Equal reverberance matching of music

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ABSTRACT

This study explores the reverberance of music in simulated auditoria. It investigates the effects of gain on the reverberance of an anechoic music recording convolved with auditorium impulse responses. Based on objective loudness modelling, our hypothesis is that gain has a positive effect on reverberance (even though it has no effect on reverberation time). In a subjective experiment, participants adjusted decay rate of auditorium impulse responses convolved with an anechoic music sample in order to match the reverberance of each music stimulus to that of a reference music sample. Results support the hypothesis, and are similar to those of a previous study in which auditorium impulse responses (without convolution) were matched similarly for reverberance.

INTRODUCTION

Since Sabine (1923) proposed reverberation time as a perceptually-derived measure of physical reverberation, there have been many efforts in developing or refining reverberation-related parameters because it has been generally agreed that reverberation time is an imperfect measure of reverberance (which can be broadly defined as the perceived amount of reverberation). After Haas (1951, translated 1972) showed that early reflections are important in the human perception of sound, a number of parameters assessing the human perception of sound clarity were proposed based on ratios of early sound energy to reverberant sound energy, or to the overall sound energy of a room impulse response: such as *Deutlichkeit* by Thiele (1953), *R* by Schultz (1965) and the clarity index by Reichardts and Lehman (1981). With respect to parameters indicating the reverberance, Atal *et al.* (1965) proposed the initial reverberation time considering on importance of early sound energy. For that parameter, the early sound energy was defined as sound energy arriving in the first 160 ms or -15 dB of the peak of the sound decay. Later, Jordan (1969) refined the evaluation range of the initial reverberation time as from the peak of sound decay to -10 dB of the peak and named the refined parameter ‘early decay time’ (EDT). One of the concepts behind early decay time is that, in listening to ‘running’ signals such as music or speech, there is little opportunity to hear the full reverberation decay, so it is logical that people would make judgments of reverberance from the initial part of the decay.

Support for EDT also comes from perceptual studies. For example, Soulodre and Bradley (1995) conducted subjective experiments in order to investigate the degree of agreement between such parameters and the human perception of reverberation (using an orchestral music stimulus). The results show that reverberation time has a correlation coefficient of $r = 0.740$ with reverberance when it is averaged over all octave bands from 125 Hz to 4 kHz. When averaged over mid-frequencies, 500 Hz to 1 kHz in octave bands, the correlation coefficient slightly increases to $r = 0.799$. Compared to reverberation time, EDT yields much better correlation with human perception ($r = 0.971$) when all octave bands are averaged. The use of EDT to assess reverberance is now reinforced by the key standard for this field, ISO 3382-1 (2009).

Although EDT is highly correlated with the reverberance of the particular stimulus set used by Soulodre and Bradley (1995), there is some reason to suspect that EDT would not correlate so well with the reverberance of a wider range of stimuli. One reason is that the parameter does not take into account non-temporal factors that are likely to influence the reverberance, such as level of the source signal and backgroud noise. In ISO 3382-1 (2009), reverberation time (T20) is defined as the time taken by sound decaying from -5 dB to the -25 dB of the peak of the best-fit regression line over this range of the reverse integration curve, multiplied by three. EDT is defined as time taken for sound decaying from the peak to -10 dB of the best-fit regression line over this range of the reverse integartion curve, multiplied by six. Clearly, the slope of the regression line is unaffected by signal gain, so the gain has no effect on early decay time and reverberation time.

According to Hase *et al.* (2000), reverberance is strongly affected by listening level of a stimulus. That study also shows that the signal level has a stronger effect on reverberance than reverberation time for the tested music and speech samples. This is indicated by an *F*-ratio of the sound level being 312.03 compared to that of reverberation time being 32.94 in assessments of the reverberance of music. Lee and Cabrera (2009b) examined effects of the signal level and of background noise on reverberance. Although that study used an obviously different type of stimulus (room impulse responses listened to directly), findings are consistent with the study by Hase *et al.* in terms of the effect of gain. Moreover, Lee and Cabrera found that background noise has a strong negative effect on reverberance. The effects of these two factors could be related by the masked hearing threshold. When the gain increases, a part of the stimulus previously below the masked hearing threshold becomes audible. Conversely, introducing background noise obscures some parts of a stimulus below the masked hearing threshold, so that the audibility of the reverberation is reduced.

Another reason for the effect of gain on reverberance is that the loudness of the decay over time is likely to have a gain-dependent slope (Lee and Cabrera, 2009a). Sound pressure level in decibels does not account for the many complexities of the human perception of loudness, such as auditory filter banks, spectral masking and temporal integration and so forth (Zwicker and Fastl, 1999).

In this paper, we examine the reverberance of an anechoic music recording convolved with room impulse responses (RIRs). This study emulates aspects of Bradley and Soulodre's study (1995) by using similar stimuli and the same definition of reverberance. Unlike their study, this one is a magnitude production experiment (rather than magnitude estimation). The experiment is similar in method to the experiment described by Lee and Cabrera (2009b) – except that the stimuli in the former experiment were RIRs that had not been convolved with a signal. The experiment mainly aims to examine the relationship between listening level and reverberance, and analyses this relationship using an objective dynamic loudness model (Chalupper and Fastl 2002). As proposed previously by the authors (Lee and Cabrera, 2009a, 2009b) this approach has some potential to provide an alternative way of modelling reverberance.

METHOD

The experiment of this paper is a subjective magnitude matching task: subjects were asked adjust one stimulus so that it had the same reverberance as a reference stimulus. This adjustment changed the decay rate of an impulse response that was then convolved with a fixed music signal.

The experiment was conducted in an anechoic chamber with circumaural headphones (Sennheiser HD600). The background noise level of the anechoic chamber was below the threshold of hearing specified in ANSI S12.2 (1995). In the experiment, eight measured room impulse responses (RIRs) were convolved with a music passage after scaling them by +5 dB, 0 dB and -5 dB. Hence twenty-four RIRs were tested in the experiment. Since all the RIRs were monaural, the variation in binaural spatial chracteristics of the auditoria (e.g., the different interaural cross correlation functions) was excluded from the test, so that we could focus on the effect of gain. All RIRs were measured as described by Farina and Ayalon (2003) in a complex of three auditoria, *Parco della Musica*, located in Rome. The small auditorium has 700 audience seats, the medium auditorium has 1200, and the large auditorium has 2800. The relative levels of the RIRs were retained by measuring with identical equipment and gain, and the RIRs were not normalized. The music passage used in the experiment was the *Overture* to *Le Nozze di Figaro* by Mozart from *Ancehoic Orchestral Music Recordings* (1995), which is the same stimulus as that used by Soulodre and Bradley (1995). The passage is 16 seconds long (bars 1-18) and sampled in the anechoic conditions specified in ISO 3745 (Denon Professional Test CDs).

Table 1. Source-receiver distance, L_{Aeq} of the music passage convolved with receiver positions, $\mathrm{EDT}_{\mathrm{mid}}$ and $\mathrm{RT}_{\mathrm{mid}}$ for the eight RIRs

	S1	S2	M1	M2	M3
Distance (m)	12	24	10	19	31
L_{Aeq} (dB)	76.0	75.6	75.5	73.7	72.4
$\mathrm{EDT}_{\mathrm{mid}}$ (s)	1.89	1.98	1.83	1.77	2.0
$\mathrm{RT}_{\mathrm{mid}}$ (s)	2.06	2.07	2.01	2.03	2.17
	L1	L2	L3		
Distance (m)	20.5	30	48		
L_{Aeq} (dB)	71.3	71.2	65.1		
$\mathrm{EDT}_{\mathrm{mid}}$ (s)	2.44	2.25	2.38		
$\mathrm{RT}_{\mathrm{mid}}$ (s)	2.66	2.60	2.53		

For each of the RIRs, Table 1 provides the L_{Aeq} of the convolved music, $\mathrm{EDT}_{\mathrm{mid}}$, $\mathrm{T20}_{\mathrm{mid}}$ and the source-receiver distance used in the measurement. The L_{Aeq} values were measured from the experiment headphones using a Brüel & Kjær type 4100 head and torso simulator. Because the music passage has a left and right channel, the L_{Aeq} values shown in the table are the power average values of two channels. In labelling the RIRs, S refers to the RIRs measured in the small auditorium and M to the medium auditorium and L to the large auditorium. As seen in the table, the three RIRs measured in the large auditorium have significantly longer $\mathrm{RT}_{\mathrm{mid}}$ and $\mathrm{EDT}_{\mathrm{mid}}$ values than those measured in the small and medium auditoria.

In this experiment, the subject adjusted the reverberation time of the stimuli by multiplying the RIR by an exponential function. However, in doing this, the noise floor positioned at the tail of the RIRs has the potential to create artificial echoes at the end of the convolved sounds, because the originally steady noise floor will grow if reverberation time is lengthened in this way. This noise floor cannot be simply deleted or faded because the time that it is reached depends on frequency. For example, 1 kHz octave band of M1 has the noise floor staring at approximately 2 s as seen in Figure 1 (upper figure), while 4 kHz octave band of the same RIR has the noise floor starting at approximately 1.5 s. To avoid this, each RIR was filtered into octave bands centred on 31.5 Hz to 16 kHz, and the octave band noise floors were treated as shown in Figure 1, before being recombined into a single RIR. The noise floor starting point was determined visually for each octave band and the noise floor was multiplied by an exponential function with an appropriate coefficient so as to yield a slope that is a continuation of RIR's decay. In the filtering process, the RIRs were filtered twice, in foward and reverse directions, so as to avoid phase distortion (using a fourth-order Butterworth filter in each direction). Discrepancies in EDT and T20 between the treated RIRs and the original RIRs were less than 0.03 s from 63 Hz to 16 kHz for the eight RIRs, except L1 at 63 Hz (0.06 s), L2 at 250 Hz (0.04 s) and L3 at 63 Hz (0.04 s). Considering the fact that the just noticeable difference (JND) of reverberation time is of order of 5% (ISO 3382-1), the discrepancies are far below one unit of the JND. Hence, the discrepancies should be acceptable for this type of listening test.

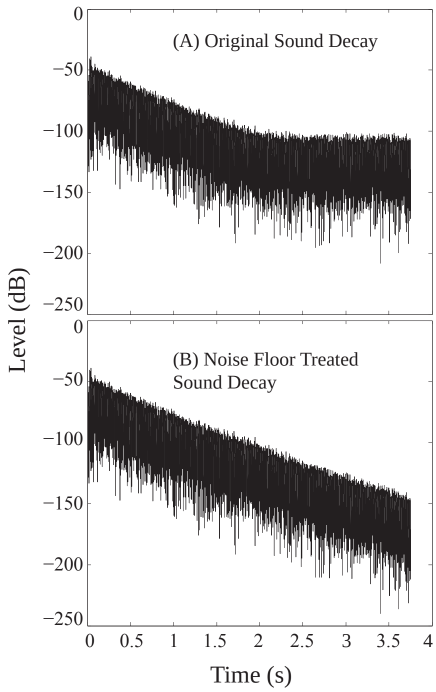

Figure 1. Sound decay of M1 in the 1 kHz octave band: (A) without the noise floor treatment; and (B) with the noise floor treatment

The music passage convolved with M1 was used as the reference stimulus. The reverberance of the comparison stimulus was adjusted by changing decay rate of the corresponding RIR, which is done by applying the equation 1.

$$p'(t) = p(t) \times \exp\left\{ \frac{\left(-3 + \left(3 \times 1.04^{d}\right)\right) \times t}{1.04^{d}} \right\} \qquad (1)$$

Here $p(t)$ is sound pressure of the RIR as a function of time, t is time in seconds, d is a decay adjustment value and $p'(t)$ is sound pressure of the RIR after decay adjustment. The equation is designed to change the average of octave band values of EDT_{mid} and RT_{mid} by approximately 4%, which is just under the one unit of the JND. Incrementing d prolongs the reverberation time. Initial d values for the comparison stimuli were randomly set in the range between -4 and +4, which corresponds to a modified reverberation time between 1.74 s and 2.29 s for an initial reverberation time of 2.0 s or to a modified reverberation time between 2.94 s and 2.11 s for an initial reverberation time of 2.5 s.

The energy of a RIR changes if its decay rate is changed (a shorter reverberation time reduces the energy). However, Lee and Cabrera (2009b) found that the reverberance of RIRs listened to directly (in an otherwise similar experiment) is not significantly affected by this change. Nevertheless, in the present experiment, we compensated for the energy change by adjusting the RIR gain. Prior to this compensation, the change in energy, expressed in decibels, is given by equation 2,

$$\Delta L_E = 10 \times \log_{10}\left(\frac{\int p'^2(t)}{\int p^2(t)} \right) \qquad (2)$$

In equation 2, ΔL_E is the amount of energy lost or gained from applying equation 1, and this was subtracted from decay rate modified RIRs before convolving with the music passage.

Fifteen subjects participated in the experiment, eleven of them with an educational background in acoustics including room acoustics. The task of the experiment was to match the reverberance of comparison stimuli to that of a reference stimulus. Since the term *reverberance* was vague to some of the subjects, its definition was explained to all the subjects as "the degree of perceived reverberation in a temporal sense. The blending of one sound into subsequent following sounds." This is the same definition as that used by Soulodre and Bradley (1995) for describing the term *reverberance* to their subjects.

RESULTS

An analysis of variance (ANOVA) for the experiment responses is shown in Table 2. As seen in the table, the gain adjustment and the RIR both significantly affect reverberance (Prob>F is 0) while there was no significant interaction effect between the two variables.

Table 2. ANOVA for the subjective experiment restuls

Variable	Sum Sq.	D.F	Mean Sq.	F	Prob>F
Gain	473.52	2	236.758	17.3	0
RIR	830.03	7	118.576	8.68	0
Gain*RIR	83.28	14	5.949	0.44	0.965
Error	4588.67	336	13.657		
Total	5975.5	359			

In order to evaluate the differences between the categories of each variable, Tukey/Kramer's *post hoc* tests (often referred to as Tukey's HSD) were performed. Table 3 shows the multi-comparison test for effect of gain variation. CI Low refers to the low end of the confidence interval and CI High refers to the high end of the confidence interval at a confidence level of 95%. As seen in the table, none of the pairs include zero within the range between the CI Low and CI High. Hence the null hypothesis (that the true difference between caterigores is zero) can be rejected, meaning that there are significant differences in d between each category.

Table 3. Multi-comparison test for effect of gain variation

Gain	CI Low	Mean Diff.	CI High
0 dB, -5 dB	-2.5295	-1.3417	-0.5138
0 dB, +5 dB	0.2788	1.4667	2.6545
-5 dB, +5 dB	1.6205	2.8033	3.9665

Figure 2 shows the average values of d for the three gain settings. As seen in the figure, the value of d decreases with gain. In other words, the subjects reduced reverberation time more for the stimuli having higher gain offset than those having lower gain offset.

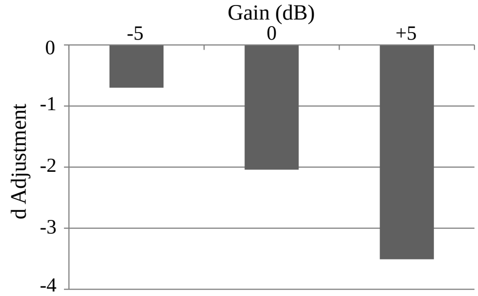

Figure 2. Mean values of d for the three gain settings.

Table 4 shows the multiple-comparison test for effect of different RIR. The table shows only pairs having significant differences in d between the categories. The results show that the siginificant differences are found mostly for pairs with a RIR from the large auditorium.

Table 4. Significant differences (only) from the multi-comparison test for the effect of RIRs

RIRs	CI Low	Mean Diff.	CI High
L1, M1	-6.4430	-4.0000	-1.5570
L1, M2	-6.1319	-3.6889	-1.2459
L1, M3	-5.0874	-2.6444	-0.2015
L2, M1	-6.1985	-3.7556	-1.3126
L2, M2	-5.8874	-3.4444	-1.0015
L3, M1	-6.1985	-3.7556	-1.3126
L3, M2	-5.8874	-3.4444	-1.0015
M1, S1	0.9348	3.3778	5.8207
M1, S2	0.1570	2.6000	5.0430
M2, S1	0.6237	3.0667	5.5096

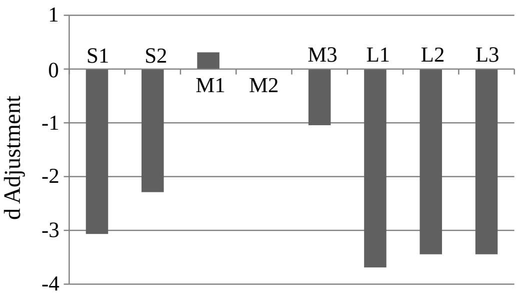

Figure 3. Mean values of d for each RIR

Figure 3 shows the average values of d for different RIRs. As the music passage convoved with M1 is the reference stimulus, the average value of d for M1 is close to zero. Considering the JND of reverberation time is slightly higher than one unit of d, the subjects almost perfectly matched the reverberance of the stimuli convolved with M1 to that of reference.

The slight discrepancy might be partly due to responses to the gain-adjusted M1.

ANALYSIS

The results clearly show that gain affects reverberance, but this concept is not reflected in the conventional reverberation parameters of ISO3382-1. Compared to the sound pressure decay, the loudness decay curve obtained from a loudness model should provide a closer match to the human perception of sound decay, as the model accounts for many complexities of perceived loudness. Therefore we have adapted the concept of reverberation time to the loudness decay curve, and defined a parameter which could be called 'loudness reverberation time', or T_N. This is helped by the fact that the loudness decay function of a typical RIR is approximately exponential (at least, at first – see Figure 4), which could be expected from steady state loudness theory (Lee and Cabrera 2009a). Hence, a linear regression of the logarithm of a selected part of the loudness decay function can be used to quantify the slope of the decay in a way analogous to EDT, T20, T30 etc (ISO3382-1). According to Stevens (1955), loudness is proportional to sound pressure raised to a power of 0.6 for tones of moderate frequency and sound pressure level (this agrees with the rule-of-thumb that halving the loudness is achieved by a gain of -10 dB). Based on this, if we were to choose a T_N evaluation range analogous to T20, we would use the part of the decay between 0.708 and 0.178 of the peak loudness. However, a larger evaluation range was needed to best model the results of Lee and Cabrera (2009b), and indeed is also needed for the results of the present experiment. The evaluation range used for T_N in the present analysis is between 0.708 and 0.022 of the peak loudness, which would correspond to the interval from -5 dB to -55 dB if Stevens' power law was accurate over such a large range (but in fact corresponds to a smaller range because of the increased steepness of the loudness growth function at low sound pressure levels, relative to moderate sound pressure levels).

Figure 4 illustrates the loudness decay curves of M1 with various gain offsets ($L_{AF,max}$ levels from 35 dB to 75 dB) and the T_N evaluation range. It can be seen that the evaluation range is unaffected by the noise floor. It can also be observed that the time interval of the evaluation range is substantially longer in the case of the 75 dB decay, compared to that of the 35 dB decay. Note that this tendency continues at higher sound pressure levels than those illustrated (Lee and Cabrera 2009a). As seen in the figure, T_N shows some promise to account for the effect of gain on reverberance, as T_N becomes longer for higher gain.

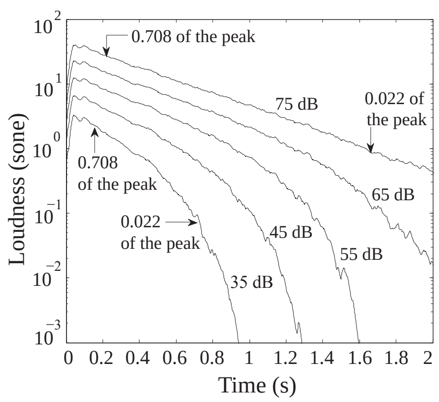

Figure 4. Loudness decay functions (derived using Chalupper and Fastl's dynamic loudness model (2002)) of the M1 RIR, showing the T_N evaluation range

Since the loudness model is sensitive to level of a signal, it is important to define listening level of the signal in calculating T_N. However, the stimuli tested in the subjective experiment were not RIRs, and there is not a straightforward relation between a RIR and a sound convolved with that RIR. This is partly because the gain of the RIR and the dry signal are confounded in the convolution. Another issue is that while L_{eq} is usually a reasonably good representation of the strength of a convolved signal, a RIR is not well suited to being quantified in that way because of its 'percussive' quality and because its duration is poorly defined (hence the use of $L_{F,max}$ above). A further difficulty is that the spectral distribution of the RIR is different to that of the convolved signal. In our loudness analysis of the RIRs, we calibrated M1 to 75.5 dB L_{AFmax} and those of the rest accordingly (i.e., retaining the level relationship between the RIRs, even though the relationship between the RIR L_{AFmax} and the convolved L_{Aeq} is not exactly consistent). As seen in Table 1, 75.5 dB is L_{Aeq} of the reference stimulus (the music passage convolved with M1) averaged over the stereo channels.

After calibrating the levels of the RIRs, $T20_{mid}$, EDT_{octave}, and T_N were calculated from the RIRs having the mean decay rate adjustment from the experiment results, and from those without the decay rate adjustment. To obtain the loudness decay curves for the RIRs, we used Chalupper and Fastl's dynamic loudness model (2002) implemented in Psysound 3 (Cabrera *et al.* 2008). EDT_{octave} referes to early decay time averaged over octave band values from 125 Hz to 4 kHz, which was the best predictor of reverberance in Soulodre and Bradley's (1995) study. Figure 5 shows the results. The vertical axis of the figure represents the parameter slope (s/dB), which is derived from a linear regression line of the corresponding parameter over the three gain offsets. Hence, zero means that there is no change over the three gain-offsets in the corresponding parameter. Since the conventional parameters are unaffected by the gain changes, the parameter slopes for $T20_{mid}$ and EDT_{octave} are always zero for the decay rate unadjusted RIRs. Therefore, *Conventional* $T20_{mid}$ and - *Conventional* EDT_{octave} shown in the figure represent the parameters obatained from the RIRs having the decay rate adjustment, and those from the RIRs without the decay rate adjustment are excluded in the figure.

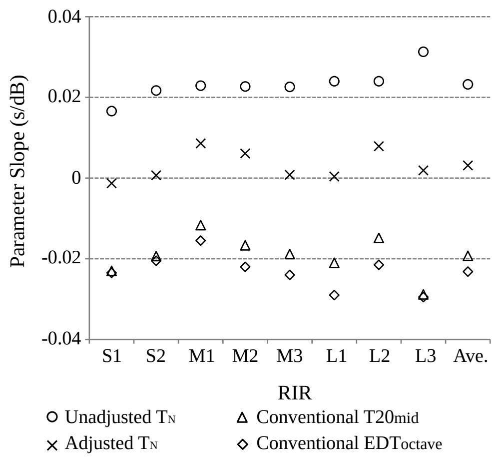

Figure 5. Slope of the linear regression of T_N, $T20_{mid}$ and EDT_{octave} for RIRs having different gain offsets and for two decay adjustement situations.

As seen in the figure, the parameter slopes of the *conventional* $T20_{mid}$ and *Conventional* EDT_{octave} are clearly displaced from the horizontal zero line. Compared to these conventional parameters, the *Adjusted* T_N yields the parameter slopes closest to zero (the average slope is 0.0031) than those of the *Unadjusted* T_N (the average slope is 0.0232). Hence, in matching the reverberance of the stimuli at the three gains, the subjects appear to be matching the loudness reverberation time, T_N, rather than the conventional paratemers.

DISCUSSION

An issue that we initially considered was how to define reverberance to the subjects. According to Barron (2001), there are two aspects to reverberance: temporal and spatial aspects. Morimoto and Asaoka (2004) conducted subjective experiments to evaluate relations between those aspects for both terminal (or stopped) reverberance and running reverberance. Their study shows that the temporal aspect has the stronger effect on the reverberance, especially for terminal reverberance. Since the present study aims to compare the results with Soulodre and Bradley's study (1995) and to extend the study by the authors (2009b), we used Soulodre and Bradley's definition, which emphasises temporal perception. In generating the stimuli, we minimised variation in the spatial aspect of reverberance by using single channel RIRs (which were convolved with the stereophonic anechoic recording).

The subjective experiment conducted in this study supports the findings of Lee and Cabrera (2009b) and of Hase *et al*. (2000) that the gain has a strong positive effect on reverberance. As seen in Figure 3, the discrepancy in *d* between the RIRs having -5 dB gain offset and those having +5 dB gain offset is about 3, which corresponds to a modified reverberation time of 2.2 s for a reverberation time of 2.0 s. Considering the JND of reverberation time is of order of 5%, the 0.2 seconds deviation for the reverberation time of 2 s should be noticeable.

As mentioned previously, Soulodre and Bradley's study (1995) shows that EDT_{octave} yields the highest agreement with the reverberance (r = 0.971) for the tested stimulus among the parameters evaluating this aspect of room acoustics perception. Figure 5 is consistent with their results, as *Conventional* EDT_{octave} is closer to the horizontal axis of zero than *Conventional* $T20_{mid}$. However T_N provides a closer match to the reverberance than EDT_{octave} as it yields the parameter slopes closer to the horizontal zero line. Considering that we tested stimuli similar to those used in Soulodre and Bradleys' study (but convolved with different RIRs, and with artificial gain offsets of -5, 0 and +5 dB), the refined parameter performs better than their best match for reverberance in modelling (and explaining) the effect of gain.

Figure 3 partially confirms the findings of Hase *et al*. (2000). Althougth the RIRs measured in the large auditorium have EDT_{mid} and $T20_{mid}$ substantially longer than for those measured in the small auditorium, the average *d* values shown in the figure are almost identical (less than one unit of the JND) for the RIRs measured in both the auditoria. This shows again that there is not a solid relation between the conventional parameters and reverberance. Lee and Cabrera (2009b) performed an experiment almost identical to that conducted in this paper, except the RIRs used in this study were listened to directly in the previous study (without the noise floor treatment) rather than being convolved with the anechoic music passage. In the previous study, the mean *d* response for L3 was positive (in contrast with the present study's negative response). L3 has a longer reverberation time, but much lower sound pressure level, than the reference stimulus. In the previous study, we interpreted the positive response as reflecting its substantially lower sound pressure level, which is consistent with the relationship between gain and reverberance. This discrepancy between the studies shows a difference between terminal and running reverberance.

It is also interesting to note that the L_{Aeq}, EDT and reverberation time values of S1 are close to those of the reference stimulus (within one JND), but nevertheless it received a significant negative *d* response. The source-receiver distances are also similar (within 2 m) – the only obvious difference is the size of the auditorium, which would affect the fine structure of the reverberation decay. Hence there seem to be factors that affect reverberance that are not reflected in simple measurements of reverberation decay or sound pressure level, although we are not able to explicitly define these in this paper.

CONCLUSION

This study confirms that the gain has a positive effect on the reverberance of music and supports the concept of loudness reverberation time, T_N, which applies a pre-existing objective psychoacoustical model in assessing reverberation decay. T_N accounts for gain changes, unlike the objective reverberation parameters. More research remains to be done in exploring the wider applicability of loudness reverberation time in assessing reverberance. Other loudness-based parameters could also be tested: most obviously, the overall modelled loudness of a RIR may be a good indicator of the 'loudness' of RIRs (in analogy to strength factor).

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