ACCOUNTING FOR LISTENING LEVEL IN THE PREDICTION OF REVERBERANCE USING EARLY DECAY TIME

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Reverberance, which is an auditory attribute describing the extent to which a room or system is reverberant, is conventionally estimated using early decay time (similar to reverberation time). In a series of recent studies, the authors have shown that reverberance is better estimated using loudness decay parameters, i.e., parameters derived from the decay function of a room impulse response analysed using an objective time-varying loudness model. This approach is based on the notion that the experience of sound decaying in a room is an experience of loudness decay. One reason for the success of this approach is that the loudness decay rate depends on listening level, and this dependency corresponds to subjective experimental data on reverberance. However, loudness-based analysis is neither simple nor computationally efficient, and so this paper proposes a simplified approach to reverberance estimation, using listening level to modify early decay time or reverberation time values.

INTRODUCTION

Reverberation is one of the most important features of room acoustics, so many studies have examined ways of predicting the auditory attribute describing the perceived amount of reverberation, which is referred to as reverberance. Well-established objective parameters are reverberation time (T) proposed by Sabine [1] and early decay time (EDT) proposed by Jordan [2]. These two parameters are similar in concept, which is to estimate reverberance by determining the time taken for the reverse-integrated sound pressure envelope following an impulse to decay over a certain decibel range (known as the evaluation range) [3]. These two parameters differ by their evaluation range: the evaluation range for EDT is from the envelope peak to -10 dB; and that for T_{20} is from -5 dB to -25 dB. The EDT evaluation range was inspired by Haas' work [4], which showed the special importance of early reflections in auditory perception. Because EDT emphasises the early decay, it is well-suited to account for the reverberance of running signals such as music, in which the vast majority of sound decays are partially masked by subsequent sound events. The efficacy of EDT over T for estimating reverberance has been demonstrated in subjective listening experiments by Soulodre and Bradley [5] and Barron [6].

A limitation of EDT and T is that these parameters are derived from sound pressure envelope of room impulse responses (RIRs), whereas the perception of sound decay may be more closely related to the loudness envelope of a signal (such as music) in a reverberant environment. As outlined by Zwicker and Fastl [7], many factors (including, but not limited to, sound pressure) affect loudness, and the calculation of loudness takes into account processes such as temporal integration, spectral masking, auditory filter banks, functions relating auditory excitation to specific loudness and so forth.

Previous studies have shown that reverberance is not only affected by the period of sound decay over the evaluation range, but it is also affected by listening level [8-10]. Simply increasing the listening level of a reverberant stimulus yields increased reverberance. This effect occurs for impulsive stimuli and also for music and speech stimuli. In the case of impulsive stimuli, Lee and Cabrera [9] showed that reverberance is related to both the slope of the loudness decay function (when expressed in logarithmic units) and the duration of the audible decay. For music stimuli, the slope of the loudness decay function dominates, because audiences are unlikely to detect long sound decays [10]. Unlike the logarithmic pressure envelope's slope, the loudness decay function's slope varies with listening level (decaying more rapidly when the listening level is reduced), and changing the slope is a plausible way of manipulating the reverberance of stimuli.

Various recent studies have used an auditory model to estimate aspects of reverberance. The approach of van Schuitman and de Vries [11] was to extract the reverberant sound field from an input signal using an auditory model with a peak detection algorithm, and then to average the reverberant sound field from 250 Hz to 4 kHz over the whole duration of the input signal to predict reverberance. For situations where a dry signal and its reverberant counterpart are both available, Uhle et al. [12] proposed a number of reverberance predictors using a loudness model, from which the unmasked part of the reverberant signal could be predicted. Similarly, Zarouchas and Mourjopoulos [13] estimate the perceived sound alteration due to reverberation using a computational auditory masking model. Matsumoto et al. [14] compared the sound pressure decay envelope of RIRs filtered by simplified auditory filters (dynamic compressive Gammachirp filter) and by the conventional band-pass filters, demonstrating that the auditory filters account better for reverberance. These various approaches show that an auditory model can provide more accurate representations of reverberance than the conventional approach.

Lee and Cabrera [9] proposed loudness-based reverberance predictors, T_N and EDT_N (the subscript '*N*' stands for loudness), using computational objective loudness models such as Glasberg and Moore's Time-varying Loudness Model [15] and Chalupper and Fastl's Dynamic Loudness Model [16] (for this purpose of deriving reverberance predictors, these two models perform equally well). After calculating the loudness decay function of a RIR at the relevant listening level, T_N and EDT_N may be calculated in close analogy with their respective counterparts (*T* and *EDT*). The loudness decay function of an RIR is approximately exponential, and so a linear regression can be conducted after taking the logarithm of the function. According to Stevens [17], loudness approximates sound pressure raised to a power of 0.6 for tones of moderate frequency and listening level, which is consistent with the well-known rule-of-thumb that doubling or halving loudness corresponds to ±10 dB. Hence, an evaluation range from peak to half of the peak loudness is used for EDT_N in analogy to *EDT*, and an evaluation range from 0.708 to 0.178 of the peak loudness is analogous to the evaluation range of T_{20}. Details of T_N and EDT_N calculations are described by Lee et al. [9, 10].

While these loudness-based predictors of reverberance have been shown to be substantially more effective reverberance predictors than *EDT*, they are neither straightforward to apply nor easily interpreted. The present paper examines whether a simpler and more accessible approach to estimating reverberance could be made, by using a combination of familiar parameters. Results from the experiments previously conducted by the authors are re-analysed, and a simple model is proposed.

DESCRIPTION OF EXPERIMENTS

Five listening experiments were conducted. These experiments, which have been described in detail previously, followed a similar methodology, and their results have been analysed previously in terms of loudness decay parameters [9, 10, 18, 19]. The participants' task in all experiments was to adjust the reverberance of each stimulus to match that of a reference stimulus. This adjustment was achieved within the computer-based experiment software by altering the decay rate of the room impulse response (RIR) associated with the stimulus, by multiplying it by an exponential function. This was implemented in the experiment software as per Equation 1, where *d* is used to increment or decrement the reverberation time. In equation (1), $p(t)$ is the sound pressure of the RIR, *t* is time in seconds, *d* is the decay rate adjustment value and $p'(t)$ is the sound pressure of decay-rate adjusted RIR. Further details relating to such manipulation of RIRs are given by Cabrera et al. [20]. Hence, the participant would press the 'More' or 'Less' button on the graphical user interface (GUI) to incrementally increase or reduce the reverberation time of each stimulus, so as to perceptually match it to a reference stimulus. Figure 1 shows a screenshot of the Matlab-based GUI used in all the experiments, except for Experiment 1, which was realized using different software (Max/MSP), but with a similar GUI. Note that the maximum stimulus number in Figure 1 was changed for different experiments. The initial value of *d* for each stimulus was randomised by the software. Note that the just-noticeable difference (JND) of reverberance is conventionally given as a 5% change of EDT_{mid} [3], so a unit change of *d* yields a change of 4%. (The subscript "mid" indicates an average of the 500 Hz and 1 kHz octave band values.) A side effect of the decay rate adjustments is a small change in the listening level of comparison stimuli, which was compensated for in the computer-based experiment software before presenting to the subjects. Stimuli were presented via headphones (Sennheiser HD600) and the experiments were conducted in quiet environments. Table 1 provides information about each experiment, including the stimulus signal type, the type of reverberance examined, presentation conditions, and the number of participants (following the removal of a small number of atypical and/or unreliable participants, as described in [9,10]).

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

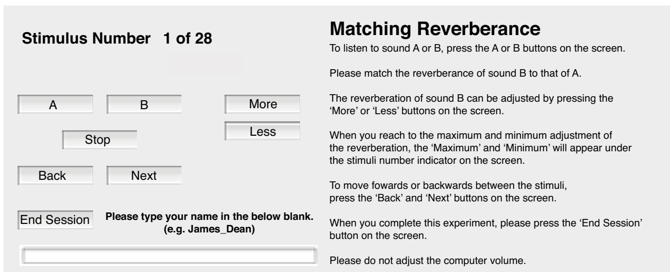

Figure 1. Matlab-based graphical user interface (GUI) used in Experiments II, III, IV and V. The reference stimuli were loaded on the 'A' button and the comparison stimuli were loaded on the 'B' button. The subjects adjusted the reverberance of comparison stimuli by pressing the 'More' or 'Less' buttons on the GUI.

Experiment I (previously reported by Lee and Cabrera [9]) tested impulsive reverberance, by presenting RIRs directly (i.e., without any convolution with a dry source such as music). The experiment used eight RIRs measured by Farina and Ayalon [21] in three auditoria within the *Parco della Musica* in Rome. The small auditorium has 700 seats, medium one has 1200 seats and the large one has 2800 seats (in Table 2 these are labeled 'S', 'M' and 'L' respectively, and the numbers after the letters indicate different receiver positions with a fixed on-stage source). The eight RIRs were recorded with identical equipment and gain, so that the RIRs retain relative levels. In Table 2, the L_{AFmax} is the maximum A-weighted sound pressure level. As the RIRs were single channel, the stimulus presentation over headphones was diotic. In order to investigate the effect of listening level on impulsive reverberance, additional gains of -5, 0 and 5 dB were applied to the RIRs. Therefore, twenty-four comparison stimuli (eight RIRs multiplied by the three additional gain settings) were generated and paired with a single reference stimulus of RIR M1. RIR M1 was chosen as the reference stimulus because it came the mid-sized auditorium, and chosen over the other two stimuli in that auditorium because RIR M2 has the lowest values of conventional parameters and RIR M3 was measured at a source-receiver distance not available in the small auditorium. The listening levels shown in Table 1 take the additional gains of ±5 dB into account.

Experiments II and III (reported by Lee et al. [10]) used the same set of RIRs as Experiment I (including the ±5 dB additional gain), but the RIRs were convolved with anechoic music recordings. For Experiment II, the music was orchestral (bars 1-18 of the *Overture* to *The Marriage of Figaro* by Mozart – which is the same excerpt as that used by Soulodre and Bradley [5]). Because this anechoic recording is stereophonic [22], the presented stimuli (after convolution with a single-channel RIR) are best described as stereophonic, unlike the diotic stimuli used in other experiments. Table 2 shows the L_{Aeq} (the power-average of A-weighted sound pressure level) of the experiment stimuli (at 0 dB gain). For Experiment III, the music was a recording of an opera singer singing the final sixteen bars (11.5 s) of *Torna a Surriento* by Ernesto Di Curtis (which is an Italian song in *bel canto* style). Apart from Experiment III, all of the experiments were conducted in the anechoic room at the University of Sydney; Experiment III was conducted in an audiometric booth in the Advanced Acoustic Information Systems Laboratory at the Research Institute of Electrical Communication at Tohoku University in Japan.

Table 1. Summary of the five experiments

Exp.No.	Stimulus Signal	Type of Reverberance	Headphone Presentation	Listening Level (dBA)	Reverberation Time	No. of Participants
I	Real RIRs	Impulsive	Diotic	58.7 to 80.4	2.01 s to 2.66 s	18
II	Orchestral Music	Overall	Stereophonic	60.1 to 81.0	2.01 s to 2.66 s	16
III	Tenor Singing	Overall	Diotic	60.2 to 82.5	2.01 s to 2.66 s	11
IV	Synthetic RIRs	Impulsive	Diotic	50.0 to 80.0	1.00 s to 3.00 s	10
V	Orchestral Music	Running	Diotic	60.0 to 80.0	1.00 s to 3.00 s	10

Table 2. Source-receiver distance, mid-frequency early decay time (EDT_{mid}), mid-frequency reverberation time (T_{mid}) and maximum sound pressure level (L_{AFmax}) of the RIRs (Experiment I); and equivalent sound pressure level (L_{Aeq}) of dry signals convolved with corresponding RIRs (Experiments II-III)

	S1	**S2**	**M1**	**M2**	**M3**	**L1**	**L2**	**L3**
Distance	12	24	10	19	31	20.5	30	48
EDT_{mid} (s)	1.89	1.98	1.83	1.77	2.00	2.44	2.25	2.38
T_{mid} (s)	2.06	2.07	2.01	2.03	2.17	2.66	2.60	2.53
Exp. I L_{AFmax} (dB)	75.4	74.9	75.0	72.7	70.9	69.9	69.5	63.7
Exp. II L_{Aeq} (dB)	76.0	75.6	75.5	73.7	72.4	71.3	70.7	65.1
Exp. II L_{Aeq} (dB)	77.5	76.1	76.8	74.6	73.5	72.1	71.1	65.2

Experiment IV [18] (like Experiment I) tested impulsive reverberance by presenting RIRs directly as stimuli. However, the RIRs of Experiment IV were synthesized (rather than measured from real rooms). The synthetic RIRs were generated using octave-bands of white noise (centered on 31.5 Hz – 16 kHz), which were multiplied by exponential decay functions, following a simple impulse representing the direct sound. Details of the procedure of generating the synthetic RIRs are provided by Lee et al. [18, 19]. As seen in Table 1, Experiment IV tested impulsive reverberance over a greater range of listening levels and reverberation times than Experiment I. The two main reasons for performing Experiment IV was to determine if the loudness-based predictors (i.e., T_N and EDT_N) perform well over a wider range of listening levels and reverberation times (when reference stimuli also have various listening levels and reverberation times); and to construct equal-reverberance contours for impulsive signals. Figure 2 shows the structure of Experiment IV. In Part A (hereafter, Experiment IV-A), the effect of listening level on impulsive reverberance was tested using reference stimuli with a fixed T value of 2 s and various listening levels (L_{AFmax}) from 50 dBA to 80 dBA. Part B (hereafter, Experiment IV-B) tested the effect of T on impulsive reverberance with reference stimuli having a constant listening level of 60 dBA and various T values ranging from 1 s to 3 s. Four comparison stimuli were paired with each reference stimulus and the participants adjusted the reverberance of comparison stimuli to match the reverberance of the corresponding reference stimulus. Hence, Experiments IV-A and IV-B tested sixteen pairs each (four comparison stimuli multiplied by four reference stimuli). For presentation, the two parts of the experiment were mixed together in randomized order. Two sets of equal-reverberance contours were derived from this experiment. Note that there are four pairs common to IV-A and IV-B, which include the reference stimulus having a listening level of 60 dBA and T of 2 s. In order to shorten the experiment time, they were tested only once, but the results from common pairs were included in the analyses of both parts of the experiment.

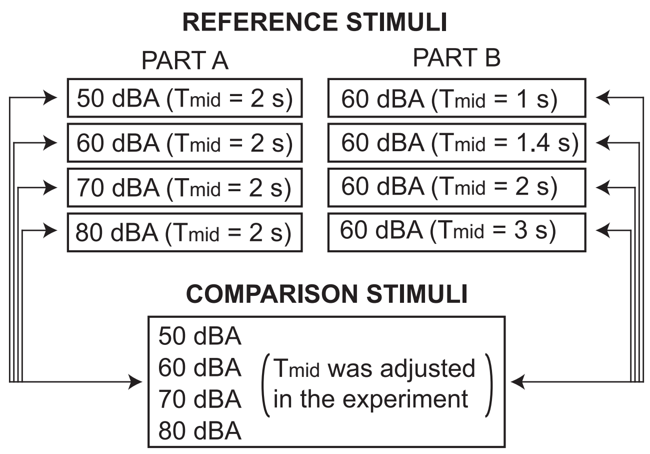

Figure 2. Structure of Experiment IV for Parts A and B

Experiment V [19] used the same synthetic RIRs as Experiment IV, but the RIRs were convolved with an anechoic musical excerpt of *Water Music* by Handel from *Denon Professional Test CDs No.2* [22]. In Experiment V, running reverberance was tested, which is the reverberance experienced while a stimulus is playing [23]. Hence, a very rapid decay was applied to the last note of the convolved musical stimulus in order to eliminate the stopped (or terminal) reverberance following the last note. This experiment was conducted with the same form as Experiment IV, except that the listening level of 50 dBA (L_{Aeq}) was excluded. Hence, Experiment V-A tested nine pairs and Experiment V-B tested twelve pairs. Similarly to Experiment IV, there were three pairs common to V-A and V-B (i.e., when the reference stimulus has a listening level of 60 dBA and T of 2 s) and they were also tested only once to shorten the experiment time.

All experiments yielded significant effects, indicating that listening level and reverberation time both significantly affect reverberance. In Experiments I-III, the RIR was an experimental variable (rather than reverberation time directly), and the effect of RIR was significant. Table 3 shows the analysis of variance (ANOVA) results for Experiments I-III combined. The effect size (which can be expressed as η^2, or the sum of squares for the factor, divided by the total sum of squares) was approximately 1.4 times greater for gain adjustment than for the RIR. While the three experiments yielded significantly different results, the effect of experiment number is substantially smaller than the effects of RIR or gain. There are no significant interactions, as none of the two-factor interaction analyses has a prob>F value less than 0.05 (this indicates that the subjective responses to one independent variable are not affected by another independent variable).

Table 3. Result of the analysis of variance (ANOVA) of Experiments I-III combined, analysed in terms of experiment number (Exp. No.), room impulse response (RIR) and additional gain of ±5 dB (Gain). Values are the sum of squares (Sum Sq.), degrees of freedom (d. f.), mean square (Mean Sq.), the F statistic, significance (prob>F) and effect size (η^2). For a confidence level of 95%, prob>F must be 0.05 or less, and the respective sizes of significant effects are shown as η^2

Source	Sum Sq.	d. f.	Mean Sq.	F	prob>F	η^2
Exp. No.	546	2	272.98	19.85	0	0.028
RIR	1610.2	7	230.02	16.73	0	0.083
Gain	2255	2	1127.48	81.98	0	0.117
Exp. No * RIR	291.5	14	20.82	1.51	0.099	
Exp. No * Gain	43.6	4	10.9	0.79	0.5302	
RIR * Gain	130.1	14	9.3	0.68	0.7996	
Error	14248	1036	13.75			
Total	19259.7	1079				

Due to their more complex structure, the statistical analysis of Experiments IV and V is more involved, and details are given in [18, 19]. In Experiment IV-A, the effects of reference stimulus listening level and comparison stimulus listening level were both significant (p<0.0001), and similarly, in Experiment IV-B, the effects of reference stimulus reverberation time and comparison stimulus listening level were also significant (p<0.0001). In Experiment V (which examined running, rather than impulsive reverberance), the effect of reference stimulus listening level was only significant at 90% confidence in V-A (p=0.0904) but the effect of comparison listening level was significant (p=0.0374); and in V-B the effect of reference stimulus reverberation time was significant (p<0.0001) along with the effect of comparison stimulus listening level (p=0.0276).

In all experiments, it was shown that loudness decay analysis provides a better model for reverberance than conventional parameters such as *EDT* [9, 10, 18, 19]. In the following section the experiment results are re-modeled using a simpler alternative approach.

RE-ANALYSIS

To derive acoustical parameters representing the experiment results, the subjective responses (represented by the decay adjustment value of d) were averaged, and adjusted RIRs were generated using the averaged d values. Since the experiment task was to match reverberance, this procedure yielded sets of RIRs with approximately equal reverberance. Then, the acoustical parameters were derived from these adjusted RIRs. This process was performed for each experiment.

A simple function that expresses reverberance in terms of listening level and *EDT* (or *T*) was sought – and possible functions were tested and refined using the parameters of the adjusted stimuli. Listening level was defined as L_{AFmax} for impulsive stimuli and L_{Aeq} for running (music) stimuli, which was previously shown to be a useful correspondence for loudness-based reverberance modeling [10]. These listening levels were those presented to the participants in the experiments (measured using a Brüel & Kjær type 4100 *Head and Torso Simulator* wearing the employed headphones). Goodness of fit was assessed by the extent to which a function yielded minimal deviation from equal reverberance for each of the sets of equally reverberant stimuli generated from the results of the five experiments.

Equations (2) and (3) are the most successful succinct functions. In these equations, L represents the listening level (L_{AFmax} for impulsive stimuli, L_{Aeq} for music), and the listening level-modified T and EDT are shown with L as a subscript. The exponent acts to compress (or expand) the relationship between the decay time (T or EDT) and the reverberance predictor (T_L or EDT_L), and the extent of this compression or expansion is determined by L. For T, the best fit comes with a unit exponent (i.e., no compression or expansion) when the listening level is 70 dBA; and for EDT, the best fit has a unit exponent when L is 80 dBA. These listening levels (70 and 80 dBA) are, of course, round numbers, but there was little to be gained from the added complexity of using more precisely determined values, given the limited experimental data. An important concept underlying the development of these functions is that the effect of listening level on reverberance is greatest when the reverberation time is long, and Experiments IV and V yielded scarcely any effect of level when the reverberation time was 1 s. The functions only apply to decay times greater than or equal to 1 s (listening level has no effect on the predictor when the decay time is 1 s).

$$T_L = T^{L/70} \quad (T \geq 1\ \mathrm{s}) \tag{2}$$

$$EDT_L = EDT^{L/80} \quad (EDT \geq 1\ \mathrm{s}) \tag{3}$$

Figure 3 compares performance of the proposed parameters ($T_{L,\mathrm{oct}}$ and $EDT_{L,\mathrm{oct}}$) with the conventional parameters (T_{oct} and EDT_{oct}) and loudness-based parameters (T_{N} and EDT_{N}). The subscript 'oct' indicates parameter values averaged over 125 Hz to 4 kHz octave bands. Note that octave-band values of the loudness-based parameters are not available, because the loudness model incorporates integration across the auditory filter-bank. The *y*-axis of the figures shows the coefficient of variation, which is the standard deviation divided by mean. This statistical parameter eliminates a mean-related bias that is likely to exist in the standard deviation (because larger means may be accompanied by larger standard deviations). As the reverberance of all the comparison stimuli was adjusted to that of a reference stimulus, an ideal reverberance predictor should yield a coefficient of variation of zero. As described in the previous section, more than one reference stimulus was used within Experiments IV and V. Hence, the coefficients of variation were calculated over subjective responses for each reference stimulus and theses values were averaged to yield a single-value representation in Figure 3. As seen in the figure, $T_{L,\mathrm{oct}}$ and $EDT_{L,\mathrm{oct}}$ perform similarly well to their respective loudness-based parameter counterparts (T_{N} and EDT_{N}), and in most cases $EDT_{L,\mathrm{oct}}$ performs somewhat better than EDT_{N}. The conventional parameters exhibit the worst performance in every case.

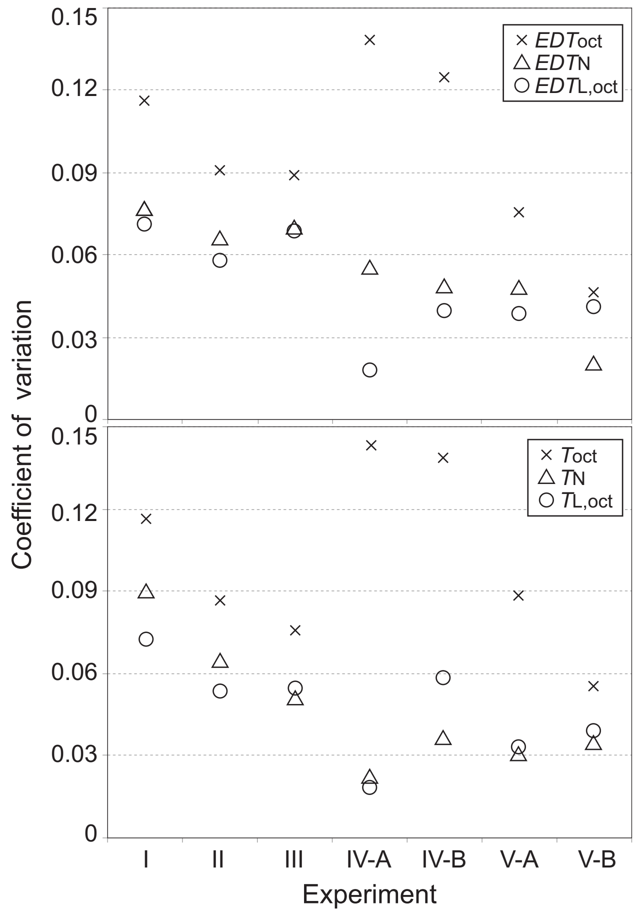

Figure 3. Comparisons of $EDT_{L,\mathrm{oct}}$ with the conventional EDT_{oct} and EDT_{N} (upper figure) and of the modified $T_{L,\mathrm{oct}}$ with the conventional T_{oct} and T_{N} (lower figure). The *y*-axis is the coefficient of variation, which is the standard deviation divided by mean.

As Experiments IV and V tested reference stimuli with various listening levels and reverberation times, the subjective responses obtained from these experiments enable the derivation of equal-reverberance contours. Figure 4 shows these equal-reverberance contours expressed in terms of the conventional parameters (EDT_{oct} and T_{oct}) and the proposed parameters ($EDT_{L,\mathrm{oct}}$ and $T_{L,\mathrm{oct}}$) for Experiment IV, as a function of the listening level of comparison stimuli. An ideal reverberance predictor should yield flat horizontal contours. As seen in the figures, the contours derived from the proposed parameters are much closer to this ideal than those from the conventional parameters. For the conventional parameters,

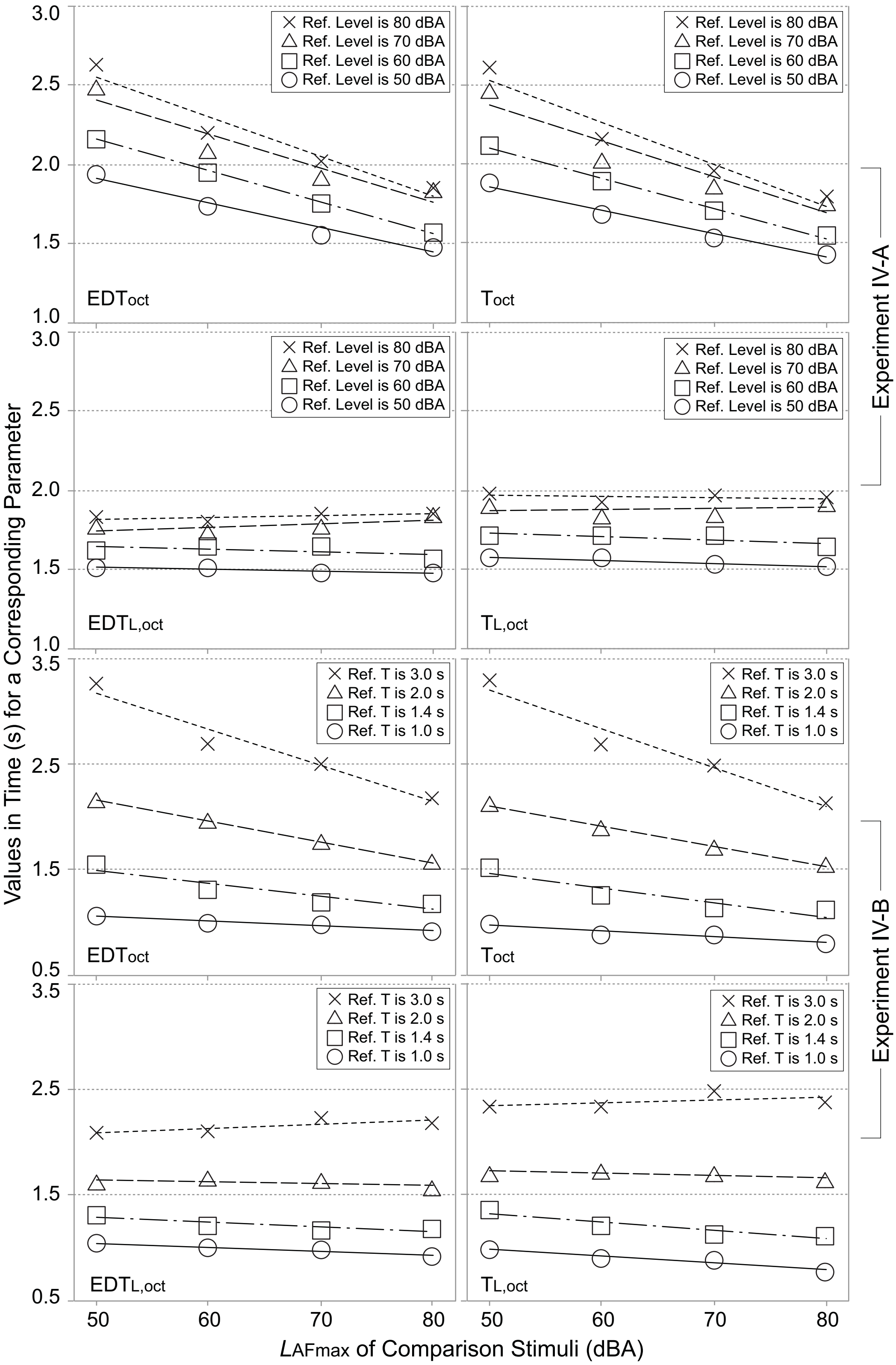

Figure 4. Equal-reverberance contours as a function of the listening level of comparison stimuli for $EDT_{L,\mathrm{oct}}$ and EDT_{oct} (the left four charts) and for $T_{L,\mathrm{oct}}$ and T_{oct} (the right four charts). The four upper charts are for Experiment IV-A, and the four lower charts are for Experiment IV-B.

as the listening level of comparison stimuli increases, the participants reduced the values of conventional parameters to match the reverberance. This implies that the participants experienced greater reverberance when the listening level increases, and this phenomenon becomes stronger as the reverberation time of reference stimuli increases (as shown in the third part of the figure). Table 4 shows the averaged slopes of the equal-reverberance contours derived from Experiments V-A and V-B for the conventional parameters and the proposed parameters. As the ideal equal-reverberance contours are flat horizontal lines, a perfect parameter would yield a value of zero in the table. The table indicates that the proposed parameters outperform the conventional parameters, and this is more substantial for Experiment V-A than Experiment V-B. In Experiment V-B, the proposed parameters exaggerate the effect of listening level on reverberance when the reference stimulus has a T value of 3 s (i.e., the values of the proposed parameters derived from the subjective responses increase with the listening level on this equal reverberance contour). This exaggeration is similar in size to the amount of reduction in the conventional values for the listening level increase. Note that the conventional parameters and the proposed parameters perform similarly when the reference stimuli had T values of 1 s and 1.4 s. When the reference stimulus has a T value of 2 s, the proposed parameters outperform the conventional parameters (for such a reference stimulus, the slope of equal-reverberance contours for EDT is -0.0145, while the slope of the contours for EDT_L is 0.0050). Hence, the proposed parameters' slightly better performance in Experiment V-B is mostly due to the subjective responses for the reference stimulus having a T value of 2 s.

Table 4. The averaged slopes over the equal-reverberance contours derived for EDT_{oct}, $EDT_{L,oct}$, T_{oct} and $T_{L,oct}$ from Experiment V. The values are time (in seconds) per comparison stimulus gain (in dB).

	EDT_{oct}	$EDT_{L,oct}$	T_{oct}	$T_{L,oct}$
Experiment V-A	-0.0148	0.0060	-0.0167	0.0032
Experiment V-B	-0.0090	0.0087	-0.0099	0.0075

DISCUSSION

The approach to modeling reverberance taken in this paper appears to be similarly effective to loudness decay modeling, and yet it is much simpler to apply. Like the loudness-based parameters, it significantly outperforms conventional parameters. The loudness-based parameters are more fundamental, in the sense that they model something of the low level auditory processing that leads to reverberance perception. The simpler approach taken here does not model auditory processing, but merely augments conventional reverberance predictors by reflecting the phenomenon that greater listening level yields greater reverberance.

The proposed models are limited to the range of listening levels and reverberation times shown in Table 1 (50 dBA $\leq L \leq$ 82.5 dBA; 1 s $\leq T \leq$ 3 s), in large rooms with the source well-beyond the near-field, and are based only on music and impulsive stimulus data (speech was not tested, and the tested music was two orchestral excerpts and solo singing). Music stimuli were not tested below 60 dBA, although there may be little practical reason to examine the reverberance of music quieter than this. Clearly, the models do not apply for reverberation times of less than 1 s, because this would invert the positive relationship between listening level and reverberance. Instead, in the absence of further experimental data it would be sensible to presume that listening level has a negligible effect on reverberance for reverberation times of less than 1 s.

The results do not provide a clear indication as to which predictor (EDT_L or T_L) is superior. In the absence of such an indication, it makes sense to choose EDT_L, because EDT is more effective than T for running stimuli (and EDT_L is simple modification of EDT). Figure 5 shows the relationship between EDT and EDT_L evaluated from Equation 3.

EDT_L combines the effects of signal and system to estimate reverberance, whereas parameters used in auditorium design tend to focus on the system alone (because the acoustician has no control over the signals subsequently emitted in an auditorium). Instead of using $L_{AF,max}$ or L_{Aeq} to represent listening level, it may be possible to generalise the approach taken in the present paper to use strength factor, G, in a modified function. Strength factor is a system response characteristic, defined as the difference between the sound pressure level measured from an omnidirectional source in the auditorium (typically with the source on stage) to a receiving position (typically in the audience area), and the sound pressure level measured from the same source (producing the same acoustic power) at a distance of 10 m in an anechoic environment [3]. For this modification to be made, some assumptions would need to be made regarding the power of a typical sound source.

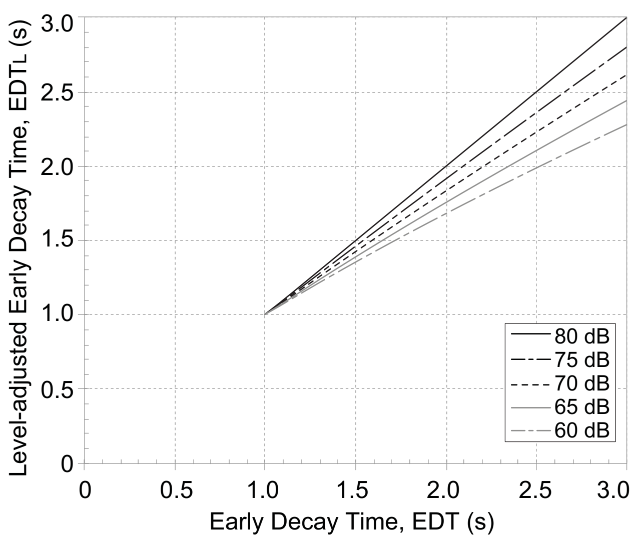

Figure 5. Relationship between the level-adjusted early decay time (EDT_L) and the conventional early decay time (EDT) for various listening levels from 60 dB to 80 dB

CONCLUSIONS

The present study proposes a simple way of more accurately estimating reverberance than offered by the conventional parameters (e.g., *T* and *EDT*) alone, by taking listening level into account. The proposed parameters work well over a range of listening levels and reverberation times commonly found in auditorium listening conditions. Previous studies show that the loudness-based parameters (which involve much more intensive calculation) obviously outperform the conventional parameters, but the present study found that the proposed listening-level modified parameters perform similarly to the loudness-based parameters. Hence parameters of the type proposed in the present paper may be of more practical value for estimating reverberance in many contexts.

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