APPROXIMATING MEASURED REVERBERATION USING A HYBRID FIXED/SWITCHED CONVOLUTION STRUCTURE

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ABSTRACT

An efficient reverberator structure is proposed for approximating measured reverberation. A fixed convolution matching the early portion of a measured impulse response is crossfaded with a switched convolution reverberator drawing its switched convolution section from the late-field of the measured impulse response. In this way, the early portion of the measured impulse response is precisely reproduced, and the late-field equalization and decay rates efficiently approximated. To use segments of the measured impulse response, the switched convolution structure is modified to include a normalization filter to account for the decay of the late-field between the nominal fixed/switched crossfade time and the time of the selected segment. Further, the measured impulse response late-field is extended below its noise floor in anticipation of the normalization. This structure provides psychoacoustically accurate synthesis of the measured impulse response using less than half a second of convolution, irrespective of the length of the measured impulse response. In addition, the structure provides direct control over the equalization and late-field frequency dependent decay rate. Emulations of EMT 140 plate reverberator and marble lobby impulse responses are presented.

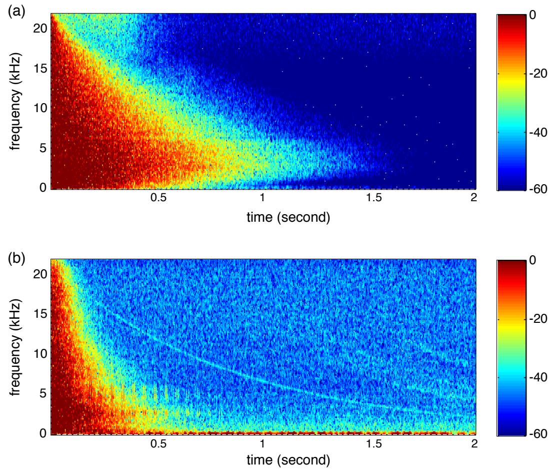

Figure 1: *Impulse Response Spectrograms; (a) EMT140 plate reverberator (b) CCRMA Lobby, Stanford University*

1. INTRODUCTION

Room reverberation is often simulated using convolution with a measured impulse response (IR) [1, 2] or with delay line network [3]. Example spectrograms of two measured IRs are shown in Figure 1. Convolution reverberation methods provide a straightforward and accurate approach, but typically lack parametric control and require significant computation and memory resources. Artificial reverberators are often implemented in the form of all-pass filters, comb filters, and feedback delay networks (FDN) [3], all of which attempt to provide a sufficient echo density and provide control of late-field decay rates. In particular, the FDN offers a high quality, efficient alternative to convolution with parametric control suited for real-time emulation, but lacks control over the important early response of measured reverberation.

Hybrid artificial reverberation structures attempt a compromise, and consist of a short convolution unit in parallel with a FDN [4, 5]. The direct convolution unit accurately models the psychoacoustically significant features of the early response, while the FDN efficiently models the late-field with parametric control. It is difficult to dovetail the convolution output with a FDN simulated late-field, motivating alternative methods [4]. One drawback of the method [3, 4] is that it is difficult for the FDN to provide a good match to the temporal and equalization characteristics of the measured response.

In this work, a switched convolution (SC) reverberator is used in a hybrid structure to efficiently and accurately approximate a measured reverberation response. The hybrid structure presented here consists of a short convolution unit in parallel with a SC reverberator, drawing its response segments from the measured IR. Doing so provides a good quality match to the desired characteristics of a given measured reverberant response.

The SC structure presented here is that presented in [6], with segments of the measured late-field reverberation used in place of the required noise sequence generation. Because the measured reverberation segments retain the desired characteristics of the measured response, little effort is required to accurately tune the SC structure. In addition, its computational cost is reduced by eliminating the need for random noise generation.

To smoothly fade between the parallel sections, a power complementary crossfade, developed in [5] is used. Additionally, when

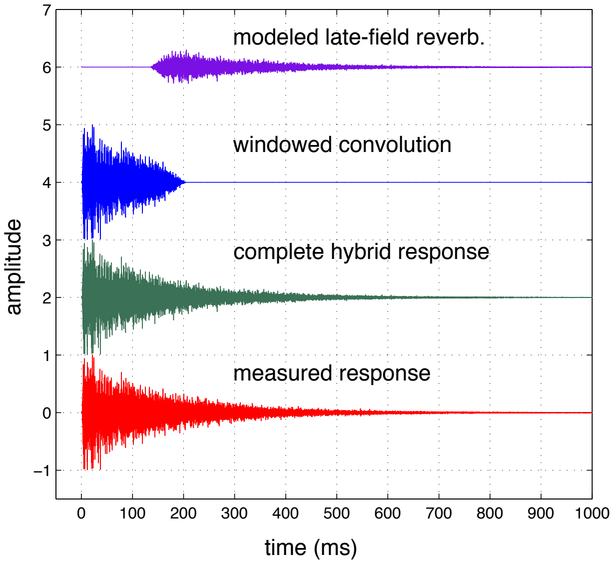

Figure 2: *Hybrid Structure Impulse Response for EMT 140 plate reverberator.*

using segments of a measured IR, a normalization is needed to account for the decay between the crossfade time and the segment positions in the measured IR. In doing so, careful consideration must be given to the frequency dependent noise floor of the measured IR.

In §2, we review the general characteristics of reverberant IRs in conjunction with the modified hybrid reverberator structure. The data-driven SC reverberator described in §3. Results and conclusions are found in §4.

2. HYBRID SWITCHED/FIXED CONVOLUTION

The hybrid structure of [4] gives a computationally efficient model in which a short convolution unit modeling the early response runs in parallel with an FDN structure for late-field reverberation. Abel, et al., demonstrated a digital emulation of the Elektromesstechnik (EMT) 140 plate reverberator using such a model [5, 7], and presented an improved crossfading method. The system response $r(t)$ is

$$r(t) = e(t) + p(t) \ , \qquad (1)$$

where $e(t)$ is the impulse response of the early portion of the IR and $p(t)$ is that of the late field. The early convolution $e(t)$ is taken directly from the measured response. The late-field $p(t)$ is assumed to have a frequency-dependent exponential decay defined by an amplitude envelope maintaining

$$l(t;\omega) = q(\omega) \cdot e^{-t/d(\omega)} \ , \qquad (2)$$

where $q(\omega)$ is an equalization, and $d(\omega)$ is a frequency-dependent decay rate. $r(t)$, $c(t)$, and $p(t)$, along with a measured response are depicted in Figure 2.

A power complementary sine window crossfade is applied to the two sections. In this way, the fade-out of the convolution unit smoothly dovetails with the fade-in of the SC reverberator output as shown in Figure 3. The cross-fade time is selected as the start

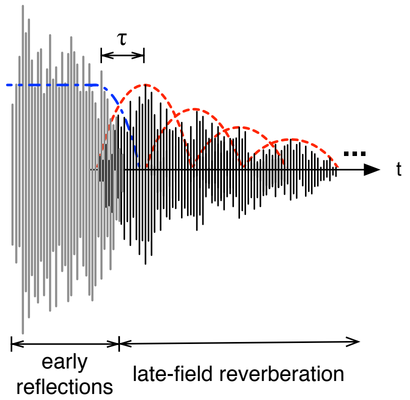

Figure 3: *Cross-fade between impulse responses of early reflections and late-field reverberation.*

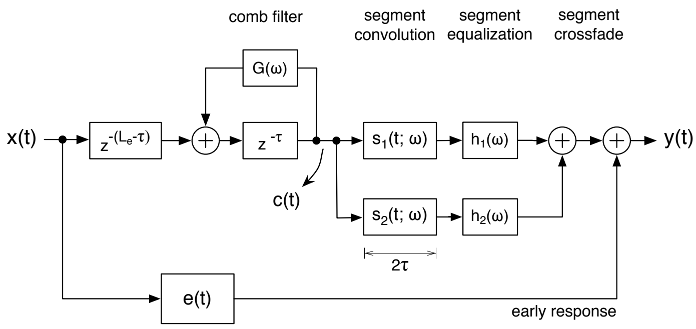

Figure 4: *The Hybrid Switched Convolution Reverberator; L_e is the length of impulse response $e(t)$, and $G(\omega)$ is a absorbent filter having the same decay times $d(\omega)$ as $h_n(\omega)$.*

of the late-field, determined according to the normalized echo density [8]. Finally, the hybrid structure uses measured IR segments for both the early and late response. The segments are assumed to be statistically independent, and a simple constant-power crossfade is sufficient to bridge the fixed and switched convolutions. Figure 4 shows the hybrid fixed/switched convolution reverberator signal flow architecture. The late-field reverberation path has a delay to align with the early reflections, as illustrated in Figure 3.

3. DESIGN CONSIDERATIONS

The switched convolution reverberator consists of a recursive comb filter having delay τ, driving a convolution with a short noise sequence. The reverberator equalization and frequency-dependent decay rate are controlled by low-order IIR filters, while the echo density is controlled via the noise sequence. When the input $x(t)$ is an impulse, the feedback gain and equalization are pure gains, resulting in an exponentially decaying pulse train output $c(t)$. The reverberator output $y(t)$ is this pulse train convolved with the equalized late-field segments. Therefore the system response to an impulse is the measured early response followed by an exponentially decaying noise sequence. Note that the frequency-dependent decay rate is controlled by a single filter $G(\omega)$. The measured decay rate may therefore be reproduced by choice of $G(\omega)$ in view of the comb filter period τ, as shown in Figures 4 and 5.

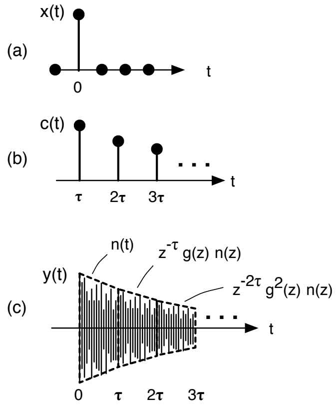

Figure 5: *Switched Convolution Reverberator Impulse Response; (a) Input Impulse, $x(t)$ (b) Comb Filter Output, $c(t)$ (c) Reverberator Output, $y(t)$*

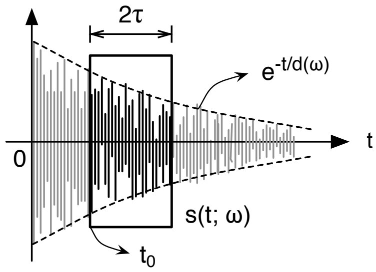

Figure 6: *A Segmented Sequence $s(t;\omega)$ in a Measured Late-field Impulse Response.*

While this structure is simple and readily generates high echo densities, if the noise sequence is unchanging the output has an unwanted periodicity at the comb filter delay τ. To overcome this difficulty, the noise sequence is replaced or *switched* with a new, statistically independent sequence every comb filter delay τ. Several techniques are described in [6], and here two measured IR segments of length 2τ with 50% overlap are used to crossfade the switched sequences at period τ.

To adapt the SC reverberator to emulate a measured IR, noise sequences taken directly from the measured late response are used. By using segments of an actual measurement, the characteristics of the measured impulse response are reproduced. Each segment is chosen from a random position in the measured IR, and time-normalized with respect to the start of the late-field decay via a filter $h(\omega)$, compensating for the late-field decay. The segments are statistically uncorrelated with each other as they are assumed to be colored Gaussian noise sequences with short correlation distances. A noise segment from time t_0 maintains a magnitude as a function of t and ω given by

$$s(t;\omega) = q(\omega) \cdot e^{-(t+t_0)/d(\omega)},\ \ 0 < t < 2\tau\ . \tag{3}$$

The equalization filter $h(\omega)$ is then

$$h(\omega) = e^{t_0/d(\omega)}\ , \tag{4}$$

where the late-field is assumed to start at $t = 0$.

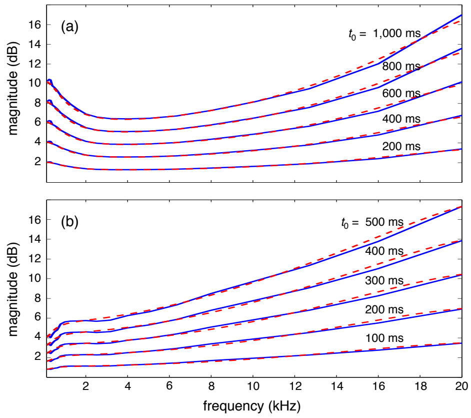

Figure 7: *Normalization Filters; (a) EMT 140 plate reverberator and (b) CCRMA Lobby. Solid lines are calculated filter responses using measured delay times and dashed lines are designed filter responses.*

To design the normalization filter $h(\omega)$, the decay time $d(\omega)$ is measured from the envelope of band-passed impulse responses based on one-third octave bands. For the impulse responses tried, a cascade of two first-order shelving filters for low and high frequency bands and a second-order peaking filter for middle frequency band was sufficient to accurately model the required characteristic. Figure 7 shows the reference equalization according to the measured decay times and the designed characteristic using the cascaded filters. Note that the designed filter provides a good match to the reference filter calculated using the measured decay times.

Since the normalization filter $h(\omega)$ is a function of the randomly chosen segment starting time t_0, $h(\omega)$ is updated with each new segment. The shelf and parametric filter cascade described above has the advantage of being parametrized, and can be easily redesigned as new impulse response segments are switched in. This filter design, however, may be avoided by using a limited set of predetermined equalization filters designed for predetermined t_0's, and randomly selected. The number of normalization filters is chosen so that the response of the reverberator has no periodic components.

In applying $h(\omega)$ according to [6], the measured impulse response is assumed to decay according to $d(\omega)$ for all time t. Measured IRs, however, turn out to follow (2) only until a frequency-dependent noise floor is reached causing the filter $h(\omega)$ to overcompensate for the decay in the presence of a noise floor. To overcome this difficulty, the noise floor can be removed by splitting the measured response into frequency bands and detecting the frequency-dependent noise floor in each band. The impulse response bands are faded out and a synthetically extended late-field is faded in. The spectrograms of an original EMT IR and synthetically extended IR are shown in Figure 8.

4. RESULTS AND CONCLUSIONS

Figure 2 shows an example of the hybrid structure impulse response for the EMT 140 plate reverberator. With a 70 ms-long comb filter delay τ in the switched convolution reverberator, the hybrid reverberator has the crossfade region of the same length as

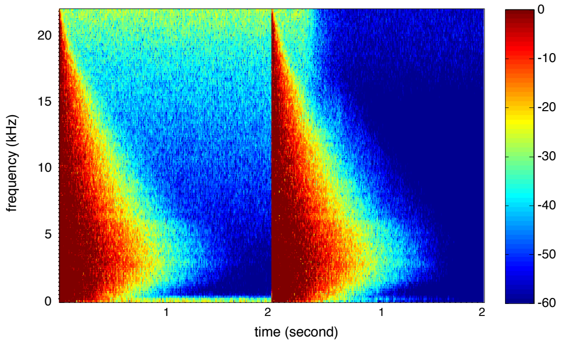

Figure 8: *Spectrograms of original (left) and synthetically extended (right) impulse responses for EMT 140 plate reverberator.*

the delay, and the modeled IR nicely matches the measured response. Figure 9 shows the frequency characteristics of the modeled impulse responses are very similar to those of measured data for an EMT reverberator and marble lobby. Informal listening tests show the hybrid reverberator structure produces a natural response with perceptually similar to the measurement.

To summarize, a hybrid reverberator structure using a switched convolution reverberator having low memory requirements and small computational complexity was presented. By using segments of a measured IR, the late-field reverberator readily produces a good match to the measured response. The transition between the early part of the response modeled by the convolution and late-field reverberation is accomplished by a power complementary crossfade. The hybrid structure presented is more efficient than the FDN in terms of memory and computational costs. Experiment results show that the proposed reverberator can faithfully reproduce details of a measured impulse response.

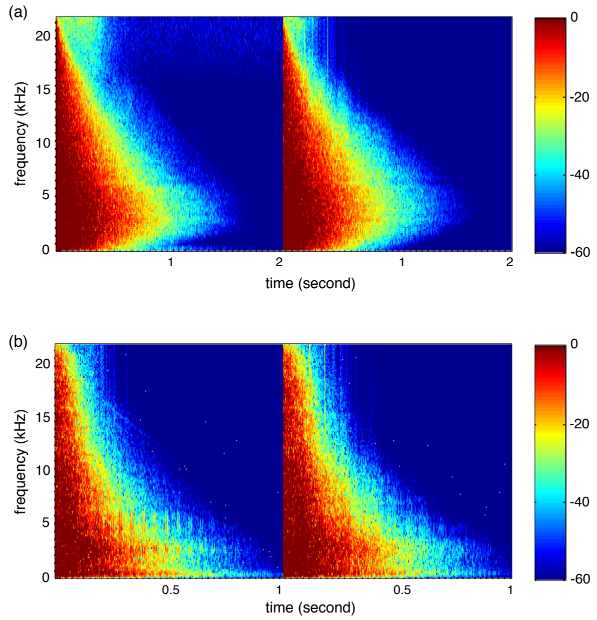

Figure 9: *Spectrograms of measured (left) and modeled (right) impulse responses; (a) EMT 140 plate reverberator (b) CCRMA Lobby.*

5. ACKNOWLEDGEMENTS

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