IMPROVEMENT OF JOT'S REVERBERATION ALGORITHM

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Abstract: The paper considers an artificial reverberation algorithm developed by Jot that simulates the behavior of a real concert hall and takes into account the reverberation time's variation with respect to frequency by using absorbent filters. This work includes both Jot's design of the absorbent filters and an improved version of it. Simulations are run and objective acoustic parameters are computed. The just noticeable difference (JND) is used to find the best artificial reverberator to ensure the best acoustic behavior. To assure the best acoustic behavior the just noticeable difference (JND) are used according to ISO standards.

Keywords: *artificial reverberation, acoustic parameters, digital filters, just noticeable difference.*

I. INTRODUCTION

Natural reverberation is the combined effect of multiple sound reflections within a room: after the source sound stops in a room, the perceived sound decays at a smooth and gradual rate [1]. The effect of reverberation is particularly important for music listening, because it adds life and sense of space [2], [3]. The reverberation is influenced by the architecture and materials of concert halls [4].

The most convenient way to obtain the reverberation effect is by building a digital filter – called reverberator – that will simulate the impulse response of a real concert hall (*Figure 1*). There are lots of reverberation algorithms (reverberators) in the literature [1]. The classical ones (Schroeder, Moorer, Datorro, Gardner) have a common disadvantage: the impossibility to specify a relation between reverberation time and frequency. Starting with Jot's reverberator, this drawback was removed. Reverberators can be compared using the objective and subjective acoustic parameters.

The definitions of objective parameters and the relations between objective and subjective acoustic parameters are presented in the second section of the paper. The third section describes the structure and design of Jot's reverberator and the fourth section presents the design of an improved version of it. The fifth section is devoted to the comparative performance analysis of the reverberators using the acoustic parameters. Finally, the last section includes the conclusions of this work.

II. ACOUSTIC PARAMETERS

The subjective acoustic appreciation as good or bad, pleasant or unpleasant sound cannot give an exact measure of the quality of reverberation. The impulse response (*Figure 1*) provides an accurate description of the acoustic properties of a room and the acoustic standardized parameters related to it measure the room's properties.

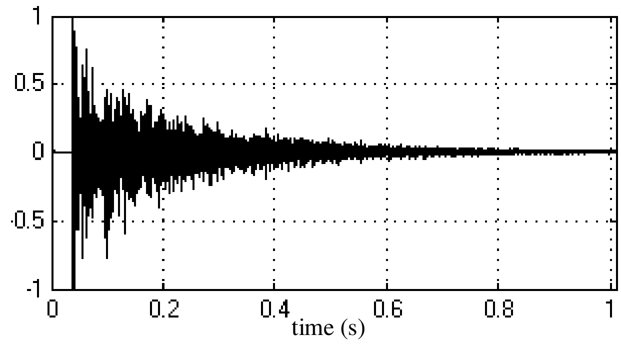

Figure 1. Impulse response of a real concert hall.

A. Objective parameters

The acoustic parameter evaluation is based upon the measurement of the impulse response of the room *h(t)* and the computation of the *energy decay curve* (*EDC*) [1]. The *EDC* is the graphical representation of the decay of the sound pressure level in a room with respect to time, after the sound source has stopped:

$$EDC(t) = \int_t^\infty h^2(\tau)\,d\tau\,. \tag{1}$$

The reverberation time (Tr) is the basic indicator of acoustic behavior and is the time required for the *EDC* to decay by 60 dB. It can be evaluated on a smaller dynamic range and then extrapolated to a 60 dB decay time. If the range is between 5 dB and 35 dB, then the corresponding time is labeled T_{30} [5].

The early decay time (EDT) is defined as the reverberation time computed by the slope of the decay in the range between 0 and –10 dB on the *EDC* [5]. The slope is evaluated by a linear regression line fitted over the appropriate portion of the decay curve. The *EDT* is strongly

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influenced by early reflections, thus depends on the measuring position and the room's geometry. The associated subjective sensation of reverberation is strongly correlated with *EDT*. The *EDT* is expected to have the same value as the reverberation time for a room exhibiting perfectly linear decay.

Clarity Index (C_{80}) is an early-to-late arriving sound energy ratio, relevant for music [2], [3], [5]:

$$C_{80} = 10 \cdot lg \frac{\int_{0}^{80ms} h^2(t)\,dt}{\int_{80ms}^{\infty} h^2(t)\,dt} \quad [dB] \ . \tag{2}$$

Definition (D_{50}) measures the early to total sound energy ratio. D_{50} is mostly used for the speech case.

$$D_{50} = \frac{\int_{0}^{50ms} h^2(t)\,dt}{\int_{0}^{\infty} h^2(t)\,dt} \quad [\%] \, . \tag{3}$$

Both clarity and definition are measures of the distinctness and clarity of speech and music [2], [4], [5].

The center time (T_c) corresponds to the center of gravity of the squared impulse response [2], [4], [5]:

$$T_c = \frac{\int_{0}^{\infty} t \cdot h^2(t)\,dt}{\int_{0}^{\infty} h^2(t)\,dt} \quad [\mathrm{s}] \, . \tag{4}$$

B. Relation between objective and subjective parameters.

Subjective studies of the acoustic characteristics of auditoria have shown that several quantities that can be obtained from measured impulse responses are correlated with particular subjective aspects of the acoustic character of an auditorium. In our paper, these subjective quantities are used for the performance analysis of reverberators.

The *just noticeable difference* (*JND*) is the smallest change in a parameter that is necessary for a human to detect a difference [5].

Table 1 presents the analyzed parameters and the associated *JND* values [5]. These values are computed at the midrange frequency (average at 500 Hz and 1000 Hz). Although the reverberation time is the representative physical characteristic of a room, a good practice is to allocate 5% *JND* for the reverberation time at the mid frequency range.

Table 1. Acoustic parameters and their JND values.

Subjective listener aspect	**Acoustic parameter**	***JND***
Perceived reverberation	*EDT*	5 %
Perceived clarity of sound	Clarity C_{80}	1 dB
	Definition D_{50}	0,05
	T_s	10 ms

III. JOT'S REVERBERATION ALGORITHM

Jean-Marc Jot proposed an algorithm that uses absorptive filters in order to model the air absorption [6]. A relation between the reverberation time and frequency response may be specified.

The reverberator shown in *Figure 2* [1] can be considered as a generalization of the Schroeder's parallel comb filters with a diagonal feedback matrix $\mathbf{A}_N$. The feedback matrix permits the delay cell output to be recirculated at the cell input, where the matrix coefficients control the weights of the feedback taps. A convenient matrix may be taken from the class of Householder matrices:

$$\mathbf{A}_N = \mathbf{J}_N - \frac{2}{N} \cdot \mathbf{u}_n \cdot \mathbf{u}_N^T \tag{5}$$

where $\mathbf{J}_N$ is a $N \times N$ permutation matrix and $\mathbf{u}_N$ is an $N \times 1$ column vector.

The reverberation time depends on the frequency, so the delay cells are each associated with an absorption filter ($h_1(z)...h_4(z)$). The absorption filter simulates the variation of the reverberation time with frequency and the energy decay relief for a real room.

The filter's gain depends on the delay m_i and the reverse of the reverberation time $Tr(\omega)$:

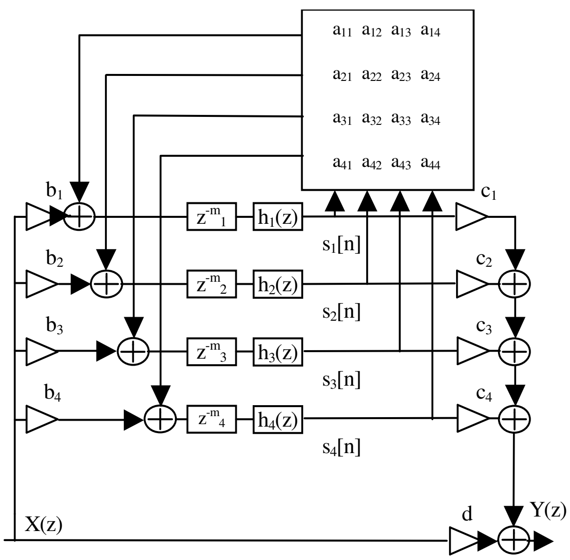

Figure 2. Jot's reverberator.

$$20 \log \left| h_i \left(e^{j\omega} \right) \right|_{dB} = \frac{-60 \cdot T}{Tr(\omega)} \cdot m_i \ . \tag{6}$$

The reverberation time being smooth with frequency, the filters are of low order and have the transfer function:

$$h_i(z) = g_i \cdot \frac{1 - b_i}{1 - b_i \cdot z^{-1}} , \tag{7}$$

where g_i provides the reverberation time for low frequencies *Tr(0)* and b_i for high frequencies *Tr(π)* [6]. The sizing equations are:

$$g_i = 10^{-3 m_i \cdot T / Tr(0)} ; \tag{8}$$

$$b_i = 1 - \frac{2}{1 + g_i^{(1 - 1/\alpha)}} ; \tag{9}$$

$$\alpha = \frac{Tr(\pi)}{Tr(0)} \ . \tag{10}$$

IV. IMPROVEMENT OF JOT'S REVERBERATION ALGORITHM

The Jot's original absorptive filter design method implies to choose 2 points on the reverberation time curve: at low frequency and at half of the sampling frequency. The sampling frequency being 44100 Hz and the audio frequency range being beyond 16000 Hz, Jot's design is impossible to be applied. At the same time, the ISO standards state that 2 points are not enough to reproduce with high precision all the acoustic parameters [5].

We improved the Jot's reverberator by considering the second point at a high frequency φ:

$$b_i = \frac{z_i \cos\varphi - 1 + \sqrt{z_i \left(z_i \cos^2 \varphi - z_i - 2\cos\varphi + 2 \right)}}{z_i - 1} ; \tag{11}$$

$$z_i = g_i^{2 \cdot \left(\frac{1}{\alpha} - 1 \right)} ; \tag{12}$$

$$\alpha = \frac{Tr(\varphi)}{Tr(0)} . \tag{13}$$

Let us consider the feedback matrix of size 4×4:

$$J_4 = \begin{bmatrix} a_{11} & a_{12} & a_{13} & a_{14} \\ a_{21} & a_{22} & a_{23} & a_{24} \\ a_{31} & a_{32} & a_{33} & a_{34} \\ a_{41} & a_{42} & a_{43} & a_{44} \end{bmatrix} = \begin{bmatrix} 0 & 0 & 1 & 0 \\ 0 & 1 & 0 & 0 \\ 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 \end{bmatrix} . \tag{14}$$

Table 2 presents the designed values for the Jot's original method. *Table 3* presents the designed values for the improved method. It is worth mentioning that the improved reverberator used 2 values at high frequencies: 2500 Hz and 16000 Hz.

Table 2. Designed values for Jot's reverberator.

Delay m_i (samples)	***Tr(0)* (s)**	***Tr(π)* (s)**	**α_i**	**g_i**	**b_i**
3001	1.757	0.3	0.1707	0.7653	0.5715
3089	1.757	0.3	0.1707	0.7593	0.5841
3191	1.757	0.3	0.1707	0.7524	0.5985
3259	1.757	0.3	0.1707	0.7479	0.6079
3347	1.757	0.3	0.1707	0.7420	0.6197
3499	1.757	0.3	0.1707	0.7320	0.6396
3581	1.757	0.3	0.1707	0.7267	0.6500
3637	1.757	0.3	0.1707	0.7231	0.6569
3739	1.757	0.3	0.1707	0.7165	0.6693
3863	1.757	0.3	0.1707	0.7087	0.6838
3967	1.757	0.3	0.1707	0.7021	0.6956
4051	1.757	0.3	0.1707	0.6969	0.7049
4139	1.757	0.3	0.1707	0.6914	0.7144
4289	1.757	0.3	0.1707	0.6822	0.7299
4397	1.757	0.3	0.1707	0.6757	0.7406
4507	1.757	0.3	0.1707	0.6691	0.7512

Table 3. Designed values for the improved Jot's reverberator.

Delay m_i (samples)	**φ (Hz)**	***Tr(φ)* (s)**	**g_i**	**α_i**	**z_i**	**b_i**
3001	16000	0.42	0.7653	0.2390	0.1821	0.4347
3089	16000	0.42	0.7593	0.2390	0.1732	0.4451
3191	16000	0.42	0.7524	0.2390	0.1635	0.4570
3259	2500	1.522	0.7479	0.8662	0.9142	0.3328
3347	2500	1.522	0.7420	0.8662	0.9120	0.3375
3499	2500	1.522	0.7320	0.8662	0.9082	0.3453
3581	2500	1.522	0.7267	0.8662	0.9061	0.3494
3637	2500	1.522	0.7231	0.8662	0.9047	0.3521
3739	2500	1.522	0.7165	0.8662	0.9022	0.3570
3863	2500	1.522	0.7087	0.8662	0.8991	0.3629
3967	2500	1.522	0.7021	0.8662	0.8965	0.3676
4051	2500	1.522	0.6969	0.8662	0.8945	0.3714
4139	2500	1.522	0.6914	0.8662	0.8923	0.3753
4289	2500	1.522	0.6822	0.8662	0.8886	0.3817
4397	2500	1.522	0.6757	0.8662	0.8860	0.3862
4507	2500	1.522	0.6691	0.8662	0.8833	0.3907

V. PERFORMANCE ANALYSIS

A comparative analysis was carried out for both original and improved Jot's reverberators designs. The reverberator algorithms were implemented in Matlab's Simulink [13] and the obtained data were analyzed with WINMLS [12], a software for performing and evaluating audio, acoustic and vibrational measurements. The acoustic parameters of the original Jot's reverberator, improved Jot's reverberator and a real concert hall are plotted with respect to the frequency in *Figure 3*.

An objective analysis can be made only for the frequencies: 500 Hz and 1 kHz, using the ISO 3382 standard. For higher frequencies we have not any information about the *JND* values. For these frequencies the performance analysis is reduced to visual inspection.

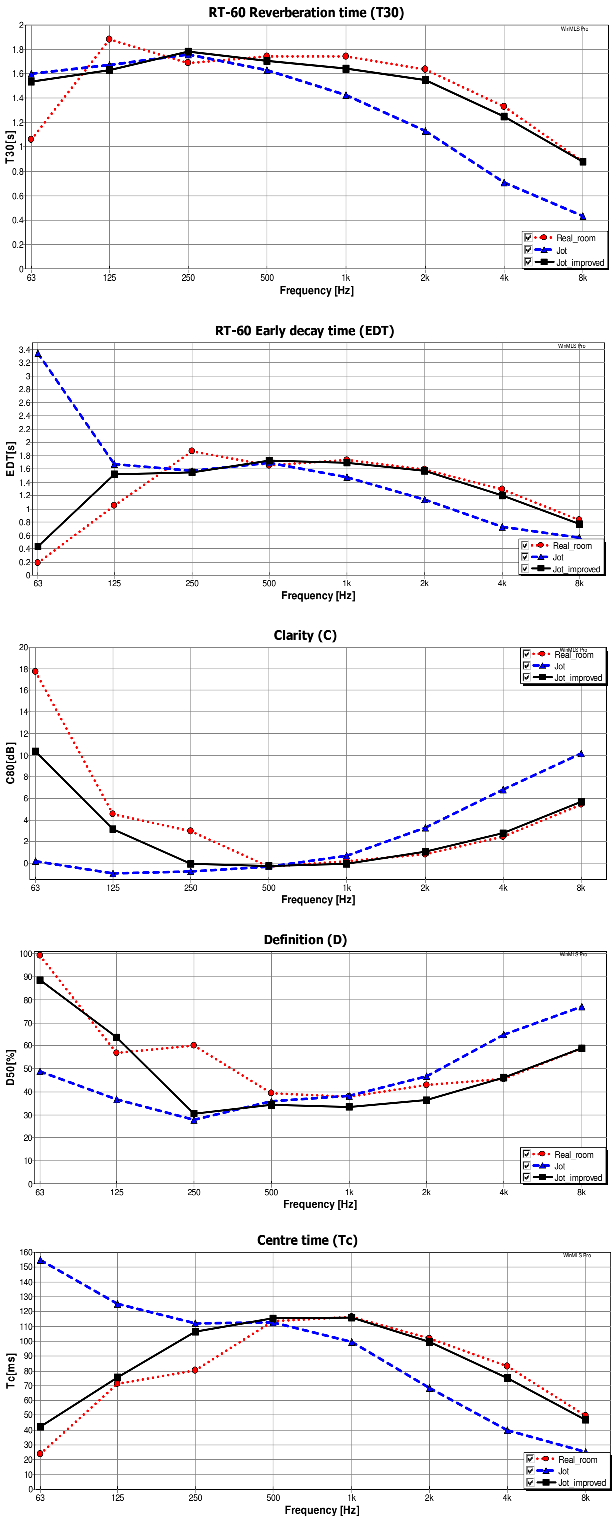

Figure 3. Acoustic parameter's evaluation.

By inspecting *Figure 3*, one can see that after 1 kHz, the improved Jot's version models more accurately the real room. For 500 Hz and 1 kHz, as seen in *Table 4*, the improved Jot's reverberator is better than the original one; the differences between the real room and the improved version acoustic parameters are below 1 *JND*, except the definition.

Table 4. Acoustic parameter's mean value for 500 Hz, 1 kHz and the corresponding JND values.

Acoustic parameter's mean value for 500 Hz, 1 kHz and *JND*	**Real room**	**Jot**	**Jot improved**
Reverberation time (s)	1.74	1.53	1.68
		2.42 *JND*	0.74 *JND*
Early decay time (s)	1.69	1.59	1.71
		1.20 *JND*	0.19 *JND*
Clarity (dB)	-0.035	0.21	-0.13
		0.25 *JND*	0.095 *JND*
Definition	0.39	0.37	0.335
		0.4 *JND*	1.1 *JND*
Centre time (ms)	114.5	106.4	115.7
		0.81 *JND*	0.12 *JND*

VI. CONCLUSIONS

The paper presents an improvement of Jot's algorithm for artificial reverberation. Simulations were performed in Matlab's Simulink and the output data analyzed with WINMLS.

The Jot's algorithm was improved due to the following reasons:

- Jot's original algorithm requires the reverberation time at half of sampling frequency, that is outside of the audio range;
- the ISO definition of acoustic parameters states that 2 points are not enough to reproduce all the acoustic parameters with high precision.

To measure the behavior of the proposed reverberator, the following acoustic parameters: reverberation time (*T30*), early decay time (*EDT*), clarity (*C80*), definition (*D50*) and centre time (*Tc*) were computed and plotted. The objective parameters of the Jot's and improved Jot's reverberators were compared to the ones of a real concert hall.

To have a subjective measure of the acoustics the *JND* at 500 Hz and 1 kHz were also computed. The best behavior is the one of the improved Jot's reverberator. At other frequencies the comparison is done only by visual inspection. The improved Jot's algorithm models more accurately the acoustic properties of the real room than the Jot's reverberator.

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