

Reverb Design

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Abstract

Reverb Design is about creating an aesthetic appealing spacial impression aimed at sweetening the listening experience for a given context.

However, the ambiguity of auditory and visual cues in reproduced sound makes it necessary to create an auditory illusion [6]. To make this illusion work, an aesthetic concept of simplicity and intelligibility is proposed based on psychoacoustic facts and recording practice.

A development kit for PD is presented to encourage reverb design from a "birds view", with complete reverberation algorithms as basic building blocks. An example topology completes previous work [19] [20] by diffuse reflections and late reverberation.

Keywords

Reverb, Design, Psychoacoustic

1 Introduction

A concert hall is a time-variant non-linear system. When directional instruments are moving on stage playing different pitches, the reflection pattern is almost changing randomly [16] including doppler shifts.

Single reflections are not perceived as echoes in a concert hall, but contribute to loudness, creating an auditory event under the umbrella term spacial impression (SI).

Our neural system seems to build expectations about reflections likely to occur, inhibiting echo perception. This so called 'precedence effect' is not yet fully understood. The effect seems to be direction or even scenario specific and builds up while adapting to the environment. [9] [12]

Reflection patterns may be remembered and are used i.e. to enhance localization [22]. Trained Listeners were able to perceive the directivity of music instruments in a complete sound field [20] [17].

Binaural hearing enables highlighting sound from a favoured source and direction guided by attention - so called cocktailparty effect [2] - and to

suppress reverberation [5].

Our perception is multimodal. Visual cues support the cocktailparty effect. Visual cues may be perceived as perceptually salient and may be intermingled with auditory cues [13].

Barron suggests that visual cues perceptually compensate for decreasing sound level in concert halls when source-receiver-distance increases [3].

In sum, these effects lead to heightened intelligibility inhibiting early and diffuse reflections, while maintaining a strong sense of SI at the same time.

2 Aesthetic Concepts

Reproducing the reverberation of a concert hall with loudspeakers makes is hard for our neural system to achieve a comparable intelligibility. Some cues are weakened (i.e. direction and frequency response of ER) or absent (i.e. visual cues). Further, extra tasks have to be performed like summing location [26], which may conflict in part with the task of echo suppression.

To compensate for these perceptual differences, specific aesthetic concepts are needed.

2.1 Intelligibility

To help intelligibility the sound field may be simplified: if lesser ER come into the way of the onset of notes clarity improves. Fortunately, the amount of SI has found to be independent on the number of early reflections (ER), but on total reflected energy [4]. Thus an IR can be made sparse within the limits of echo perception with a high level of unique reflections representing salient geometric cues.

SI may be increased by introducing small incoherences in spectra, timbre and delay time of reflections [12], i.e. by modeling sound source directivity and movement.

Spacial impression increases with total sound level [5]. Assuming a reduced dynamic in reproduced sound, longer reverb tails and stronger SI are needed to achieve a subjective response comparable to a remembered concert. The non-linear decay of nested allpass structures as described by

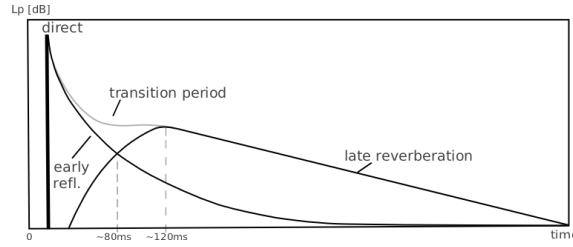


Figure 1: Transition period between early and late part of impulse response

Schroeder[21] and later explored by Gardner[10] provide these longer tails without having to increase total reflected energy - avoiding muddying the sound by simply turning up the level of an exponential decaying reverb.

Traditionally intelligibility is improved through trading a high level of direct sound (which is a close distance cue) for large distance cues in reverberation (i.e. high level and quick onset of late reverberation). This leads to an auditory contradiction, i.e. being close to the sound source and far away at the same time - often perceived as "spectacular" or "exiting". Missing this excitement may be perceived as "boring".

2.2 Texture and Coloration

ER texture changes in a characteristic way with source-receiver-distance in a concert hall. When distance to the stage increases, reflections from the sidewalls, ceiling and rear wall are moving closer to the direct sound, increasing in (relative) level. Diffuse reflections begin to dominate the physical sound field. Perceived loudness stays constant, however.

In addition to distance perception, modeling texture may be an aesthetic dimension of its own, shaping the fine structure of a reverb in a comprehensive way to our auditory system: as every reflection has its own frequency response the colour of the entire reverb changes with distance along with direct-to-reflection-ratio.

Experiments show that intelligibility and SI in post production can be enhanced if coloration, diffusivity and early reflection texture are matched to those found in the recorded hall[27]. It is therefore important to control these parameters in a natural way.

3 Design Concepts

3.1 Multistream Design

A basic principle would be to divide the impulse response (IR) into three parts: a statistical, 'colorless' late part and a geometrical, 'colorful' early part. In a transition period both aspects are blurred (see fig.1). Each of the three parts may have its own specialized algorithm.

The transition time (approx. 80...500ms) is of special interest because it determines the colour¹ and subjective feeling of envelopment[28] of a concert hall or an artificial reverb. It would be a good task for convolution to provide this part of the IR. A cheaper solution would be to use a simple Schroeder model such as SATREV or JCREV² and feed it with an extended ER pattern, as shown in chap.4.

The more channels are used for reproduction the more important it becomes to provide realistic early reflections. These may not be static: frequency response and delaytime may change. This is what happens in real halls, too. If the pattern is static, build up of the precedence effect may be so strong that hardly any SI is perceived after a short time of adaption to the environment, as earlier work has shown[19].

Separating the early and mid part - which is often referred to as a 'fingerprint' of a room by acoustics - from late reverberation results in increased flexibility: large combfilters are allowed (resulting in slow buildup of echo density); geometrical properties derived from a room model may be used for the spacing of delays, as proposed by Moorer[14], Jot[11] and Rocchesso[18]. As a rule of thumb a given geometric relation may be used throughout the entire topology. The dimensions of Boston Symphony Hall, Amsterdam Concertgebouw and Vienna Musikverein have proven to be particularly useful (see Appendix B).

¹Blauert mentions a characteristic "blur" of early and late reflections[5]

²Planet CCRMA, John Chowning, implementation in FAUST by J.O.Smith, see Appendix A

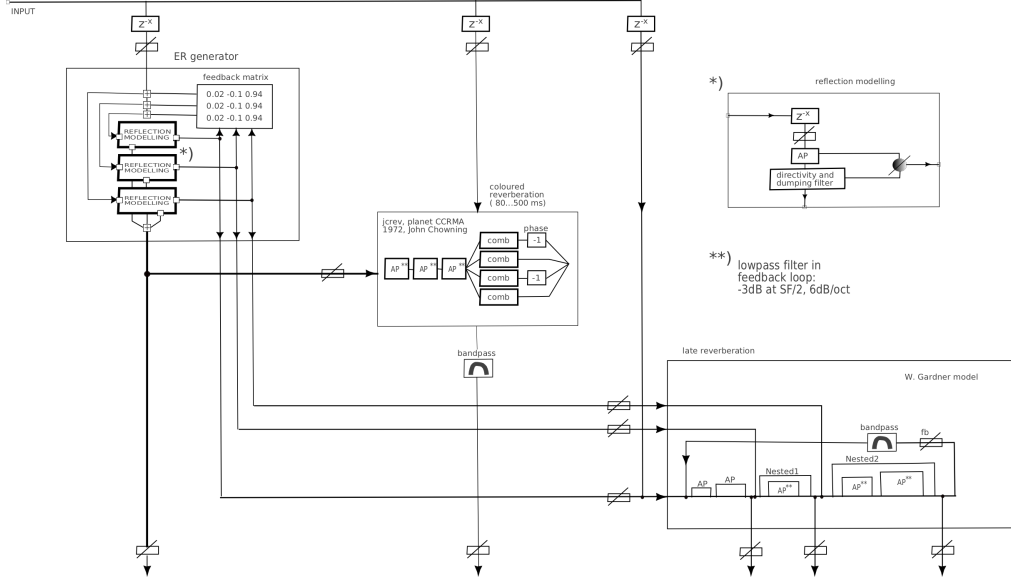


Figure 2: Example Design with separate algorithms for early, mid and late reverberation

3.2 Slope

Single reflections follow simple inverse square law. From listener's point of view the slope changes with distance d to the sound source. Normalizing the direct sound to 0dB, the level Lp of a reflection is given by:

$$Lp = 20 * \log \frac{d}{\Delta tc + d} \quad (1)$$

where Δt is the arrival time of the reflection after the direct sound.

The first echo of comb filters can be matched to this slope:

$$g = \frac{\Delta tc + d}{\Delta tc + mTc + d} \quad (2)$$

where g is the attenuation gain, Δt is the position of the comb filter in a tapped delay line after the direct sound, m is the size of the comb filter in samples, T is the sampling period and d is the distance from the direct sound source.

As a result, feedback values may be scaled making them a function of comb filter size and the distance cue.³

3.3 Modulation

Modulation can be carried out by randomization or animation. It is a key feature distinguishing recursive algorithms from i.e. most convolution based reverbs.

³allpass filters work similar. For theory on allpass filters see i.e.[10].

Randomization provides a means to overcome the lack of modal density generally found in recursive algorithms [8] by smoothly changing comb filter sizes. The resulting pitch shifting artifacts reduce metallic ringing and add "warmth" to the reverb tail.

A more expensive way to implement time variance is the use of non-transposing delay lines (see Appendix A).

Animation simulates doppler effects caused by a moving sound source, but may be disturbing for sound sources which commonly do not move, i.e. piano or organ.

3.4 Nested vs. Parallel Structures

Nested allpass structures have been mentioned early in Schroeder's work[21]. Nested allpasses show a hightened increase in echo density compared to the quadratic increase predicted by the source image model[1] which does not take into account sound scattering surfaces.

In practice, parallel nested structures have interesting aesthetical properties: if different modulation schemes are used in combination with phase inversion, a "soft-phaser-like" effect results: random phase cancellations occur, which change over time. (When listening to a Lexicon 480L without dry signal, this effect may be audible). Also high and low frequency are attenuated.

FDNs may be used to create a similar effect. FDNs and digital waveguide networks [18]

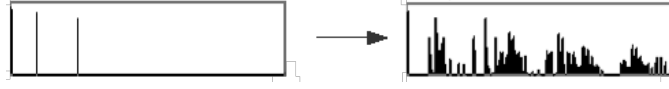


Figure 3: ER generator: Two calculated reflections extended by moderate feedback settings

have to be investigated further: probably finding new (modulated) feedback matrices and scattering junction functions (which differ from theoretical models) is necessary to satisfy psychoacoustic requirements.

3.5 Multichannel I/O

todo.

4 The Reverberation Development Kit

The use of the development kit in PD is straight forward: building blocks are abstractions or externals controlled by a powerful preset manager with built-in scaling capabilities. This makes it easy to change all or a subset of parameters at once, supporting the late design stage when hundreds of parameters are scaled and reduced to 5 or 6 GUI options.

The generated IR can be viewed, listened to and analyzed at any time thanks to PD's connectivity to jack⁴.

The collection of algorithms include complete solutions for the generation of ER and late reverberation, Schroeder sections and nested AP filters. Diffusor parts are separated for modularity.

A detailed description of every included algorithm is beyond the scope of this paper, however, references to the original papers are given in Appendix A. Key features are summarized in chap. 4.1 and 4.2.

4.1 Key Features

In previous work [19] it has been found that ER may not have the same frequency response. Reflections are individually modelled using 6-band

equalization by the ER generator abstraction. A shoe box room is used to calculate source image models for moving sound sources including spherical coordinates. These are used to read from a large table of directivity data.

Reflection modeling blocks (see fig.2 to the right) contain an allpass filter, which serves two purposes: simulation of diffusivity (small AP sizes) and additional reflections (medium AP sizes with low gain values). The delay lines are extended by a feedback matrix. The resulting extension of the IR is both correlated to the directivity of the sound source, room geometry and damping of walls, supporting individual coloration of the early IR. Fig. 3 shows an example.

Two nested allpass designs are included as externals. Nested1 contains a delay-AP-delay chain embedded into the outer allpass and Nested2 contains a delay-AP-delay-AP-delay chain, as suggested by Gardner[10]. The feedback paths can be delayed by one sample to provide additional damping. Delays are fractional to allow for modulation.

4.2 Example Design

Fig 2 shows a design example for one channel using different algorithms for early, mid and late parts of the reverb. The important transition period is controlled by both the feedback settings of the ER generator and by choosing the amount of directivity filtered jref input.

Onset of the late reverberation is controlled by feeding small amounts of direct sound to mid and late reverberation and by choosing out tap locations accordingly.

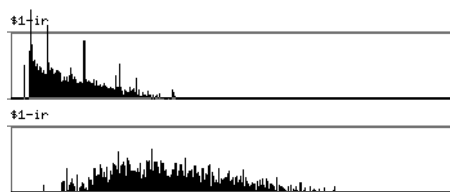


Figure 4: Example design: sparse ER pattern and onset control of late reverberation

⁴Jack Audio Connection Kit, see <http://sourceforge.net>

The decay of ER is determined by source-receiver-distance. Reverberation time is set by input level, feedback value and filtering of late reverberation. A bandpass filter with center frequency around 150Hz and $Q \sim 0.7$ achieves a more aggressive high frequency rolloff in the feedback loop than a lowpass filter would. Trading late reverberation level versus main feedback in the Gardner model controls late energy distribution.

5 Conclusion

Designing a reverb often required the examination of lower level algorithms including delay line lengths and feedback matrixes. However, the design of a complete reverberator requires the combination and control of these building blocks in a sophisticated way which may be a challenge of its own. This paper hopes to help closing this gap by providing the necessary externals and abstractions for PD, together with an introduction to basic design philosophies.

6 Acknowledgements

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Appendix A: List of Externals and Abstractions

External	Author	Description	Lang	Licence
directivity	U.Schlemmer	data set for 13 instruments	C	GPLv2
nested1a~		nested allpass no damping	FAUST ¹	GPLv2
nested1b~		inner AP damped	FAUST	GPLv2
nested1c~		inner and outer AP damped	FAUST	GPLv2
nested2a~		nested AP with two APs in inner loop	FAUST	GPLv2
nested2b~		inner APs damped	FAUST	GPLv2
nested2c~		inner and outer APs damped	FAUST	GPLv2
BPF~		band pass filter	FAUST	GPLv2
lfnoise~		low frequency random walk	FAUST	GPLv2
chorussclass~		random chorus	FAUST	GPLv2
sdelay~	GRAME ²	non-transposing delay	FAUST	LGPL
jcrev~	John Chowning ³	3-AP-4-comb Schroeder section	FAUST	STK-4.3
satrev~		3-AP-4-comb Schroeder section	FAUST	STK-4.3
fdn4~	Julius O. Smith ⁴	FDN development kit	FAUST	STK-4.3
freeverb~	Jezar at Dreampoint ⁵	8 comb 4 AP Schroeder section	FAUST	BSD
g2reverb~	Fons Adriaensen ⁶	8 AP diffusor 4 delay FDN reverb	C++	GPLv2
zita_rev1~	Fons Adriaensen ³	8 delay FDN reverb with APs in each delay line	FAUST	STK-4.3

Table 1: Externals

Abstraction	Author	Implementation		Licence
uref	U. Schlemmer	early reflection generator	PD	GPLv2
emt250		3-comb-3-AP reverb	PD	GpLv2
reverb	M. Puckette[25]	4 fdn butterfly (Hadamard matrix) prototype	PD	GpLv2
ga2	W. Gardner[10]	Gardner reverb with nested APs	PD	GPLv2
dat	J. Dattorro[7]	4 AP diffusor and "tank" structure	PD	GPLv2
ezpst		preset manager	PD	GPLv2

Table 2: Abstractions

¹[15], <http://faust.grame.fr/>,

²Centre National de Creation Musicale, Lyon, France. Source code in FAUST's music.lib

³[23], transcription for FAUST's effect.lib: J.O.Smith, for description and analysis, see https://ccrma.stanford.edu/~jos/Reverb/Reverb_4up.pdf,

⁴[24], https://ccrma.stanford.edu/~jos/pasp/FDN_Reverberators_Faust.html,

⁵[23], transcription for FAUST's examples: GRAME, for description and analysis, see https://ccrma.stanford.edu/~jos/Reverb/Reverb_4up.pdf,

⁶Source code, see <http://rev-plugins.sourcearchive.com/lines/0.3.1/>,

Appendix B: Selected Proportions

concert hall	width[m]	lenth[m]	hight[m]
Concertgebouw Amsterdam	28	44	17
scaled to 15000 m3	25.48	40.04	15.47
Boston Symphony Hall	75	125	61
scaled to 15000 m3	22.575	37.625	18.361
Musikverein Vienna	19.1	48.8	17.75
scaled to 15000 m3	18.718	47.824	17.395
next prime numbers	18719	47819	17393
Brahms Saal MV Vienna	10.3	32.5	11
scaled to 15000 m3	16.6654	52.585	17.798
Fibonacci	$0.5 \cdot 34$	55	21
scaled to 15000 m3	15.776	51.04	19.488

Table 3: proportions of selected concert halls