# 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 "bird's eye view", with complete reverberation algorithms as basic building blocks. An example topology complements previous work [23] [24] with 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[19] 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] [14]

Reflection patterns may be remembered and are used i.e. to enhance localization [27] . Trained Listeners were able to perceive the directivity of music instruments in a complete sound field [24] [20].

Binaural hearing enables highlightening sound from a favoured source and direction guided by attention - so called cocktailparty effect [2] - and to suppress reverberation [5].

Our perception is multimodal[31] and visual cues support the cocktailparty effect[13]. Visual cues may be perceived as extraordinary salient and as such be intermingled with auditory cues[16][11].

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 hightened 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 early reflections) or absent (i.e. visual cues). Further, extra tasks have to be performed like summing location [32], 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 reflections interfere with played notes clairity improves. Less reflections does not necessarily mean a decrease in SI. SI may be increased by modulating spectra, timbre and delay time of reflections[15], i.e. by modeling sound source directivity and movement. Moreover, the amount of SI has found to be independent on the number of reflections, but on total reflected energy[4]. Thus an IR can be made sparse within the limits of echo perception: depending on signal properties[21] unique early and diffuse reflections representing salient geometric cues of a modeled room may be used.

This approach encourages the design of individual reverbs for different types of music or scenes or even playing styles.

Spacial impression increases with total sound

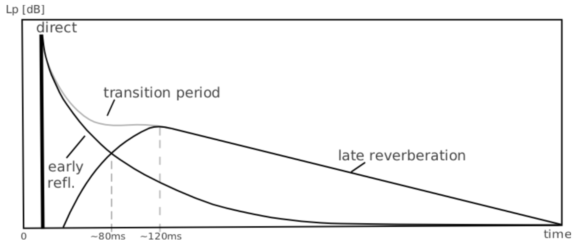


Figure 1: Transition period between early and late part of impulse response

level, which may be be explained by the "curves of equal loudness"[5]. Assuming a reduced dynamic range in reproduced sound, longer reverb tails and stronger SI may be needed to achieve a subjective response comparable to a remembered concert. The non-exponential decay i.e. of nested allpass structures as described by Schroeder[25] and later explored by Gardner[10] provide these longer tails without having to increase total reflected energy.

Traditionally intelligibility is improved by 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. beeing close to the sound source and far away at the same time - often perceived as "spectacular" or "exiting". This exitement may be missed if the physical soundfield found in a room is (exactly) reproduced at the place of the ears.

### 2.2 Texture and Coloration

Early reflection 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 will have 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[33]. 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 devide the impulse resonse (IR) into three parts: a statistical, 'colorless' late part and a geometrical, 'colorfull' 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 period is of special interrest because it determines the colour and subjective feeling of envelopment[34] of a concert hall or an artificial reverb.[1]

It would be a good task for convolution to provide this part of the IR. A cheaper solution would be to use a Schroeder model such as satrev or jcref[2] and feed it with an extended early reflections pattern, as shown in chap.4.

The more channels are used for reproduction the more important it becomes to provide realistic early reflections[3]. These may not be static: frequency response and delaytime may change. This

[1]

Blauert mentions a characterisic "blur" of early and late reflections[5]. The time when this transition is audible may be best described in terms of perceptual grouping[6]: if a sound is continuous, then there will be no maximum delay time beyond which a reflection will not contribute to the perceived spatial impression of the source [15]. Reflections tend to be *fused* with the direct sound. Only if a sound is short or when pitch or timbre changes or other localization cues are given [21] some of the reflections will be perceived to be a separate reverberant decay.

Schroeder analyzes the time after which echoes become a statistical clutter in dependence to signal width of an impulse in [26].

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

[3]Specially if listeners move around in the sound field the adaption to the reverberant environment seems to be re-evaluated[23]. This might be explained by the missmatch of auditory and visual cues[6].

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[23].

Separating the early and mid parts from late reverberation results in increased flexibility for the choice of delay line length in the late algorithm: Geometrical properties derived from a room model may be used for the spacing of delays, as proposed by Moorer[17], Jot[12] and Rocchesso[22]. These proportions (see Appendix B for examples) differ from commonly known spacings based on mutually prime numbers resulting in "gaps" of echo build-up. However, the slower buildup of echo density may be compensated for by a slow onset slope of late reverberation, which is possible if early and mid parts are modeled separately. Experiments with geometric properties of Boston Symphony Hall, Amsterdam Concertgebow and Vienna Musikverein have shown that the found proportions can be used in scaled versions troughout the entire topology.

### 3.2 Slope

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

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

where $\Delta t$ is the arrival time of the reflection after the direct sound.

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

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

where $g$ is the attenuation gain, $\Delta t$ 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.[4]

### 3.3 Modulation

Modulation is a key feature destinguishing recursive algorithms from i.e. most convolution based reverbs.

Randomization provides a means to overcome the lack of modal densitiy generally found in recursive algorithms [8] by smoothly changing comb filter sizes and output taps. The resulting pitch shifting artifacts reduce metallic ringing and add "warmth" to the reverb tail. Modulation may be introduced in multiple places. It is possible to use modulation thoughout[7].

First impressions with nested allpasses: LFO rates between 1,5...8Hz, depths of 0.1 to 1 per cent and liner ramping beween values (ramptime 300ms typical) seem to give good results. The LFO waveform was random, other authors report good results for sinusoidal modulation. Generaly it is agreed that modulation should follow different directions[8]. For big allpasses, comb sizes and late output taps, slower rates, longer ramp times and smaller depths were applied.

A more expensive way to implement time variance is the use of non-transposing delay lines. [5]

Animation simulates doppler effects caused by micro movements of the sound source and has proven to be useful for the generation of early reflection patterns[23]. Interestingly, pitch shifting artefacts seem to be more likely to be perceived as distubing 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[25]. Nested allpasses show a hightenend increase in echo density com-

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

[5] for an implementation see Appendix A

Figure 2: ER generator: Two calculated reflections extended by moderate feedback settings

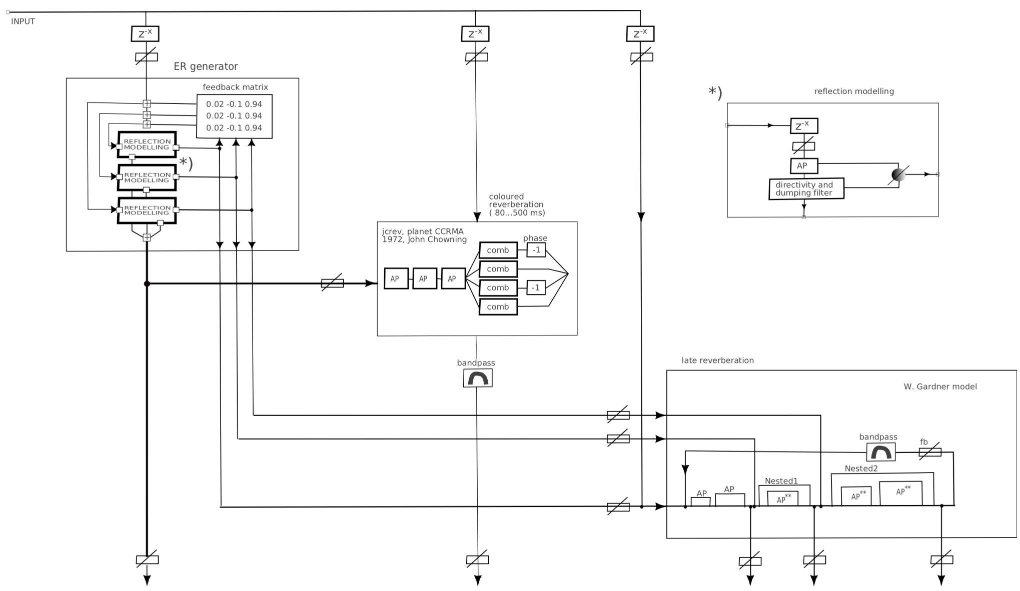


Figure 3: Example Design with separate algorithms for early, mid and late reverberation

pared to the quadratic increase predicted by the source image model[1] which does not take into account sound scattering surfaces.

In practice, nested allpasses have interesting aesthetical properties: although the overall response of these nested structures is allpass in the long term, impulsive sound's high frequency content may be attenuated in the first milliseconds. If different modulation schemes are used in combination with phase inversion, a "soft-phaser-like" effect results when summed in parallel: random phase cancelations occur, which change over time. (When listening to a Lexicon 480L without dry signal, this effect may be audible). Also high and low frequencies are attenuated in a desirable way.

FDNs may be used to create a similar effect. FDNs and digital waveguide networks [22] have to be investigated further: probabely 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

Mono input may be expanded to multichannel reverberation by controling the amount of correlation in the early, mid and late stream. Early reflections may be unique for every channel providing maximum decorrelation. For the mid stream usually a separate bulding block per channel is available fed by a unique combination of early reflections. A matrix may be used to sum different phase inverted and non-inverted outputs, as proposed by John Chowning in jcref [28].

Late reverberation may not be totaly uncorrelated for each channel. It is even possible to use the same algorithm with different output taps (Gardner uses this approach and the Lexicon 480L probabely, too) at least for two channels. In addition, it is possible to adjust correlation manually for each channel by deciding which and how many phase inverted output taps are summed for each channel. Decorrelation may be set to "mono" in an early design stage and later be rised step by step.

When separate building blocks for each channel are used, there are more options: different modulation schemes may be used or even different topologies for front and rear channels. Often rear channels reverberation is filtered by rising low frequencies by 3dB.

Stereo input to the reverb may be challanging for the reproduction of early reflections. As mentioned earlier, summing localisation may conflict with echo perception[32]. Solo instruments are often picked up with stereo microfone techniques to provide a wide image of the reproduced sound, i.e. romantic piano music will be presented using the full width of the stereo field. Early reflections may counteract this image in certain situations as previous work has shown[23]. Understandably reverb designers well known in the classical world, like Ralf Kessler (Quantec) or Dave Griessinger (Lexicon) frequently mentioned that a reverberant space *"does not differentiate between early and late reflected sound"*, indicating that their designs make very careful use of tapped delay lines to generate early reflections patterns, if at all (Quantec). As a solution, preferrably center and rear channels may be used to reproduce early reflections in order to preserve the role of L/R speakers to deliver the stereo image. This restraint does not apply to mid and late parts, however.

The principle of cross-coupling is shown in a design example by Dattorro[7]: main feedback loops are interleaved between channels so that input from one channel is bounced to the other side (at a later time) reversing stereo input. This concept could be expanded to arbitrary input and output channels if the input signals are sufficiently correlated.

## 4 The Reverberation Development Kit

The use of the development kit in PD is straight foreward: 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 limited number of GUI options.

The generated IR can be viewed, listened to and analyzed at any time thanks to PD's connectivity to jack[6].

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

A detailed description of every included algorithms 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 [23] it has been found that early reflections may not have the same frequency responses. 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.3 to the right) contain an allpass filter, whichs 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. 2 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 feedforeward paths may be delayed by one sample to adjust "brightness"[7]. Delays are fractional to allow for modulation, using third-order Lagrange interpolation.

### 4.2 Example Design

Fig 3 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 chosing the amount of directivity filtered jcref input.

By feeding small amounts of direct sound and allpass filtered reflections into the late model first, then choosing out tap locations accordingly, the onset of late reverberation can be shaped. In Fig.4 small AP feedback and late out taps lead to a slow

[6] Jack Audio Connection Kit, see http://sourceforge.net

[7] If the feedforeward path is unequal to the feedback path, the transfer function is no longer allpass.

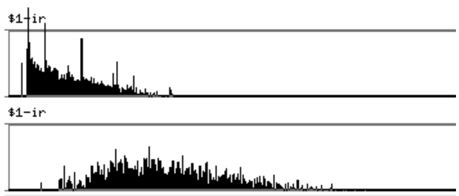


Figure 4: Example design: sparse early reflection pattern and onset control of late reverberation

onset.

The decay of early reflections 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˜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 sofisticated way which may be a challange 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 | Desciption | Lang | Licence |
|---|---|---|---|---|
| directivity | U.Schlemmer | data set for 13 instruments | C | GPLv2 |
| nested1a~ | | nested feedback-feedforeward combfilters | FAUST[1] | GPLv2 |
| nested1b~ | | nested feedback-feedforeward combfilters | FAUST | GPLv2 |
| nested1c~ | | nested allpass | FAUST | GPLv2 |
| nested2a~ | | nested feedback-feedforeward combfilters | FAUST | GPLv2 |
| nested2b~ | | nested feedback-feedforeward combfilters | FAUST | GPLv2 |
| nested2c~ | | nested AP with two APs in inner loop | 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[30] | 4 fdn butterfly (Hadamard matrix) prototype | PD | GpLv2 |
| ga2 | W. Gardner[10] | Gardner reverb with nested APs | PD | GPLv2 |
| jon | J. Dattorro[7] | 4 AP diffusor and "tank" structure | PD | GPLv2 |
| ezpst | | preset manager | PD | GPLv2 |
| burst | | random test tone generator | PD | GPLv2 |

Table 2: Abstractions

[1] [18], http://faust.grame.fr/
[2] Centre National de Creation Musicale, Lyon, France. Source code in FAUST's music.lib
[3] [28], transcription for FAUST's effect.lib: J.O.Smith, for description and analysis, see https://ccrma.stanford.edu/~jos/Reverb/Reverb_4up.pdf
[4] [29], https://ccrma.stanford.edu/~jos/pasp/FDN_Reverberators_Faust.html
[5] [28], transcription for FAUST's examples: GRAME, for description and analysis, see https://ccrma.stanford.edu/~jos/Reverb/Reverb_4up.pdf
[6] Source code, see http://rev-plugins.sourcearchive.com/lines/0.3.1/

**Appendix B: Selected Proportions**

| **concert hall** | **width[m]** | **lenth[m]** | **hight[m]** |
|---|---|---|---|
| Concertgebow Amsterdam | 28 | 44 | 17 |
| scaled to 15000 m3 | 25.48 | 40.04 | 15.47 |
| Boston Symphony Hall | 22.86 | 38.1 | 18.59 |
| 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