Claims

1. An audio signal processing decoder having at least one frequency band ~~(36)~~ and being configured for processing an input audio signal ~~(37)~~ having a plurality of input channels ~~(38)~~ in the at least one frequency band ~~(36)~~, wherein the decoder ~~(1)~~ is configured

   to align the phases of the input channels ~~(38)~~ depending on inter-channel dependencies ~~(39)~~ between the input channels ~~(38)~~, wherein the phases of input channels ~~(38)~~ are the more aligned with respect to each other the higher their inter-channel dependency ~~(39)~~ is; and

   to downmix the aligned input audio signal to an output audio signal ~~(40)~~ having a lesser number of output channels ~~(41)~~ than the number of the input channels ~~(38)~~.

2. A decoder according to claim 1, wherein the decoder ~~(2)~~ is configured to analyze the input audio signal ~~(37)~~ in the frequency band ~~(36)~~, in order to identify the inter-channel dependencies ~~(39)~~ between the input audio channels ~~(38)~~ or to receive the inter-channel dependencies ~~(39)~~ between the input channels ~~(38)~~ from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

3. A decoder according to claim 1 or 2, wherein the decoder ~~(2)~~ is configured to normalize the energy of the output audio signal ~~(40)~~ based on a determined energy of the input audio signal ~~(37)~~, wherein the decoder ~~(2)~~ is configured to determine the signal energy of the input audio signal ~~(37)~~ or to receive the determined energy of the input audio signal ~~(37)~~ from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

4. A decoder according to one of the claims 1 to 3, wherein the decoder ~~(2)~~ comprises a downmixer ~~(42)~~ for downmixing the input audio signal ~~(37)~~ based on a downmix matrix ~~(M, $\boldsymbol{M}_{\mathrm{PA}}$)~~, wherein the decoder ~~(1)~~ is configured to calculate the downmix matrix ~~(M, $\boldsymbol{M}_{\mathrm{PA}}$)~~, in such way that the phases of the input channels ~~(38)~~ are aligned based on the identified inter-channel dependencies ~~(39)~~ or to receive a downmix matrix ~~(M, $\boldsymbol{M}_{\mathrm{PA}}$)~~ calculated in such way that the phases of the input channels ~~(38)~~ are aligned based on the identified inter-channel dependencies ~~(39)~~ from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

5. A decoder according to claim 4, wherein the decoder ~~(2)~~ is configured to calculate the downmix matrix ~~(M, $\boldsymbol{M}_{\mathrm{PA}}$)~~ in such way that the energy of the output audio signal ~~(41)~~ is normalized based on the determined energy of the input audio signal ~~(37)~~ or to receive the downmix matrix ~~(M, $\boldsymbol{M}_{\mathrm{PA}}$)~~, calculated in such way that the energy of the output audio signal ~~(41)~~ is normalized based on the determined energy of the input audio signal ~~(37)~~ from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

6. A decoder according to one of the claims 1 to 5, wherein the decoder ~~(2)~~ is configured to analyze time intervals ~~(43)~~ of the input audio signal ~~(37)~~ using a window function, wherein the inter-channel dependencies ~~(39)~~ are determined for each time frame ~~(43)~~ or wherein the decoder ~~(2)~~ is configured to receive an analysis of time intervals ~~(43)~~ of the input audio signal ~~(37)~~ using a window function, wherein the inter-channel dependencies ~~(39)~~ are determined for each time frame ~~(43)~~, from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

7. A decoder according to one of the claims 1 to 6, wherein the decoder (2) is configured to calculate a covariance value matrix $(C, \boldsymbol{C_y})$, wherein the covariance values $(c_{i,j}, C_{y,A,B})$ express the inter-channel dependency (39) of a pair of input audio channels (38) or wherein the decoder (2) is configured to receive a covariance value matrix $(C, \boldsymbol{C_y})$, wherein the covariance values $(c_{i,j}, C_{y,A,B})$ express the inter-channel dependency (39) of a pair of input audio channels (38), from an external device, such as from an encoder (1), which provides the input audio signal (37).

8. A decoder according to claim 7, wherein the decoder (2) is configured to establish an attraction value matrix $(A, \boldsymbol{P})$ by applying a mapping function $(f(c^{t}_{i,j}), T_{A,B})$ to the covariance value matrix $(C, \boldsymbol{C_y})$ or to a matrix (C^{t}) derived from the covariance value matrix $(C, \boldsymbol{C_y})$ or to receive an attraction value matrix $(A, \boldsymbol{P})$ established by applying a mapping function $(f(c^{t}_{i,j}), T_{A,B})$ to the covariance value matrix $(C, \boldsymbol{C_y})$ or to a matrix (C^{t}) derived from the covariance value matrix $(C, \boldsymbol{C_y})$, wherein the gradient of the mapping function $(f(c^{t}_{i,j}), T_{A,B})$ is preferably bigger or equal to zero for all covariance values $(c_{i,j}, C_{y,A,B})$ or values $(c'_{i,j}, ICC_{A,B})$ derived from the covariance values $(c_{i,j}, C_{y,A,B})$ and wherein the mapping function $(f(c^{t}_{i,j}), T_{A,B})$ preferably reaches values between zero and one for input values between zero and one.

9. A decoder according to claim 8, wherein the mapping function $(f(c^{t}_{i,j}), T_{A,B})$ is a non-linear function $(f(c^{t}_{i,j}), T_{A,B})$.

10. A decoder according to claim 8 or 9, wherein the mapping function $(f(c^{t}_{i,j}), T_{A,B})$ is equal to zero for covariance values $(c_{i,j}, C_{y,A,B})$ or values $(c'_{i,j}, ICC_{A,B})$ derived from the covariance values $(c_{i,j}, C_{y,A,B})$ being smaller than a first mapping threshold and/or wherein the mapping

function ~~$(f(c'_{i,j}), T_{A,B})$~~ is equal to one for covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ or values ~~$(c'_{i,j}, ICC_{A,B})$~~ derived from the covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ being bigger than a second mapping threshold.

11. A decoder according to one of the claims 8 to 10, wherein the mapping function ~~$(f(c'_{i,j}), T_{A,B})$~~ is represented by a function forming an S-shaped curve.

12. A decoder according to one of the claims 7 to 11, wherein the decoder ~~(2)~~ is configured to calculate a phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\text{int}})$~~, wherein the phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\text{int}})$~~ is based on the covariance value matrix ~~$(C, \boldsymbol{C}_y)$~~ and on a prototype downmix matrix ~~$(Q, \boldsymbol{M}_{\text{DMX}})$~~ or to receive a phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\text{int}})$~~, wherein the phase alignment coefficient matrix $(V, \boldsymbol{M}_{\text{int}})$ is based on the covariance value matrix ~~$(C, \boldsymbol{C}_y)$~~ and on a prototype downmix matrix ~~$(Q, \boldsymbol{M}_{\text{DMX}})$~~, from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

13. A decoder according to claim 12, wherein the phases and/or the amplitudes of the downmix coefficients ~~$(m_{i,j}, M_{\text{PA},A,B})$~~ of the downmix matrix ~~$(M, \boldsymbol{M}_{\text{PA}})$~~ are formulated to be smooth over time, so that temporal artifacts due to signal cancellation between adjacent time frames ~~(43)~~ are avoided.

14. A decoder according to claim 12 or 13, wherein the phases and/or the amplitudes of the downmix coefficients ~~$(m_{i,j}, M_{\text{PA},A,B})$~~ of the downmix matrix ~~$(M, \boldsymbol{M}_{\text{PA}})$~~ are formulated to be smooth over frequency, so that spectral artifacts due to signal cancellation between adjacent frequency bands ~~(36)~~ are avoided.

15. A decoder according to one of the claims 12 to 14, wherein the decoder ~~(2)~~ is configured to establish a regularized phase alignment coefficient matrix ~~$(\widetilde{M}, \boldsymbol{M}_{\mathrm{mod}})$~~ based on the phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\mathrm{int}})$~~ or to receive a regularized phase alignment coefficient matrix ~~$(\widetilde{M}, \boldsymbol{M}_{\mathrm{mod}})$~~ based on the phase alignment coefficient matrix $(V, \boldsymbol{M}_{\mathrm{int}})$ from an external device, such as from an encoder ~~(1)~~, which provides the input audio signal ~~(37)~~.

16. A decoder according to claim 15, wherein the downmix matrix ~~$(M, \boldsymbol{M}_{\mathrm{PA}})$~~ is based on the regularized phase alignment coefficient matrix ~~$(\widetilde{M}, \boldsymbol{M}_{\mathrm{mod}})$~~.

17. An audio signal processing encoder having at least one frequency band ~~(36)~~ and being configured for processing an input audio signal ~~(37)~~ having a plurality of input channels ~~(38)~~ in the at least one frequency band ~~(36)~~, wherein the encoder ~~(1)~~ is configured

to align the phases of the input channels ~~(38)~~ depending on inter-channel dependencies ~~(39)~~ between the input channels ~~(38)~~, wherein the phases of input channels ~~(38)~~ are the more aligned with respect to each other the higher their inter-channel dependency ~~(39)~~ is; and

to downmix the aligned input audio signal to an output audio signal ~~(40)~~ having a lesser number of output channels ~~(41)~~ than the number of the input channels ~~(38)~~.

18. An audio signal processing encoder having at least one frequency band ~~(36)~~ and being configured for outputting a bitstream ~~(7)~~, wherein the bitstream ~~(7)~~ contains an encoded audio signal ~~(37)~~ in the frequency band ~~(36)~~, wherein the encoded audio signal ~~(37)~~ has a plurality of encoded channels ~~(38)~~ in the at least one frequency band

~~(36)~~, wherein the encoder ~~(1)~~ is configured

~~to determine inter-channel dependencies (39) between the input channels (38) of the input audio signal (37) and to output the inter-channel dependencies (39) within the bitstream (7); and/or~~

~~to determine the energy of the encoded audio signal (37) and to output the determined energy of the encoded audio signal (37) within the bitstream (7); and/or~~

to calculate a downmix matrix ~~$(M, \boldsymbol{M}_{PA})$~~ for a downmixer ~~(3)~~ for downmixing the encoded audio signal ~~(37)~~ based on the downmix matrix ~~$(M, \boldsymbol{M}_{PA})$~~ in such way that the phases of the encoded channels ~~(38)~~ are aligned based on identified inter-channel dependencies ~~(39)~~, preferably in such way that the energy of an output audio signal of the downmixer ~~(41)~~ is normalized based on determined energy of the encoded audio signal ~~(37)~~ and to output the downmix matrix ~~$(M, \boldsymbol{M}_{PA})$~~ within the bitstream ~~(7)~~, wherein in particular the phases and/or amplitudes of downmix coefficients ~~$(m_{i,j}, M_{PA,A,B})$~~ of the downmix matrix ~~$(M, \boldsymbol{M}_{PA})$~~ are formulated to be smooth over time, so that temporal artifacts due to signal cancellation between adjacent time frames ~~(43)~~ are avoided and/or wherein in particular the phases and/or amplitudes of downmix coefficients ~~$(m_{i,j}, M_{PA,A,B})$~~ of the downmix matrix ~~$(M, \boldsymbol{M}_{PA}))$~~ are formulated to be smooth over frequency, so that spectral artifacts due to signal cancellation between adjacent frequency bands ~~(36)~~ are avoided; and/or

~~to analyze time intervals (43) of the encoded audio signal (37) using a window function, wherein the inter-channel dependencies (39) are determined for each time frame (43), and to output the inter-channel dependencies (39) for each time frame (43) within the bitstream (7);~~

~~and/or~~

~~to calculate a covariance value matrix $(C, \boldsymbol{C_y})$, wherein the covariance values $(c_{i,j})$ express the inter-channel dependency (39) of a pair of encoded audio channels (38) and to output the covariance value matrix $(C, \boldsymbol{C_y})$ within the bitstream (7); and/or~~

to establish an attraction value matrix ~~$(A, \boldsymbol{P})$~~ by applying a mapping function ~~$(f(c^t_{i,j}), T_{A,B})$~~, wherein the gradient of the mapping function ~~$(f(c^t_{i,j}), T_{A,B})$~~ is preferably bigger or equal to zero for all covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ or values ~~$(c'_{i,j}, ICC_{A,B})$~~ derived from the covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ and wherein the mapping function ~~$(f(c^t_{i,j}), T_{A,B})$~~ preferably reaches values between zero and one for input values between zero and one, in particular a non-linear function ~~$(f(c^t_{i,j}), T_{A,B})$~~, in particular a mapping function ~~$(f(c^t_{i,j}), T_{A,B})$~~, which is equal to zero for covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ or values ~~$(c'_{i,j}, ICC_{A,B})$~~ derived from the covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ being smaller than a first mapping threshold and/or which is equal to one for covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ or values ~~$(c'_{i,j}, ICC_{A,B})$~~ derived from the covariance values ~~$(c_{i,j}, C_{y,A,B})$~~ being bigger than a second mapping threshold and/or which is represented by a function forming an S-shaped curve, to the covariance value matrix ~~$(C, \boldsymbol{C_y})$~~ or to a matrix ~~(C^t)~~ derived from the covariance value matrix ~~$(C, \boldsymbol{C_y})$~~ and to output the attraction value matrix ~~$(A, \boldsymbol{P})$~~ within the bitstream ~~(7)~~; and/or

to calculate a phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\text{int}})$~~, wherein the phase alignment coefficient matrix ~~$(V, \boldsymbol{M}_{\text{int}})$~~ is based on the covariance value matrix ~~$(C, \boldsymbol{C_y})$~~, and on a prototype downmix matrix ~~$(Q, \boldsymbol{M}_{\text{DMX}})$~~; ~~and/or~~

~~to establish a regularized phase alignment coefficient matrix ($\widetilde{M}, \boldsymbol{M}_{\mathrm{mod}}$) based on the phase alignment coefficient matrix V and to output the regularized phase alignment coefficient matrix ($\widetilde{M}, \boldsymbol{M}_{\mathrm{mod}}$) within the bitstream (7)~~.

19. An audio signal processing encoder according to claim 18, wherein the encoder is configured to determine inter-channel dependencies between the input channels of the input audio signal and to output the inter-channel dependencies within the bitstream; and/or

to determine the energy of the encoded audio signal and to output the determined energy of the encoded audio signal within the bitstream.

20. An audio signal processing encoder according to claim 18 or 19, wherein the encoder is configured to analyze time intervals of the encoded audio signal using a window function, wherein the inter-channel dependencies are determined for each time frame, and to output the inter-channel dependencies for each time frame within the bitstream.

21. An audio signal processing encoder according to one of the claims 18 to 20, wherein the encoder is configured to calculate a covariance value matrix , wherein the covariance values express the inter-channel dependency of a pair of encoded audio channels and to output the covariance value matrix within the bitstream.

22. An audio signal processing encoder according to one of the claims 18 to 21, wherein the encoder is configured to establish a regularized phase alignment coefficient matrix based on the phase alignment coefficient matrix V and to output the regularized phase alignment coefficient matrix within the bitstream.

~~19.~~23. A system comprising:

an audio signal processing decoder ~~(2)~~ according to one of the claims 1 to 16, and

an audio signal processing encoder ~~(1)~~ according to one of the claims 17 ~~or 18~~to 22.

~~20.~~24. A method for processing an input audio signal ~~(37)~~ having a plurality of input channels ~~(38)~~ in a frequency band ~~(36)~~, the method comprising the steps:

analyzing the input audio signal ~~(37)~~ in the frequency band ~~(36)~~, wherein inter-channel dependencies ~~(39)~~ between the input audio channels ~~(38)~~ are identified;

aligning the phases of the input channels ~~(38)~~ based on the identified inter-channel dependencies ~~(39)~~, wherein the phases of the input channels ~~(38)~~ are the more aligned with respect to each other the higher their inter-channel dependency ~~(39)~~ is;

downmixing the aligned input audio signal to an output audio signal ~~(40)~~ having a lesser number of output channels ~~(41)~~ than the number of the input channels ~~(38)~~ in the frequency band ~~(36)~~.

~~21.~~25. A computer program for implementing the method of claim 2~~0~~4 when being executed on a computer or signal processor.