CLAIMS

1. A method for processing an audio signal (504, 802) in accordance with a room impulse response (300), the method comprising:

separately processing (502, 514, 812, 816a, 816b) the audio signal (504, 802) with an early part (301, 302) and a late reverberation (304) of the room impulse response (300), wherein processing the late reverberation (304) comprises generating a scaled reverberated signal; and

combining the audio signal processed with the early part of the room impulse response and the scaled reverberated signal,

wherein generating a scaled reverberated signal comprises

setting a gain factor according to a predefined correlation measure of the audio signal (504, 802) having a fixed value determined empirically on the basis of an analysis of a plurality of audio signals, and applying the gain factor, or

calculating a correlation measure of the audio signal (504, 802) using a correlation analysis of the audio signal (504, 802) and the gain factor, and applying the gain factor.

2. The method of claim 1, wherein the scaling is dependent on a condition of one or more input channels of the audio signal (504, 802), wherein the condition of the one or more input channels of the audio signal (504, 802) comprises one or more of the number of input channels, the number of active input channels, and an activity in the one or more input channel.

3. The method of claim 2, wherein the gain factor is determined based on the condition of the one or more input channels of the audio signal (504, 802).

4. The method of claim 3, wherein generating the scaled reverberated signal comprises applying the gain factor before, during or after processing the late reverberation (304) of the audio signal (504, 802).

5. The method of claim 3 or 4, wherein the gain factor is determined as follows:

$$g = c_u + \rho \cdot (c_c - c_u)$$

where

ρ = predefined or calculated correlation measure for the audio signal ~~(504, 802)~~,

c_u, c_c = factors indicative of the condition of one or more input channels of the audio signal ~~(504, 802)~~, with c_u referring to totally uncorrelated channels, and c_c relating to totally correlated channels,

wherein c_u and c_c are determined as follows:

$$c_u = 10^{\frac{10 \cdot \log_{10}(K_{in})}{20}} = \sqrt{K_{in}}$$

$$c_c = 10^{\frac{20 \cdot \log_{10}(K_{in})}{20}} = K_{in}$$

where

K_{in} = number of active input channels of the audio signal.

6. The method of one of claims 1 to 5, wherein the correlation analysis of the audio signal ~~(504, 802)~~ comprises determining for an audio frame of the audio signal ~~(504, 802)~~ a combined correlation measure, and wherein the combined correlation measure is calculated by combining correlation coefficients for a plurality of channel combinations of one audio frame, each audio frame comprising one or more time slots.

7. The method of claim 6, wherein combining the correlation coefficients comprises averaging a plurality of correlation coefficients of the audio frame.

8. The method of claim 6 or 7, wherein determining the combined correlation measure comprises:

(i) calculating an overall mean value for every channel of the one audio frame,

(ii) calculating a zero-mean audio frame by subtracting the mean values from the corresponding channels,

(iii) calculating for a plurality of channel combination the correlation coefficient, and

(iv) calculating the combined correlation measure as the mean of a plurality of correlation coefficients.

9. The method of one of claims 6 to 8, wherein the correlation coefficient for a channel combination is calculated as follows:

$$\rho[m,n] = \left| \frac{1}{(N-1)} \cdot \frac{\sum_i \sum_j x_m[i,j] \cdot x_n[i,j]^*}{\sum_j \sigma(x_m[j]) \cdot \sigma(x_n[j])} \right|$$

where

$\rho[m,n]$ = correlation coefficient,

$\sigma(x_m[j])$ = standard deviation across one time slot *j* of channel *m*,

$\sigma(x_n[j])$ = standard deviation across one time slot *j* of channel *n*,

x_m, x_n = zero-mean variables,

$i \forall [1,N]$ = frequency bands,

$j \forall [1,M]$ = time slots,

$m,n \forall [1,K]$ = channels,

$*$ = complex conjugate.

10. The method of one of claims 1 to 9, comprising delaying the scaled reverberated signal to match its start to the transition point from early reflections to late reverberation ~~(304)~~ in the room impulse response ~~(300)~~.

11. The method of one of claims 1 to 10, wherein the audio signal ~~(504, 802)~~ is a multichannel audio input signal, and wherein processing the late reverberation comprises applying the multichannel audio input signal to a downmixer ~~(410)~~ for downmixing the multichannel audio input signal ~~(504, 802)~~ to a signal having a lower number of channels and applying the downmixed audio signal to a reverberator.

12. A non-tangible computer product including a computer readable medium storing instructions for carrying out the method of one of claims 1 to 11 when being executed by a computer.

13. A signal processing unit, comprising:

an input for receiving an audio signal ~~(504, 802)~~, an early part processor for processing the received audio signal ~~(504, 802)~~ in accordance with an early part ~~(301, 302)~~ of a room impulse response ~~(300)~~,

a late reverberation processor for processing the received audio signal ~~(504, 802)~~ in accordance with a late reverberation of the room impulse response ~~(300)~~, the late reverberation processor configured to generate a scaled reverberated signal; and

an output for combining the processed early part of the received audio signal ~~(504, 802)~~ and the scaled reverberated signal into an output audio signal,

wherein the late reverberation processor configured to generate a scaled reverberated signal by

setting a gain factor according to a predefined correlation measure of the audio signal ~~(504, 802)~~ having a fixed value determined empirically on the basis of an analysis of a plurality of audio signals, and applying the gain factor, or

calculating a correlation measure of the audio signal ~~(504, 802)~~ using a correlation analysis of the audio signal ~~(504, 802)~~ and the gain factor, and applying the gain factor.

14. The signal processing unit of claim 13, wherein the late reverberation processor comprises:

a reverberator receiving the audio signal ~~(504, 802)~~ and generating a reverberated signal; and

a gain stage coupled to an input or to an output of the reverberator and controlled by the gain factor.

15. The signal processing unit of claim 13 or 14, comprising a correlation analyzer generating the gain factor dependent on the audio signal ~~(504, 802)~~.

16. The signal processing unit of one of claims 13 to 15, further comprising at least one of:

a low pass filter coupled to the gain stage, and

a delay element coupled between the gain stage and an adder, the adder further coupled to the early part processor and the output.

17. A binaural renderer, comprising a signal processing unit of one of claims 13 to 16.

18. An audio encoder for coding audio signals, comprising:

a signal processing unit of one of claims 13 to 16 or a binaural renderer of claim 17 for processing the audio signals prior to coding.

19. An audio decoder for decoding encoded audio signals, comprising:

a signal processing unit of one of claims 13 to 16 or a binaural renderer of claim 17 for processing the decoded audio signals.