MODELING OF NON-LINEAR REVERB EFFECT USING BLACK-BOX APPROACH

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Abstract: In this paper we propose a method for modeling hardware reverb effects that manifests non-linear character. The method is based on an identification of device under a test using input-output measurements utilizing a synchronized swept-sine signal. The model is implemented on a personal computer and on digital signal processor SHARC SC589. Both implementations are compared on the basis of time complexity.

Keywords: Generalized Hammerstein model, Synchronized sweep-sine, System identification, Non-linear model

1 INTRODUCTION

Reverbators are one of the most common effects used in an audio engineering field. They are used for a simulation of wave propagation in reverberation environments. Characteristics of these environments are based either on a measurement of real world places or on an artificial creation using miscellaneous digital signal processing algorithms [3]. For a realization, usually a software implementation on digital signal processors (DSP) or in form of audio plug-ins are used. Besides software reverbs, hardware reverbs based on discrete and analogue circuits have been created during last decades. The advantage of these reverbs is a unique artificial sound, nevertheless the drawback is a rarity of these effects which results in high prizes of these devices. Furthermore, due to special circuits, the devices exhibits a lot of non-linerites which contribute to a uniquenes of the sound of these reverbs.

![Figure 1: Block diagram of Generalized Hammerstein model](image-url)

Recently, there has been a tremendous effort to model these reverbs in form of software audio plugins. One can obtain a software algorithm using one of approaches: white-box modeling, which is reliant on a precise understanding of the inner structure, or black-box modeling based on non-linear
models such as Volterra model [4] or extended Kalman filtering [5] etc. In this paper, a nonparametric
generalized polynomial Hammerstein model (Fig. 1) will be considered which consists of $N$ branches
containing powered input signal $x(t)$ filtered by linear filters $g_n(t)$, so called kernels. The $N$ expresses
an order of the non-linearity. Several approaches exist for estimation of the kernels $g_n(t)$. One of
them is identification using synchronized exponential swept-sine signal [1]. The main disadvantages
of modeling reverbs using this model are computation complexity of the $N$ convolution and memory
demand for storing $N$ kernels $g_n(t)$.

2 SYNCHRONIZED SWEPT-SINE METHOD

The method is based on a non-linear convolution [2] extended by [1]. The non-linear device under a
test (DUT) is exposed to a synchronized exponential sweep-sine signal $s(t)$ defined as

$$s(t) = \sin \left\{ 2\pi f_1 L \left[ \exp \left( \frac{t}{L} \right) - 1 \right] \right\},$$

(1)

$$L = \frac{T}{\log \left( \frac{f_2}{f_1} \right)},$$

(2)

where $f_1$ is the initial frequency and $f_2$ the stop frequency, $T$ expresses length of the signal.

Processing the convolution of output signal $y(t)$ with an inversed input signal $\tilde{s}(t)$, a non-linear im-
pulse response (NLIR) $h_a(t)$ is obtained, defined as

$$h_a(t) = y(t) \ast \tilde{s}(t) = \sum_{m=1}^{\infty} h_m(t + \Delta t_m).$$

(3)

Besides the main impulse response $h_1(t)$ there are impulse responses (IR) of nonlinear components
$h_m(t)$, as illustrated in Fig. 2. The distance between kernels is defined as

$$\Delta t_m = L \log(m).$$

(4)

Figure 2: Output of nonlinear convolution $h_a(t)$.

Each IR can be separated using a weighting window $w(t)$. Referring to [1], one can obtain kernels
$g_n(t)$ for the Generalized Hammerstein model using a transformation defined as
\( \mathbf{G}_n(f) = (\mathbf{A}^T)^{-1} \mathbf{H}_n(f), \)  
\( \mathbf{H}_n(f) \) denote vectors of transformations of kernels \( g_n(t) \) and impulse responses \( h_n(t) \) to frequency domain. \( (\mathbf{A}^T)^{-1} \) denotes inversion of transpose matrix \( \mathbf{A} \) which is defined as

\[
A_{n,m} = \begin{cases} 
\frac{(n-m)^2}{2^{n-m}} \left( \frac{n}{2^n} \right), & \text{for } n \geq m \text{ and } (n+m) \text{ is even} \\
0, & \text{else.}
\end{cases}
\]

3 MEASUREMENT

The problem of modeled non-linear reverbs is a reverberation time which influences the length of impulse response itself. Subsequently, it must be ensured that the tales of impulse responses of each kernel will not overlap. With some configuration of reverbs, an envelope of IR can fall under a noise level in 15 s. In order to have a sufficient distance between kernels \( \Delta t_m, \ T \) should be chosen as long as possible. To control the reverberation time, an appropriate weighting window is used, which will decrease an influence of late reverbs, such as a Gaussian window.

For the modeling, a rare hardware reverb exhibiting distortions was chosen. The reverb was measured using stimulus with an amplitude \(-1 \text{dBFS}\) sweeping in 240 s from 20 Hz to 20000 Hz at sample rate 44.1 kHz. Obtained kernels are depicted in Fig.3. For the IR separation Gaussian window with length of 8 s was chosen.

![Figure 3: Estimated time responses of kernels.](image)
4 SIMULATION

The gathered kernels were used for Generalized Hammerstein model which was implemented on PC with configuration Intel i5, 8GB RAM and on digital signal processor SHARC SC589 from Analog Devices. Since the most time demanding procedure is the convolution operation which should be done without small input-output delay, an efficient algorithm called Frequency-domain Delay Line (FDL) was used [6]. The implementation of the algorithms is in ANSI C with maximal optimization level. The input-output delay is controlled by a length of block (BL). The delay for BL=32 is 0.7256 ms, for BL=64 is 1.4512 ms and for BL=128 2.9025 ms.

The time complexity was measured for an order of non-linearity $N=5$, with different BLs and with different lengths of kernels that were controlled by the weighting window $w(t)$. Simulation was done using offline processing on input signal $x(t)$ of duration 72.56 ms (3200 samples). Obtained results are in Fig.4. Furthermore, on the SHARC SC589 additional optimization using FFT accelerators can be utilized which leads to a decrease of duration of the computation, as in can be seen on dashed curves in Fig.4 (b).

![Figure 4](image-url)  
**Figure 4:** Time complexities of simulation of nonlinear reverb using order of non-linearities $N=5$, different lengths of kernels $g_n(t)$ and different block sizes for FDL convolution. Implementation on PC (a) and on SHARC SC589 (b).

From the figures in can be seen that the time complexity strongly depends on the block length and on the lengths of kernels, as it could be expected. After the utilization of FFT accelerator it can be observed the duration decreased by four time. Since a human ear cannot perceive small latencies such as 3 ms, the BL=128 can be used.

Nevertheless, both implementation are unfortunately insufficient for real-time realization, since non of them can achieve a critical time for the processing of 3200 samples. Therefore any further optimizations have to be used. For instance, the reverbs does not operate in full frequency range, hence the processing can be done in lowered sampling domain. Additionally, the processing effort can be separated to different cores of processors, etc.
5 CONCLUSION

Simulation of nonlinear system reverb effects using Generalized Hammerstein model is very efficient approach. Nevertheless, due to the time demanding computation of convolutions with kernels that should cover the reverberation time of the reverbs, the method is hardly usable in a real-time processing without any additional optimization such as vector, multi-core processing, calculation with lowered sampling rate etc.

REFERENCES


