Doctoral School on Engineering Sciences
Università Politecnica delle Marche

Extended summary

Advanced Algorithms for Audio Quality Improvement in Musical Keyboards Instruments

Curriculum: Ingegneria Elettronica, Elettrotecnica e delle Telecomunicazioni

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Date: 23-01-2013

Abstract. The focus of this thesis is on the signal processing techniques used to increase the audio quality of the most common digital audio effects employed in electronic musical instruments also taking into account the feasibility of the proposed algorithms' implementation in accordance with the design constraints and the available computational limits. More in detail four different issues have been analyzed throughout this dissertation: artificial reverberation, analysis and emulation of nonlinear devices, audio morphing, and room response equalization.

Among the audio effects, one of the most used is definitely artificial reverberation. A great deal of research has been devoted in the last decades to improve the performance of digital artificial reverberators. Thanks to the progress of technology the traditional techniques based on recursive structures (i.e., IIR filters) are accompanied by new approaches based on fast convolution techniques and hybrid reverberator structures. On this basis, an efficient real-time implementation of a fast convolution algorithm has been proposed taking into account an embedded system. Moreover, a technique for reducing the computational load required by this operation using psychoacoustic expedients has been presented considering a joint assessment of the energy decay relief and the absolute threshold of hearing. Finally, some techniques for the approximation of the convolution operation with recursive structures at low computational cost have been suggested.

Although the convolution operation allows the exact reproduction of a linear system, it is important to consider that most of the audio effects are nonlinear systems (i.e., compressors, distortion, amplifiers). For this reason, the most commonly used techniques for the emulation of nonlinear systems based on a black box approach have been studied and analyzed. In particular, a technique for the approximation of the dynamic convolution operation by exploiting the principal component analysis has been proposed. Using this procedure it is possible to reduce the cost of dynamic convolution without lowering the perceived audio quality. An adaptive algorithm for the identification of nonlinear systems using orthogonal functions has also been presented.

In order to provide greater flexibility and major artistic expression to musicians, several audio morphing techniques have been analyzed. In particular, this procedure makes possible to combine two or more audio signals in order to create new sounds that are acoustically interesting. This study has led to the development of an audio morphing algorithm for percussive hybrid sound generation. The main features of the presented approach are preprocessing of the audio references performed in the frequency domain and time domain linear interpolation to execute the morphing.

Finally, equalization techniques for improving the quality of sound reproduction systems by compensating the room transfer function have been taken into account. In particular, two algorithms for adaptive minimum-phase equalization and a mixed-phase equalization technique have been proposed. In order to verify the suitability of the proposed systems, experiments on a realistic scenario have been carried out.

Keywords. Digital audio effects, Fast convolution, Artificial reverberation, Hybrid reverberator, Nonlinear system approximation, Dynamic convolution, Nonlinear convolution, Audio morphing, Room response equalization.

1 Problem statement and objectives

Technology is the tool that has allowed music to develop over time, and has ensured artists have ever-growing possibilities of communication and expression. The scientific development, which led to the construction of pianos and violins, today, thanks to digital signal processing techniques, is allowing the creation of new tools and new musical forms. On this basis, engineering activity now works hand in hand with traditional craft aiming to modify the means of musical expression adapting them to today's socio-cultural context. Since the 70s, the progress of technology has allowed, through sound engineering and digital signal processing techniques, the artificial reproduction of many sound effects that can be used in all forms of musical expression. Since then, the development and the recent deployment of commercial embedded systems at high computational power, pave the way for the development of new innovative commercial products, characterized by high sound quality, expressiveness, and realism.

In this dissertation, the problem of audio quality improvement in musical keyboard instruments has been addressed. This work is intended to provide an overview of the most used audio effects in musical keyboard instruments and also to consider the cutting-edge technologies and the most recent algorithms useful to guaranteeing a high quality acoustic reproduction. Although focalized on the analysis and implementation of high quality audio effects, this work also focuses on the implementation aspects of these algorithms. Indeed, for most of the presented techniques, real-time implementations using commercial DSP platforms have been presented also reporting the expedients and the tricks introduced in order to meet the design constraints lowering the required computational cost demand.

Starting from the consideration that reverb is probably one of the audio effects most employed by musicians [1], a great deal of research has been devoted in the last decades to improve the performance of digital artificial reverberators. Thanks to the progress of technology the traditional techniques based on recursive structures (i.e., IIR filters) [2] [3] [4] [5] [6] [7] are accompanied by new approaches based on fast convolution techniques [8] [9] [10] [11] [12] [13] [14] [15] [16] [17] [18] and hybrid reverberator structures [19] [20] [21] [22]. On this basis several innovative contributions have been proposed in the field of artificial reverberation. Regarding the fast convolution techniques, a psychoacoustic approach has been proposed aiming to reduce the computational cost required to perform the convolution operation considering the human ear sensitivity [23]. In more detail, taking into account a joint assessment of the energy decay relief and the absolute threshold of hearing, this technique allows to decrease the number of complex multiplications needed to perform the convolution operation without lowering the perceived audio quality. As a matter of fact, a PC [24] and a DSP [25] based implementations of a fast convolution algorithm has been proposed employing a non uniform partitioning of the impulse response. Moreover, several techniques for the approximation of the convolution operation, occurring in low computational complexity structures (e.g., IIR filters), have been explored. Three hybrid reverberation algorithms [26] [27] [28] have been presented in more detail, together with an automatic procedure for tuning the hybrid reverberator parameters taking into account a real impulse response to be emulated.

Although fast convolution techniques allow the exact emulation of a time invariant linear system, most of sound effects are classified as nonlinear systems and just few of them can be approximated using linear convolution. Starting from this consideration, several techniques for the approximation of nonlinear audio effects have been explored taking into account the most common state-of-art approaches useful to accomplish black box modeling (i.e., nothing about the internal structure of the nonlinear system is known) [29] [30] [31] [32] [33]. More in detail, regarding the dynamic convolution operation, a new technique useful to approximate the convolution operation exploiting the principal component analysis (PCA) has been presented in [34] [35]. While, an adaptive identification procedure based on orthogonal nonlinear functions has been proposed in [36].

In order to provide a greater flexibility and a major artistic expression to musicians, several audio morphing techniques have been proposed in the literature during the last decades [37] [38] [39] [40]. Thus, with the aim of combining two or more percussive samples to create new perceptually interesting sounds with intermediate timbre and duration, an automatic audio morphing procedure for hybrid percussive sound generation has been presented [41].

Finally, to better emphasize the sound produced by musical keyboard instruments, the room equalization problem has been taken into account. Aiming to improve the objective and subjective quality of sound reproduction systems by compensating the room transfer function, numerous approaches have been proposed in the literature considering both single [42] and multiple positions [43] [44] [45] [46] [47] [48] equalization. On this basis, taking into account the multiple position equalization problem three room response equalizers have been developed based on minimum-phase [49] [50] and mixed-phase solutions [51].

2 Research planning and activities

PhD activity has been organized starting from the state of the art of the chosen topics. Subsequently, the novel ideas have been investigated, first, from a theoretical point of view and then, through experimental tools, such as Matlab and the Nu-Tech framework. More specifically, Nu-Tech is a platform that allows to implement and test real-time algorithms even in multi-channel scenarios [52]. Moreover, for most of the presented techniques, real-time implementations using commercial DSP platforms have been presented also reporting the expedients and the tricks introduced in order to meet the design constraints lowering the required computational cost demand.

Several international conferences were attended in order to present the novel contributions and to have a chance of relating with other researchers interested in the same topics (128th AES Convention, London, UK, 22-25 May 2010; 130th AES Convention, London, UK, 13-16 May. 2011; EUSPICO2011, Barcellona, ES, 29 Ago.-2 Sep. 2011; 45th AES Conference, Helsinki, FI, 29 Feb.-4 Mar. 2012; EDERC2012, Amsterdam, NL, 9-15 Sep. 2012; 133th AES Convention, San Francisco, US, 25 Oct.-1 Nov. 2012.). In addition, the research activity from Apr 2011 to Jul 2011 was carried out at Queen Mary University, London (Center for Digital Music) under the supervision of Dr. Joshua D. Reiss.

3 Analysis and discussion of main results

This work focuses on the signal processing techniques used to increase the audio quality of musical keyboard instruments. More in detail, taking into account the most used audio effects employed in electronic musical instruments, four different issues have been analyzed throughout this dissertation: artificial reverberation, analysis and emulation of nonlinear devices, audio morphing and, room response equalization.

3.1 Artificial reverberation

Several innovative contributions have been proposed in the field of artificial reverberation with the aim of enhance the obtained audio quality exploiting fast convolution algorithms and hybrid structures.
For what concerns the fast convolution techniques, an embedded DSP-based implementation of a fast convolution algorithm has been presented in [25], employing a non uniform partitioning of the impulse response (NUPOLS) and taking into account the OMAPL137 platform. In order to reduce the computational cost required to perform the convolution operation, several DSP expedients have been introduced (i.e, efficient memory management based on non blocking DMA, basic psychoacoustic approach, manual code partitioning).
Figure 1 shows the workload required to perform a NUPOLS on the DSP platform OMAP-L137 as a function of the impulse response length for different partitionings. The presented results demonstrate the feasibility to implement long convolution operations with a low input/output latency on general purpose DSPs.

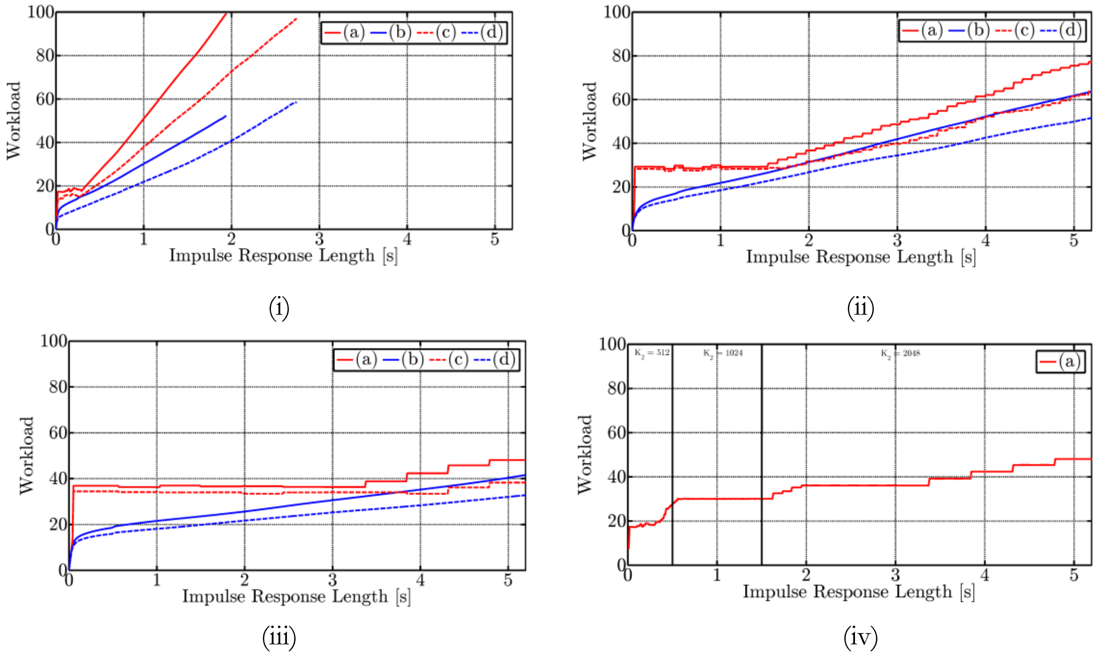
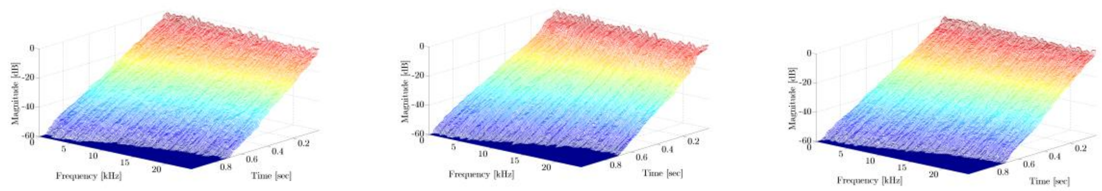


Figure 1. Non uniform partitioned overlap and save workload as a function of 4 different partitionings (K_1=64 K_2=512 (i), K_1=64 K_2=1024 (ii), K_1=64 K_2=2048 (iii), optimal partitioning (iv)). Max (a) and mean (b) workload for classic implementation. Max (c) and mean (d) workload using psychoacoustic approach.

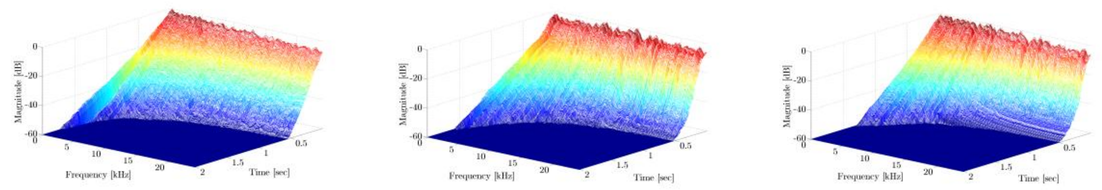
On the same basis, a PC-based implementation of a fast convolution algorithm has been proposed in [24] considering: a multithreading implementation, psychoacoustic expedients, and an automatic impulse response partitioning.
In order to reduce the computational cost required to perform the convolution operation taking into account the human ear sensitivity, a psychoacoustic approach has been presented in [23]. More specifically, considering a joint assessment of the energy decay relief and the absolute thresholds of hearing, this approach allows to reduce the number of complex multiplications needed to perform the convolution operation without altering the perceived audio quality.

As audio quality, computational saving is an important aspect in sound effect evaluation. On this basis, several techniques for the approximation of the convolution operation, exploiting low computational complexity structures (e.g., IIR filters), have been explored. In particular, three hybrid reverberation algorithms [26] [27] [28] have presented in more detail, together with a procedure for the automatic tuning of the hybrid reverberator parameters taking into account a real impulse response to be emulated. Typically based on mixed FIR/IIR structures, these approaches aim to reduce the computational cost required to perform a convolution operation without lowering the perceived audio quality. Several tests have been carried out in order to prove the effectiveness of the presented approaches in terms of subjective and objective comparisons. Figure 2 show the energy decay relief (EDR) relative to a medium and a large room artificial IRs reproduced through the hybrid reverberator structures and the autotuning procedure presented in [27] and [28]. The effectiveness of the presented approaches is proved since the artificial reverberation shows the same harmonic content and the same T_{60} of the real room under test.



(i) (ii) (iii)

Figure 2. Medium room EDR. Real IR (i), Freeverb [28] IR (ii), Jot [27] IR (iii).



(i) (ii) (iii)

Figure 3. Large room EDR. Real IR (i), Freeverb [28] IR (ii), Jot [27] IR (iii).

Moreover, in order to highlight the improvement in terms of computational cost reduction obtained using hybrid reverberator structures with respect to fast convolution algorithms, Table 1 reports the workloads required to artificially reproduce both environments (i.e., medium and large rooms) on the DSP platform OMAP-L137.

Table 1. Workload required to reproduce the artificial effect of medium and large room using hybrid reverberator structures [27] [28] and fast convolution algorithm [25].

		Workload [%]		
Impulse response	T_{60} [s]	Freeverb [27]	Jot [28]	NUPOLS [25]
Medium	1	21	26	30
Large	1.8	19	24	33

3.2 Nonlinear device emulation

Several innovative contributions have been proposed in the field of nonlinear device emulation aiming to reproduce the acoustic behaviour of real nonlinear sound effects exploiting a black block approach (i.e., nothing about the internal structure of the nonlinear system is known).
More in detail, focusing on the dynamic convolution operation [29] [33], a new technique useful to approximate the convolution operation exploiting the principal component analysis (PCA) has been presented in [34] [35]. Decreasing the dimension of the multidimensional IRs dataset and retaining as much as possible the variation of the original dataset, this procedure allows one to improve the dynamic convolution approach reducing the workload required to perform the operation without lowering the perceived audio quality.
The proposed approach [35] results divided in three main stages: identification of the nonlinear system under test; preprocessing of the obtained data using the PCA in order to reduce the system complexity; emulation of the nonlinear system using static waveshapers and FIR filters pairs.

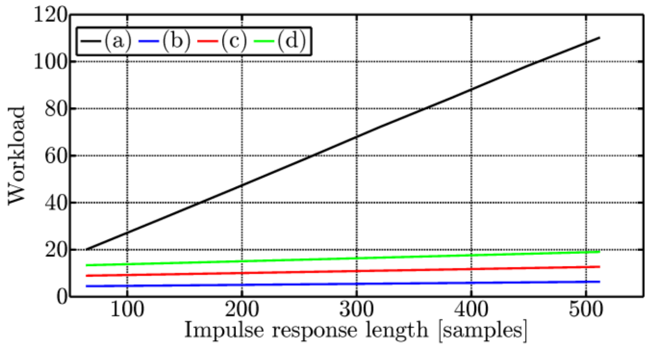


Figure 4. Analysis of the computational complexity in terms of workload as a function of the IR length. (a) Dynamic convolution, (b) proposed approach using 1 principal component, (c) proposed approach using 2 principal components, (d) proposed approach using 3 principal components.

Figure 4 depicts the workload required to perform the dynamic convolution and the PCA based approach [35] using various principal components as a function of the impulse response length on the DSP platform OMAP-L137.
Moreover, an adaptive identification procedure based on orthogonal nonlinear functions has been proposed in [36].

3.3 Audio Morphing

In order to provide a greater flexibility and a major artistic expression to musicians, an automatic audio morphing procedure for hybrid percussive sound generation has been presented in [41] aiming to combine two or more percussive samples to create new perceptually interesting sounds with intermediate timbre and duration. As depicted in Figure 5, the proposed approach results divided in two main stages: preprocessing of the audio references performed in the frequency domain and time domain linear interpolation to execute the morphing.

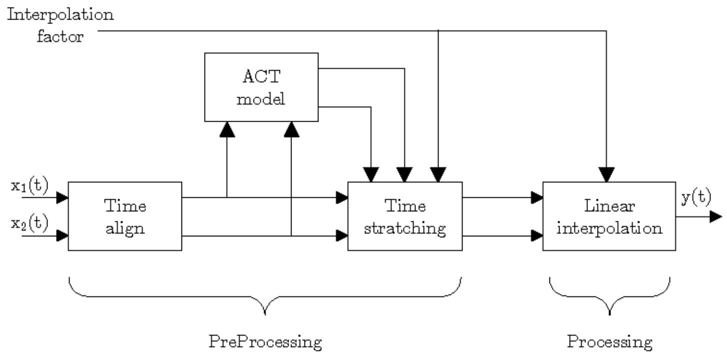


Figure 5. Block diagram of the proposed morphing algorithm [41].

Several tests have been carried out to evaluate the effectiveness of the proposed approach [41] through subjective and objective comparisons employing multiple real percussive samples extracted from the BFD2 database [53].

3.4 Multipoint room response equalization

Finally, to better emphasize the sound produced by musical keyboard instruments, the room equalization problem has been taken into account. Aiming to improve the objective and subjective quality of sound reproduction systems by compensating the room transfer function, the most common equalization approaches have been analyzed focusing on multipoint equalization. On this basis, three room response equalizers are presented taking into account minimum-phase and mixed-phase solutions. A mixed-phase multiple position room equalizer is proposed in [51]. The innovation lies on the use of an all-pass FIR phase equalizer, designed in the time domain which takes into account a suitable time reversed version of a prototype function and additionally performs a mixing time evaluation. On the other hand, the presented minimum-phase multiple position equalizers are based on an adaptive estimation of the impulse responses [49] [50]. Based on similar procedures to iteratively estimate the impulse responses and to generate the equalization filters, the approach presented in [49] has been extended in [50] by estimating the equalizer in warped domain in order to improve the equalization performance at low frequencies.

4 Conclusions

Research activities focused on the signal processing techniques used to increase the audio quality in musical keyboard instruments. In greater detail, taking into account the most common audio effects employed in electronic musical instruments, four different issues have been analyzed in this dissertation: artificial reverberation, analysis and emulation of nonlinear devices, audio morphing, and room response equalization.

On this basis, in order to enhance the audio quality of artificial reverberators, fast convolution techniques and hybrid reverberator structures have been explored. For what concerns fast convolution approaches, a DSP and PC based implementations have been developed taking into account a non uniform partitioning of the impulse response. While, a psychoacoustic technique has been proposed aiming to reduce the computational cost required to perform the convolution operation considering the human ear sensitivity. Moreover, several techniques for the approximation of the convolution operation exploiting recursive structure have been presented.

Taking into account the nonlinear system emulation problem, a technique for the approximation of the dynamic convolution operation by exploiting the principal component analysis and adaptive identification procedure based on nonlinear orthogonal functions have been proposed.
In order to provide a major artistic expression to musician, an algorithm for hybrid percussive sound generation has been presented.
Finally, several approaches for multipoint room response equalization have been developed taking into account mixed-phase and minimum-phase solutions.
Future works will be focused on the refinement of the nonlinear system emulation procedure and to the extension of the adaptive multipoint room response equalizer considering more than one loudspeaker.

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