Evaluation of All-Pass Reverberators

Marina Dana Țopa¹, Norbert Toma¹, Victor Popescu¹, Vasile Țopa¹
¹Technical University of Cluj-Napoca
Str. Ctin. Daicoviciu nr. 15, 400020 Cluj-Napoca, Romania
phone: + (40) 264 401243, fax: + (40) 264 591689, email: mtopa@bel.utcluj.ro

Abstract — The paper presents the structure and the performance of signal processing algorithms that simulate natural room reverberation. It deals with three digital complete artificial reverberators, built up of Schroeder’s early reverberator and a late reverberator built with all-pass filters: Schroeder’s, Gardner’s or absorbent all-pass reverberator. For each of the presented reverberation algorithms acoustic parameters are computed and compared with those of a concert hall.

I. INTRODUCTION

Natural reverberation is the combined effect of multiple sound reflections within a room [1]. After the source sound stops, reverberation in a room causes the perceived sound to decay at a smooth and gradual rate. The effect of reverberation is particularly important for music listening, because it adds life and sense of space. The reverberation is associated with the architecture and acoustic of concert halls.

The reverberation starts with production of a sound in a room. The acoustic wave meets the walls, ceiling and other surfaces, where the energy is absorbed and reflected. The reflected energy leads to reverberation.

Figure 1 presents the impulse response of a concert hall. The impulse response provides an accurate description of the acoustical properties of the room. If a direct path exists between the source and the listener, the listener will hear first the direct sound followed by reflections of the nearby surfaces, these being called early reflections. After some tens of milliseconds, the number of reflected waves becomes very large with a decreased evolution in time, characterized by a dense collection of echoes moving in all directions, their intensity being independent from the location in the room. These are late reflections.

II. ACOUSTIC PARAMETERS

The most important acoustic parameters are defined with the help of the impulse response and related plots [1]. The energy decay curve (EDC) is obtained by integrating the impulse response $h(\tau)$:

$$ EDC(t) = \int_{-\infty}^{\infty} h^2(\tau) d\tau. \quad (1) $$

EDC computes the energy remaining in the impulse response after the time $t$.

The energy decay relief (EDR) is the time-frequency representation of reverberation. This can be done as follows: start with the impulse response, bandpass filter it into frequency bands, compute the EDCs and display the result as a 3-D surface.

The standardized parameters [2], [3] are the following:

- **The reverberation time** $T60$ is the time required for the EDC to decay 60 dB. It is not necessary to acquire a full 60 dB of decay by measurement, but a smaller portion of the available dynamic range may be evaluated and the result simply scaled to 60 dB. $T3$ ($T20$) is the time required for the EDC to decay from -5 to -35 dB (-25 dB) and then scaled to -60 dB.
- **The early decay time** (EDT) was defined as the reverberation time from the decay range between 0 and -10 dB on the EDC [3].
- **Clarity** ($C50$, $C80$) is defined as the logarithmic ratio of an impulse response’s energy before time $t_e$ and the energy after $t_e$, where $t_e$ equals 50 or 80 ms. When the clarity is related to the musical perception, the time interval $t_e$ is limited to 80 ms, whereas if the clarity is related to speech, $t_e$ is set to 50 ms.
The clarity is the ratio, expressed in dB, between the “useful energy” which is received in the first 50 (80) ms and the “detrimental energy” which is received afterwards [3], [4].

- **The definition** is similar to clarity, but expressed in % instead of dB [5]:

$$D_e = \frac{\int_0^t h^2(t)dt}{\int_0^\infty h^2(t)dt} \cdot 100\%$$  \hspace{1cm} (3)

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

- **The center time (Tc)** corresponds to the center of gravity of the impulse response energy:

$$T_c = \frac{\int_0^\infty t \cdot h^2(t)dt}{\int_0^\infty h^2(t)dt}$$  \hspace{1cm} (4)

A low Tc suggests a sensation of clarity, whereas large values of Tc suggest a reverberant sound [2].

### III. REVERBERATION ALGORITHMS

This section describes an early and three late all-pass reverberation algorithms used in sound systems [10], [11].

- **Schroeder’s early reverberator**

  The FIR structure proposed by Schroeder (Figure 2) contains a set of delay cells \( m_i \), taps \( a_i \) generating equally spaced echoes and a low-pass filter (LPF) for improving the sound quality. The delay times are chosen between 10 and 80 milliseconds, but there is no recipe for the choice of the gains \( a_i \) and parameters of the low-pass filter.

- **All-pass late reverberators**
  - **Schroeder’s all-pass reverberator**
    It is a cascade of 5 all-pass filters. (Figure 3). Using this structure, single reflections are expanded into many reflections that build an entire infinite all-pass impulse response [6], [7].
  - **Gardner’s reverberator**
    It is based on nested all-pass filters, where the delay is replaced by a series connection of a delay and another all-pass filter. Gardner suggested three structures for different size rooms [8]. In Figure 4 the structure of the medium room reverberator is depicted.
  - **Absorbent all-pass reverberator**
    A valid method for designing reverberators is based on cascading all-pass filters (Figure 5). The so-called “absorbent all-pass reverberator” contains a chain of six absorbent all-pass filters (AA1÷AA6) [9].

### IV. ANALYSIS OF ALL-PASS REVERBERATION ALGORITHMS

A complete reverberator has the structure depicted in Figure 6. The Delay must be adjusted to obtain a correct early-late energy ratio. The all-pass filter \( APF \) simulates a diffuse sound
energy component between early reflections. The Late reverberator is one of the three algorithms presented in the 3-rd section.

The early reverberation starts after the time gap at approximately 40 milliseconds and finishes at about 80 milliseconds. Figure 7 presents the early reverberation of the concert hall whereas Figure 8 the modeled early reflections using an early Schroeder reverberator, that is quite close to the natural phenomenon.

The quality of the complete reverberators was measured with the help of the parameters described in section 2. The algorithms were implemented in Matlab’s Simulink [13] and the obtained data was analyzed with WINMLS [12], a software for performing and evaluating audio, acoustical and vibrational measurements. The Figures 9 ÷ 12 show the variation of the acoustic parameters curves with respect to the frequency for all the three reverberators in comparison with the real concert hall.

If for the reverberation time T30 (Figure 9), the absorbent all-pass is the best choice (identical values for 250-2000Hz and least errors for lower and higher frequencies), for the early decay time EDT (Figure 10) the absorbent all-pass and Schroeder reverberators behave similarly: for 250-2000 Hz very close to the real room, but for lower frequencies the Schroeder is the best and for higher frequencies the absorbent all-pass reverberator is the best. Regarding speech, Figures 11 and 12 show that the best algorithms are: the absorbent all-pass and the Schroeder ones.
The energy decay relief (EDR) diagrams from the Figures 13÷16 show an overall vision of the different reverberation algorithms. These pictures, as well as the Figures 9÷12 state that the best choice among the three presented reverberators is the absorbent all-pass one: it models very closely the real room for frequencies between 250 and 2000Hz and with acceptable errors beyond and below these frequency limits.

V. CONCLUSIONS

The paper presents the quality evaluation of some all-pass algorithms for artificial reverberation used in enhancement systems. Artificial reverberation algorithms involve two processes: the modeling of early reflection response corresponding to the early echoes and the simulation of the late reverberation or diffuse reverberation. The early (Schroeder’s) reverberator provides a set of equally spaced impulses with a diffuse sound energy. The late reverberator was chosen to be based on all-pass filters and is a late Schroeder’s, a Gardner’s or an absorbent all-pass reverberator. The main problem of the late reverberator is to model the frequency dependence of the absorption.

The reverberation algorithms were simulated in Matlab’s Simulink and the output data analyzed with WINMLS software. The following acoustic parameters were computed: reverberation time (T30), early decay time (EDT), clarity (C80) and centre time (Tc). The energy decay reliefs (EDR) present an overall vision of the reverberation algorithms.

The acoustic parameters of the three reverberators were compared with the ones of a concert hall. For high frequencies the Schroeder’s and Gardner’s reverberator properties are deteriorated due to inexistence of a relation between reverberation time and frequency. The absorbent all-pass reverberator approximates best the concert halls characteristics: it models perfectly the real room for frequencies between 250 and 2000Hz and with acceptable errors beyond and below these frequency limits.

REFERENCES