic estimators were calculated, such as Reverberation Time (RT), Early Decay Time (EDT), Bass Ratio (BR), Treble Ratio (TR), Clarity for Speech and Music (C50 and C80), and Central Time (CT). All the parameters were calculated according to the ISO 3382 standard [26] using Easera software [27]. Table 2 also shows the resultant room acoustic parameters for all the examined processors for the initial values of the processors. It is worth noting that the obtained values vary greatly, which is quite obvious because they sound completely different. It is interesting that the classically defined reverberation time calculated from the measured impulse responses in most cases differs a lot from the configured one.

Most of the modern reverbs are nonlinear [28] and the measured, static impulse responses are loaded with error. Analysis of a nonlinear system's impulse response measurement with ESS method [29] and [30] shows that the error obtained in that way is small enough to be omitted for current work purposes. The other issue that should be taken into account is a Signal to Noise Ratio (SNR). In room acoustics, the background noise is always present during the measurements; therefore noise reduction algorithms are being used. In current research only the reverberation algorithms are being tested and it is impossible to find out the signal or noise output of them. Theoretically, only the dynamic range of this plug-in could be objectively measured but even if there is internal 16-bit processing (the worst-case scenario), the SNR is higher than 80 dB, which is above the room acoustics realities.

Subsequently, each parameter from Table 1 was subject to change and the same measurement procedure was conducted. In total, over 60 combinations have been measured (5 reverbs x 5 parameters x 3 values + initial settings). From each combination (single impulse response), 8 room acoustic parameters were calculated and analyzed (over 500 results). The captured impulse responses are available for future research or analysis by permalink [31].

Additionally, simple basic listening assessment of the compared reverbs was performed. The settings that obtained the largest relative differences according to the default presets were evaluated by 4 expert listeners (sound engineers with a minimum of 10 years of active work experience). The reverb was applied to 3 different sound samples: drums, saxophone, and male vocal. Reverberation was mixed with dry sounds with a -14 dB ratio (the optimal mixing value according to [2]). Listeners verbally described their sensations and the most interesting and relevant comments are mentioned in the Results section. These listening conditions were set to be as close as possible to the typical reverb processor application.

5. RESULTS

The conducted experiments allowed selected reverb processor settings and their impact on produced reverb to be examined. The obtained results are presented in Tables 3

Table 1. Examined reverb processors along with selected settings and evaluated values.

Processor	IR	HR	TV	VH	LX
Parameter (**abbreviation**) range	Size (**SIZ**) 0.25 - 2 – 4	Size (**SIZ**) 0.5 - 1 – 2	Size (**SIZ**) 50 - 10,000 – 20,000	Late size (**SIZ**) 0 – 1	Color (**COL**) Dark 1 – Neutral 2 – Light 3
	Density (**DEN**) 0.25 - 2 - 4	Buildup (**BUP**) 0 - 0.24 - 2	Density (**DEN**) 0 - 0.5 - 1	Mode (**MOD**) Large chamber – Nostromo - sulfaco	Diffusion (**DIF**) 50 - 100%
	Decorrelation (**DCR**) 0 - 50 - 100	Er/Tail (**TIL**) 0 - 50 - 100	Dimension (**DIM**) 1 - 2 -3.99	Bass Mult (**BAM**) 0.5 – 1.25 - 2	Reflection (**REF**) 0 – 8 – 15
	Predelay (**PRE**) 0 - 10 - 30	Predelay (**PRE**) 0 - 10 - 30	Predelay (**PRE**) 10.5 - 30	Predelay (**PRE**) 0 - 30	Predelay (**PRE**) 0 - 10 - 30
	Resonance - (**RES**) 0.25 - 2 - 4	Er Type- (**ERT**) No: 1 – 3 - 10	Balance (**BAL**) -48 - 0 - 12	Diffusion -Early (**DIF**) 0 – 0,5	Early level (**REL**) off - -47 - 0

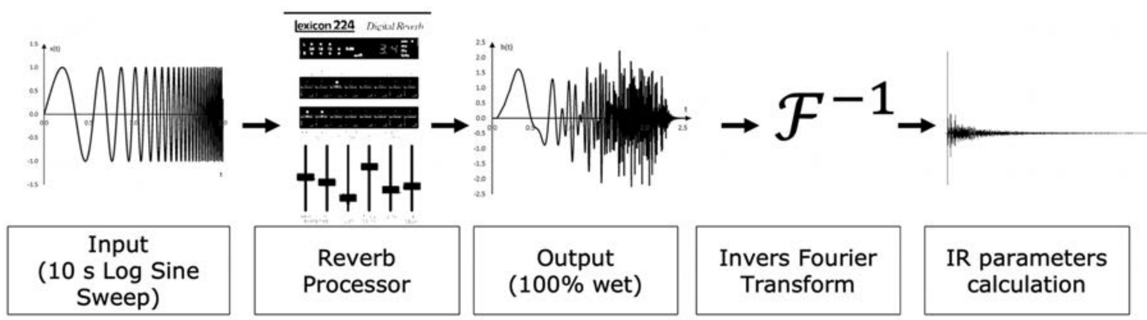

Fig. 2. Scheme of the reverb processors' impulse response measurement.

Table 2. Initial values of examined reverbs and their settings.

Reverb Processor	IR1	HR	TV	VH	LX
PARAMETER (Default value)	**SIZ** (1)	**SIZ** (1)	**SIZ** (5,516)	**SIZ** (0.5)	**Color** (Dark 2)
	DEN (1)	**BUP** (0.025)	**DEN** (0.85)	**MOD** (LR)	**DIF** (0%)
	DCR (0)	**TIL** (73.2)	**DIM** (3)	**BAM** (1)	**REF** (6)
	PRE (0)	**PRE** (24)	**PRE** (88.9)	**PRE** (10)	**PRE** (16)
	RES (1)	**ERT** (no. 5)	**BAL** (3)	**DIF** (1)	**REL** (−3.5 dB)
Reverb length Default/initial value [s]	**Convolution** (1.4)	**Reverb Time** (4.21/1.8)	**Decay Time** (1.2/1.4)	**Decay** (4/1.4)	**Reverb Time** (1.8/1.2)
Calculated room acoustic parameters for default or initial settings					
RT [s]	1.4	1.4	1.5	1.4	1.4
EDT [s]	1.9	1.6	1.0	1.4	1.4
BR	0.95	0.4	0.01	1.0	1.1
TR	0.86	0.04	0.8	0.9	0.9
C50 [dB]	0.5	−3.5	5.3	1.3	−3.0
C80 [dB]	3.1	0.1	9.6	3.3	−0.3
CT [ms]	78	127	32.6	47	95

Table 3. Relation of HR reverb processor parameters to room acoustic estimators where: • indicates default values and its results; **bold** indicates that JND threshold were exceeded; ↑ ↓ shows increasing and decreasing trends respectively and negative indicates that there is both a clear trend and JND exceeded; underline shows the settings that were evaluated subjectively.

HR		SIZ (•1)		BUP (•0.025)			TIL (•73.2)			PRE (•24 ms)			ERT (•no.5)		
		0.5	2	0	0.24	2	0	50	100	0	10	30	1	3	10
RT	•1.4	**51%↓**	**-47%↓**	**20%↓**	**-42%↓**	**-47%↓**	**58%**	**51%**	**50%**	**55%**	**-49%**	**-14%**	**50%**	**51%**	**-32%**
EDT	•1.6	**8%↓**	**-32%↓**	-1%	**38%**	**-32%**	**-46%↑**	1%↑	**11%↑**	**7%**	**-52%**	**-100%**	**10%**	**10%**	**-30%**
BR	•0.4	**30%**	**196%**	**61%**	**304%**	**196%**	**171%**	**138%**	**133%**	**125%**	**268%**	**241%**	**146%**	**145%**	**362%**
TR	•0.04	**2k%**	**1k%**	**1k%**	**23k%**	**1k%**	**2k%**	**1.7%**	**2k%**	**1k%↓**	**276%↓**	**-100%↓**	**2k%**	**2k%**	**2k%**
C50 [dB]	•-3.5	**-1.8**	**-1.8**	-0.8	**-12.0**	**-1.8**	0.7	**-1.9**	**-2.5**	**-1.3**	**-2.0**	**-4.9**	**-8.2**	**-7.6**	**-2.3**
C80 [dB]	•0.1	**4.6**	**5.0**	**5.2**	**-8.2**	**5.0**	**13.0**	**6.5**	**4.2**	**5.0**	**4.7**	**3.0**	**1.0**	**1.5**	**4.5**
CT [ms]	•127	**-12**	**-15**	**-19**	**152**	**-15**	**-56**	**-23**	**-6**	**-12↓**	**-10↓**	6↓	**19**	**17**	**-12**

to 7. These extended tables present the relative changes in room acoustic estimators with different reverb processor settings in relation to the values in Table 2 and are indicated by the • symbol. All the values in **bold** show that the Just-Noticeable Difference (JND) of the examined room acoustic parameters according to [26] was exceeded. Arrows (↑ or ↓) mark an increasing or decreasing trend of the obtained changes (information if changes of particular reverb processor settings cause regular/monotonical changes in a particular room acoustic estimator). When the trend is clear and the obtained relative changes are higher than JND, the values are highlighted as negative. The chosen settings that were evaluated aurally are underline.

Based on Table 3, most of the obtained differences (over 85%) exceed the JND levels (except STI), so changes in the HR reverb configuration should be immediately perceived. The configuration settings of the HR reverb strongly influence reverberation time and other room acoustic parameters. Most of the settings influence RT by almost 50%. In particular, the Er/Tail ratio (TIL) changes all the room acoustic parameters significantly but without any tendency. For the Size (SIZ) and Buildup (BUP) settings there are quite regular decreasing trends (the higher the values, the lower the reverberation time). On the other hand, changes in the EDT values are not proportional in the same way. Exceptionally high relative changes can be observed for timbre parameters (BR and TR) so the HR processor settings strongly filter the frequency content of reverb for selected settings. Only the Pre-Delay (PRE) setting changes TR values with a noticeable trend, which is problematic to explain as the Pre-Delay setting operates in the time domain only in a known, predictable, and defined way. In the listening evaluation, the SIZ (2) setting was described as smaller and tighter than the default SIZ (1). This corresponds with the obtained room acoustic parameters. TIL (0) was described as much shorter than the default TIL (73.2) and PRE (30) as darker and smaller than PRE (24).

Changes in the configuration of the IR reverb cause moderate changes in the room acoustic estimators (Table 4). Almost 50% of the calculated results exceed the JND level. Reverberation time and EDT changes with the SIZ parameter but an increasing trend is clear only for RT. The SIZ setting also monotonically controls clarity, CT, STI, and TR. There are some clear trends for the Resonance (RES), Decorrelation (DCR), and Density (DEN) settings as well, but only 3 of 11 exceed the JND levels. Changes in the PRE delay cause subtle and constant changes for most room acoustic parameters. In the listening comparison, SIZ (0.25) was described as shorter than SIZ (1) as well as RES (4) in comparison to RES (1). In addition, differences in timbre have been noticed by listeners, which probably corresponds with TR changes.

Table 5 shows that the changes in configuration of the LX reverb processor cause high changes in reverberation time and other parameters when compared with the previous reverbs. More than 90% of the obtained results exceed the JND level and 8 trends can be noticed for the Color (COL), Diffusion (DIF), and Early Level (REL) settings. The COL setting works according to logic and intuition, so lower values of Color increase Bass Ratio and decrease Treble Ratio. But COL also changes reverberation time. For the DIF setting, there are clear trends for EDT, BR, and CT. There are no trends for the Reflection (REF) and PRE settings but relative changes in parameters for these settings are much higher than JND. In the subjective assessment, REF (8) was described as denser or wider than REF (6). REL (−47) was described as larger and longer than REL (−3.5).

Room acoustic reverberation parameters calculated for different settings of the TV reverb (Table 6) are higher than the JND level for only about 25%. For BR, some changes are extreme due to the very low value of the default settings. On the other hand, many trends have been noticed (17), and 8 of them are connected with exceeded JND. The TV reverb architecture means that the length of RT and EDT differ a lot from each other, but most of the settings change these values with some regularity, but the magnitude of the changes is rather small. The configuration of the Balance (BAL) setting modifies all of the room acoustic estimators with some regularity and higher than the JND levels. On the contrary, the Density (DEN) setting almost does not modify any parameters. In the listening comparison, BAL (−48) was described as smaller and shorter but more defined than BAL (3), probably due to the much shorter RT and EDT

Table 4. Relation of IR reverb processor parameters to room acoustic estimators (the same legend as in Table 3).

IR		SIZ (•1)			DEN (•1)			RES (•1)			DCR (•0)		PRE (• 0 ms)	
		0.25	2	4	0.25	2	4	0.25	2	4	50	100	10	30
RT	•1.4 s	-24%↑	6%↑	10%↑	-1%	-1%	1%	4%↓	-13%↓	-17%↓	7%	4%	4%	4%
EDT	•1.9 s	-69%	-10%	-15%	15%	5%	6%	-4%	1%	-30%	-12%	-11%	-4%	-4%
BR	•0.95	19%	1%	-1%	6%	8%	2%	-9%	9%	-7%	8%↓	15%↓	-4%	-4%
TR	•0.86	-2%↑	6%↑	10%↑	9%	1%	-1%	19%↓	-8%↓	-28%↓	-6%	-3%	-2%	-2%
C50 [dB]	•0.5	8.5↓	-1.8↓	-3.4↓	-3.1↑	0↑	0.2↑	0	-0.3↓	-1↓	0.2↓	0.5↓	9	9
C80 [dB]	•3.1	8.2↓	-1.8↓	-3.6↓	-4.1↑	0.1↑	0.4↑	0	-0.2↓	-1.2↓	0.1↓	0.2↓	8	8
CT [ms]	•78	-60.34↑	18.47↑	39.67↑	46.82	-0.52	-1.56	0.1	4.15	6.02	-0.96↑	-5.54↑	-60.5	-60.5

and higher clarity. SIZ (50) was described as larger and brighter than SIZ (5,516).

Configuration of the last examined algorithmic reverb processor, VH, does not modify room acoustic parameters much (Table 7). Only around 30% of the obtained results exceed the JND level. Additionally, only 1 clear trend is noticed for the Bas Mult (BAM) setting and TR parameter changes. The BAM setting also modifies RT, EDT, and BR significantly but without any regular trend. The DIF and SIZ settings cause a few trends in parameter changes but their magnitude is very low. In the listening evaluation, BAM (0.5) was described as smaller and tighter than BAM (1), which is clearly reflected in the calculated room acoustic parameters.

6. SUMMARY AND CONCLUSIONS

On the basis of the conducted experiment, several conclusions can be formulated. The relation of reverb processor configuration to room acoustic estimators of the produced reverb is very complex and not straightforward. The same or similar reverb settings can influence the room acoustic parameter in very different ways depending on the particular processor. For example, the very common SIZ setting changes reverberation time in different ways (with its rise monotonically increasing RT in one reverb but with the opposite trend in the other two). Table 8 summarizes all the obtained results and conclusions. Cells with arrows ↑↓ in negative show increasing/decreasing trends with results that exceeded the JND levels. Regular arrows ↑↓ show increasing/decreasing trends but with low value changes. The symbol > indicates no noticeable tendency but values exceeding the JND levels for at least 2 values. Cells with the · symbol indicate no trend and relative changes that are lower than the JND levels. Table 8 shows that most of the examined reverb processors modify RT or EDT when changing some other settings while the actual reverberation setting built in the processor is set to a constant value. In

Table 5. Relation of LX reverb processor parameters to room acoustic estimators (the same legend as in Table 3).

LX		COL (•DRK2)			DIF (•0%)		REF (•6)			PRE (•16 ms)			REL (•-3.5 dB)		
		DRK 1	NTL 2	LGT 3	50	100	0	8	15	0	10	30	-inf	-47	-0
RT	•1.4 s	11%↓	-9%↓	-32%↓	6%	2%	2%	-57%	3%	-38%	5%	3%	4%	-3%	4%
EDT	•1.9 s	29%	16%	4%	27%↑	31%↑	36%	-16%	16%	10%	23%	30%	24%	14%	27%
BR	•0.95	17%↓	-10%↓	-35%↓	9%↑	17%↑	15%	36%	-46%	-64%	13%	14%	-41%	-57%	13%
TR	•0.86	-27%↑	9%↑	42%↑	-17%	-13%	-14%	99%	-15%	41%	-16%	-15%	-15%	-9%	-15%
C50 [dB]	•0.5	-4.4	-4.6	-4.2	-6.1	-6.2	-1.8	-3.2	-3.7	-2.7	-2.9	-6.1	-9.5	-9.4	-1.8
C80 [dB]	•3.1	-3.6	-3.6	-3.6	-5.8	-5.9	-2.7	-4.0	-1.1	-3.7	-3.7	-5.5	-5.6	-5.9	-1.9
CT [ms]	•78	74	70	62	87↑	93↑	61	67	42	71	66	84	89	92	52

Table 6. Relation of TV reverb processor parameters to room acoustic estimators (the same legend as in Table 3).

TV		SIZ (•5516)			BAL (•3 dB)			DEN (•0.85)			DIM (•3)			PRE (•88.9 ms)	
		50	10000	20000	-48	0	12	0	0.5	1	1	2	3.99	10.5	30
RT	•1.5	3%↓	-2%↓	-6%↓	-94%↑	-3%↑	3%↑	2%	0%	0%	1%↓	0%↓	-1%↓	-3%↓	-1%↓
EDT	•1.0	47%	3%	6%	-69%↑	-3%↑	50%↑	2%↓	1%↓	-1%↓	3%↓	0%↓	5%↓	-1%	-1%
BR	•0.01	18k%	-100%	8k%	3k%	3%	1k%	-51%	0%	0%	347%↓	351%↓	-100%↓	18k%	1%
TR	•0.8	1%	0%	-1%	84%	-2%	0%	-1%	0%	-1%	0%	0%	0%	-3%↓	-2%↓
C50 [dB]	•5.3	-1.5	-0.5	-1.2	2.6↓	1.1↓	-6.0↓	0.1	0.1	0.0	1.5↓	0.6↓	-1.8↓	1.2↓	1.1↓
C80 [dB]	•9.6	-3.7	0.0	-2.5	8.1↓	2.3↓	-8.0↓	0.0	-0.1	0.0	0.0	0.1	-1.2	2.6	3.0
CT [ms]	•32.6	17	0	2	-13↓	-6↓	47↓	0	0	0	-1	-1	4	-8↑	-6↑

Table 7. Relation of VH reverb processor parameters to room acoustic estimators (the same legend as in Table 3).

VH		BAM (•1)			DIF (•1)		MOD (•LR)			PRE (•10)		SIZ (•0.5)	
		0.5	2	125	0	0.5	LC	NOSTR	SUL	0	30	0	1
RT	•1.4	**-34%**	**53%**	**15%**	0%	0%	-2%	**-9%**	-2%	1%	2%	**6%↓**	-1%↓
EDT	•1.4	**-32%**	**59%**	**16%**	**-5%↑**	-4%↑	**-13%**	-2%	4%	-3%	**-5%**	-2%↑	2%↑
BR	•1.0	**-18%**	**25%**	**8%**	2%↓	1%↓	**5%**	3%	2%	2%	4%	3%	**5%**
TR	•0.9	**34%↓**	**-27%↓**	**-11%↓**	-3%↑	-1%↑	1%	**5%**	2%	-2%	-3%	**-8%↑**	2%↑
C50 [dB]	•1.3	0.3	-0.4	-0.3	-0.1	-0.3	**3.9**	**1.7**	**-1.4**	-0.1	-0.2	0.6↓	**-1.7↓**
C80 [dB]	•3.3	0.7	-0.7	-0.2	-0.6↑	-0.2↑	**3.4**	**1.6**	-0.7	0.0	0.0	**1.0↓**	-0.4↓
CT [ms]	•47	-8	**14**	4	-2	0	**-28**	**-23**	6	-1	0	-9↑	6↑

a few cases, the obtained results consist of expectations, simple intuition, and a textbook definition of a particular setting, like Color or Diffusion in the LX processor or Size in the IR processor. There are some settings in the selected processors that allow the controlled shaping of room acoustic parameters but in a very limited and undefined range, such as the DIM setting in the TV processor or the BAM setting in the VH processor.

Generally, most of the examined settings change many room acoustic parameters in a significant way but without any noticeable relation between them: ERT in the HR processor, REF and REL in the LX processor, and SIZ in the TV and VH processors. In addition, the SIZ parameter that is present in all processors, except LX, changes reverberation time in different ways for each processor. For IR and TV changes are almost proportional and for HR they are inversely proportional. There is no trend only for the VH processor. The listening evaluation of reverberation with SIZ changes also confirms that there are different and contrary interpretations of the SIZ setting according to particular processors. It was also reported by all the listeners that the audible differences between the samples were small and subtle while only the reverbs with the largest changes in settings and room parameters were examined.

The obtained results show that the perceptive magnitude of the reverb setting varies very much from one processor to another. In some cases, extreme settings of a particular reverb parameter caused changes lower than the JND threshold and, on the other hand, in some cases the obtained differences were enormous. Despite there being a few observations for particular processor settings and room acoustic parameters, it is impossible to formulate general tendencies or relations. It could be stated that reverb processors in the audio engineering domain and room acoustic approach do not overlap. In the authors' opinion, due to the massive research dedicated to room acoustics and exceptional auditory capabilities of sound engineers, these two worlds should definitely collaborate more. Certainly, there is a lack of a tool that would allow users to freely and under control produce a reverb effect with room acoustic parameters as settings and its design would be a technological challenge. This kind of software would be helpful for both sound engineers and acoustic consultants. A potentially similar solution was proposed in [32] and [33] for the automatic and perceptually based reverberation applications. In these works, the basic reverberation algorithm can be found, which was originally proposed in [34] and then developed by [32]. The proposed solution allows generation of reverberation and control of selected room acoustic parameters through its configuration. Results of [33] show that characteristics derived by that algorithm "does not necessarily lead to intended perceptual effects." Moreover, the

Table 8. Summary of relation of examined reverb processor settings to room acoustic estimators.

	HR					IR					LX					TV					VH				
	SIZ	BUP	TIL	PRE	ERT	SIZ	DEN	RES	DCR	PRE	COL	DIF	REF	PRE	REL	SIZ	BAL	DEN	DIM	PRE	BAM	DIF	MOD	PRE	SIZ
RT	**↓**	**↓**	>	>	>	**↑**	·	**↓**	·	·	**↓**	·	·	>	·	↓	↑	·	↓	↓	>	·	·	·	↓
EDT	**↓**	>	**↑**	>	>	>	>	·	>	·	>	**↑**	>	>	>	·	**↑**	↓	↓	·	>	↑	·	·	↑
BR	>	>	>	>	>	·	>	>	**↓**	·	**↓**	**↑**	>	>	>	>	>	·	**↓**	↓	>	↓	·	·	·
TR	>	>	>	**↓**	>	↑	·	**↓**	·	·	**↑**	>	>	>	>	·	·	·	·	·	**↓**	↑	·	·	↑
C50	>	>	>	>	>	**↑**	↑	↓	↓	>	>	>	>	>	>	>	**↓**	·	**↓**	**↓**	·	·	>	·	↓
C80	>	>	>	>	>	**↓**	↑	↓	↓	>	>	>	>	>	>	>	**↓**	·	·		·	↑	>	·	↓
CT	>	>	>	↓	>	**↓**	·	·	↑	·	>	**↑**	>	>	>	·	**↓**	·	·	↑	·	·	>	·	↑

	↑↓	increasing/ decreasing trend and JND exceeded	↑↓	increasing/ decreasing trend but low changes
	>	no trend but JND exceeded for at least 2 values	·	no trend and relative changes lower than JND

authors of [33] suggest the future work to focus on implementation of a "model agnostic architecture" that could be used with commercially available reverberation effects. Current results show that this is probably not the best solution.

The selection of reverb processors and their settings in the current examination was arbitrary and very limited, so all the general conclusions are strongly biased. On the other hand, based on the authors' knowledge and experience, other popular reverb tools have very similar features to the examined ones. Additionally, the limitation to only 3 or 4 values per setting examination makes it hard to formulate unambiguous presumptions. However, in many examined cases, it was shown that the relation between reverb settings and room acoustic parameters is definitely irregular and sometimes even stochastic.

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