

**APPARATUS AND METHOD FOR THE TIME-ORIENTED
EVALUATION AND OPTIMIZATION OF STEREOPHONIC OR
PSEUDO-STEREOPHONIC SIGNALS**

The invention relates to devices and methods
5 for stereophonizing a mono signal resp. for obtaining
pseudostereophonic signals.

In particular, time resp. phase differences
between different signals are more closely analyzed, in
order on the one hand to be able to draw conclusions as
10 to the acoustic properties of the signals and on the
other hand to synthesize stereophonic or
pseudostereophonic signals (this term also includes
signals with more than two channels) that also exhibit
these or other acoustic properties in an ideal form.

15 In particular, stereophonic or
pseudostereophonic signals are analyzed that have been
generated according to devices or methods according to
EP1850639 or WO2009/138205 or WO2011/009649 or
WO2011/009650 or CH01264/10 resp. PCT/EP2011/063322 and
20 that are to be optimized either in terms of their
psychoacoustic properties or that are to be matched
with existing stereophonic or pseudostereophonic
signals in terms of their psychoacoustic properties.

Methods so far in connection with EP1850639 or
25 WO2009/138205 optimize the parameters exclusively in
terms of an angle-dependent virtualisation of a classic
MS array. According to the invention, this arrangement
is additionally subjected to a time-dependent
virtualization.

30 The present invention not only explores all
possibilities of such a virtualization - partly through
the radical simplification of the systems existing with

EP1850639 or WO2009/138205 or WO2011/009649 or
WO2011/009650 or CH01264/10 resp. PCT/EP2011/063322 -
but even achieves their automatization through the
unexpected reformulation of the so-called inverse
5 problem for spatial audio.

Hereinafter, the state of the art will be
described, in particular with respect to devices or
methods for obtaining, improving or optimizing
10 stereophonic or pseudostereophonic audio signals.

EP0825800 (Thomson Brandt GmbH) proposes the
generation of different kinds of signals from a mono
input signal by means of filtering, said signals being
used - for example by using a method proposed by
15 Lauridsen based on amplitude and time difference
corrections, depending on the recording situation - to
generate virtual single-band stereo signals separately,
which are then subsequently combined to form two output
signals.

20 WO/2009/138205 as well as EP1850639 describe
among others a method for methodically evaluating the
angle of incidence for the sound event that is to be
mapped, said angle of incidence being enclosed by the
main axis of the microphone and the directional axis
25 for the sound source, this being achieved by applying
time differences and amplitude corrections which are
functionally dependent on the original recording
situation (which may be interpolated by using the
system). The contents of WO/2009/138205 as well as of
30 EP1850639 are hereby incorporated as a reference.

US5173944 (Begault Durand) applies HRTFs (Head
Related Transfer Functions), which correlate with 90,
120, 240 and 270 degrees azimuth respectively, to the
differently delayed but uniformly amplified monophonic

input signal, the formed signals in turn being finally
superimposed on the original mono signal. In this case,
the amplitude correction and the time difference
corrections are chosen independently of the recording
5 situation.

US5671287 (Michael A. Gerzon) proposes among
others cascaded all-pass filters for forming a pseudo-
stereophonic signal. A further suggestion relates to the
use of all-pass filters in both channels, with a
10 frequency-dependent rotation matrix being connected
downstream thereof; although this method manages to
disperse sound sources having the same frequency, there
is no perceptible spatial separation of these sound
sources as achieved by EP1850639 or W02009/138205 or
15 W02011/009649 or W02011/009650 or PCT/EP2011/063322.

W02011/009649 proposes the ostensibly not
purposeful downstream connection of one or more
panoramic potentiometers or equivalent means in a
device according to W02009/138205 or EP1850639 after
20 stereo decoding has taken place (after an MS matrix,
for which the relation

$$L = (M + S) * 1/\sqrt{2}$$

and

$$R = (M - S) * 1/\sqrt{2}$$

25 applies, has been passed through), which - unlike in
the case of intensity stereophonic signals, i.e. for
stereo signals which differ exclusively in terms of
their levels but not in terms of time resp. phase
differences or different frequency spectra - do not
30 result in the intended narrowing of the image width or
in the intended shifting of the localisation direction
of the obtained stereo signals, but rather result in

the degree of correlation being increased or decreased. The contents of CH701497 resp. W02011/009649 are hereby incorporated as a reference.

W02011/009650 enables an optimum choice of
 5 those parameters that form the basis for generating stereophonic or pseudostereophonic signals. The user is provided with means for specifying the degree of correlation, the definition range, the loudness as well as further parameters of the resulting signals
 10 according to psychoacoustic aspects, and hence for preventing artifacts. The contents of CH01776/09 resp. W02011/009650 are hereby incorporated as a reference.

CH01264/10 resp. PCT/EP2011/063322 for the first time makes it possible to evaluate invariants for
 15 example of the combination of two or more at least partly slightly decorrelated signals or of their transfer functions, wherein these signals or transfer functions appear to be completely random (such as for example audio signals), so that for example for two or
 20 more different signal sections it is possible to draw conclusions as to their properties (for example the sum of the transfer functions

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

$$25 \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

for a stereophonic audio signal $x(t)$, $y(t)$, where $x(t)$ represents the function value of the left input signal
 30 at the point in time t , $y(t)$ represents the function value of the right input signal at the point in time t , and thus for example devices or methods for obtaining, improving or optimizing stereophonic or pseudostereophonic audio signals can consequently be
 35 calibrated. Since the contents of this document have not been published at the time of the present

application, their contents are reproduced in full hereafter.

Disclosure of the invention

According to one aspect of the invention, a
 5 method for stereophonizing a mono signal resp. for
 obtaining pseudostereophonic signals is proposed,
 wherein calculated time differences are multiplied,
 prior to their being used on a mono signal to be
 rendered stereophonic, with a time parameter (s), which
 10 is greater than zero, and thus yield new time
 differences ($L_A' = L_A * s$, $L_B' = L_B * s$ resp. $L_\alpha' = L_\alpha * s$, $L_\beta' = L_\beta * s$).

According to another aspect of the invention, a
 further time parameter s is introduced in addition to
 15 the parameters f (resp. n) which describe the
 directional pattern of the signal that is to be
 rendered stereophonic, the angle φ - to be ascertained
 manually or by metrology - enclosed by the main axis
 and the sound source, the fictitious left opening angle
 20 α and the fictitious right opening angle β , the
 attenuations λ or else ρ for forming the resulting
 stereo signal in the case of W02009/138205 resp. the
 angle φ enclosed by the main axis and the sound source
 as well as the attenuation λ or else ρ for forming the
 25 resulting stereo signal in the case of EP1850639. This
 additional time parameter s, when multiplied with the
 time differences L_α and L_β (in the case of
 W02009/138205) resp. with the time differences L_A and
 L_B (in the case of EP1850639), determines new time
 30 differences L_α' and L_β' (in the case of W02009/138205)
 resp. new time differences L_A' and L_B' (in the case of
 EP1850639), which replace the former time differences
 L_α and L_β resp. L_A and L_B .

Therefore, to start with, for any $s > 0$ the following is in principle true:

$$(1D) \quad L_{\alpha}' = L_{\alpha} * s = \left\{ -\frac{f(\alpha)}{2\sin\alpha} + \sqrt{\frac{f^2(\alpha)}{4\sin^2\alpha} + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi) / \sin\alpha} \right\} * s$$

and

$$(2D) \quad L_{\beta}' = L_{\beta} * s = \left\{ -\frac{f(\beta)}{2\sin\beta} + \sqrt{\frac{f^2(\beta)}{4\sin^2\beta} + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta} \right\} * s$$

10 (in the case of W02009/138205) resp.

$$(3D) \quad L_A' = L_A * s = [\sqrt{5/4 - \sin\varphi} - 1/2] * s$$

and

$$(4D) \quad L_B' = L_B * s = [\sqrt{5/4 + \sin\varphi} - 1/2] * s$$

(in the case of EP1850639).

15 The choice of s , as shown in practice, is not trivial. If s is chosen too small, the pseudostereophonic effect to be achieved disappears, if s is chosen too great, disturbing artifacts will result. If s is about 100 milliseconds, this will yield
20 for a device or a method according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 ideal pseudostereophonic signals that exhibit the same quality as with a classic MS recording technique.

25 Tests have altogether shown, that the ideal range for s is between 29 milliseconds and 146 milliseconds.

A variant embodiment of the invention that is advantageous for the user thus affords the possibility of freely selecting $s > 0$. Similarly, the devices and methods or systems represented in EP1850639 or
 5 W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 allow the automatized or interactive determination of the new parameter s . For example, in FIG. 15D, in addition to f (resp. n), which describe the directional pattern of
 10 the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left opening angle α and the fictitious right opening angle β , in the same manner the new
 15 parameter s , which has a direct influence on the sum of the transfer functions

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

$$20 \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

for a stereophonic audio signal $x(t)$, $y(t)$ (where $x(t)$ represents the function value of the left input signal at the point in time t , $y(t)$ represents the function
 25 value of the right input signal at the point in time t), is optimized iteratively in the same fashion as f (resp. n), φ , α , β .

This means in particular automatically and optimally choosing such parameters which form the basis
 30 for the generation of stereophonic or pseudostereophonic signals, resp. a method and a device for optimally and automatically determining particularly the parameters (φ , λ , ρ resp. f (resp. n), α , β - and henceforth newly s) while generating said
 35 stereophonic or pseudostereophonic signals.

Such a method resp. such a device are intended to be used to select, from a plurality of decorrelated, in particular pseudostereophonic, signal variants, those whose decorrelation is found to be particularly advantageous.

In particular, it should be possible to influence the selection criteria themselves in a form as efficient and compact as possible in order to be able to convert signals of different nature (for example speech in contrast to music recordings) into the optimized reproduction thereof.

According to one aspect, W02011/009650 proposes a device and a method for obtaining pseudostereophonic output signals $x(t)$ and $y(t)$ by using a stereo decoder, wherein $x(t)$ is the function value of the resulting left output channel at the time t , and $y(t)$ is the function value of the resulting right output channel at the time t , in which the obtainment is iteratively optimized until $\langle x(t), y(t) \rangle$ is within a predetermined definition range.

If there are dropouts or similar defects, however, an insignificant quantity of single points may lie outside the definition range. In this case, the obtainment is iteratively optimized until a portion of $\langle x(t), y(t) \rangle$ is within the predetermined definition range.

The desired definition range is preferably stipulated by a single numerical parameter a , where preferably $0 \leq a \leq 1$. This parameter and hence the definition range can be usefully stipulated for example by the inequality

$$\operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} \cdot 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1$$

wherein the relations

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

and

$$g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

- 5 apply for the complex transfer functions $f^*[x(t)]$ and $g^*[y(t)]$ of the output signal $x(t)$, $y(t)$.

The user can arbitrarily stipulate such a definition range, on the basis of the unit circle of the complex number plane resp. of the imaginary axis
 10 (if the maximum level of the output signal $x(t)$, $y(t)$ has been normalized on the unit circle), by using the parameter a , $0 \leq a \leq 1$.

This principle also remains valid when a reference system other than the unit circle of the complex number plane is chosen and a different new
 15 definition range is defined. "Definition range" is therefore understood generally to mean an admissible range of values for $\langle x(t), y(t) \rangle$ of the output signal $x(t)$, $y(t)$, which, overall, is intended to contain
 20 $\langle x(t), y(t) \rangle$ in full or in part (for example in the case of defective sound recordings which show what are known as dropouts).

In a preferred variant embodiment, the degree of correlation of the output signals ($x(t)$ and $y(t)$) is
 25 normalized. In a preferred variant embodiment, the level of the maximum of the resulting left and right channel is normalized. In this way, certain parameters can be iteratively optimized in order to attain the desired definition range, without said parameters
 30 affecting the degree of correlation or the level of the

maximum of the resulting left channel and right channel.

It also makes sense if – for extremely different parameterizations for φ resp. f (resp. n), α , β and henceforth newly s – criteria which are dependent on $|\langle x(t), y(t) \rangle|$ are used for the stipulation. For this purpose, according to the invention, a corresponding range of values dependent on $|\langle x(t), y(t) \rangle|$ is normalized, so as to constitute a criterion for the optimization of the parameters.

In one embodiment, a method for obtaining pseudostereophonic output signals $x(t)$ and $y(t)$ by using a converter is therefore proposed, wherein $x(t)$ is the function value of the resulting left output channel at the time t , wherein $y(t)$ is the function value of the resulting right output channel at the time t , wherein the complex transfer functions $f^*[x(t)]$ and $g^*[y(t)]$ of the output signals are defined:

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

$$g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i),$$

in which the obtaining is iteratively optimized until the following criterion is satisfied:

$$\operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1,$$

where $0 \leq a \leq 1$ stipulates the desired definition range.

A remarkable aspect of the methods for obtaining pseudostereophonic signals according to WO/2009/138205 or according to EP1850639 is the fact that they always provide a perfect center signal. For this reason, the short time cross correlation

$$(1B) \quad r = (1/2T) * \int_{-T}^T x(t)y(t) dt$$

$$* (1/x(t)_{eff}y(t)_{eff})$$

5 is introduced here for the time interval $[-T, T]$ and the output signals $x(t)$ from the left channel and $y(t)$ from the right channel.

As already mentioned, it makes sense if a uniform degree of correlation is attained for extremely
 10 different parameterizations for φ resp. f (resp. n), α , β and henceforth newly s . For this purpose, according to the invention, the degree of correlation between the output signals ($x(t)$ and $y(t)$) is normalized. This normalization can preferably be stipulated by means of
 15 the specific variation of λ (left attenuation) or ρ (right attenuation).

On the basis of the uniform degree of correlation, the signal attained can now be systematically subjected to evaluation criteria that
 20 can be influenced by the user.

It also makes sense if a uniform level for the maximum of the resulting left and right channel is being attained for extremely different parameterizations for φ resp. f (resp. n), α , β and henceforth
 25 newly s . For this purpose, in the represented system, the level of the maximum of the resulting left and right channel is normalized, so that this level is not influenced by the optimization of the parameters.

It makes sense, for example, for the level
 30 control for the maximum of the left signal L and of the right signal R to initially be uniformly confined for example to 0 dB by means of a first logic element.

It also makes sense if – for extremely different parameterizations for φ resp. f (resp. n), α , β and henceforth newly s – criteria which are dependent on $\langle x(t), y(t) \rangle$ or $|\langle x(t), y(t) \rangle|$ are used for the stipulation. For this purpose, according to the invention, a corresponding range of values is normalized, so as to constitute a criterion for the optimization of the parameters.

$x(t)$ and $y(t)$ are mapped within the unit circle of the complex number plane. The function $f^*[x(t)] + g^*[y(t)]$ can now be analyzed in more detail in order to draw conclusions concerning the quality of the respective output signal from a device according to WO/2009/138205 or EP1850639, for example. Any decorrelation between the two signals $f^*[x(t)]$ and $g^*[y(t)]$ is in this case equivalent to a deflection on the real axis when analyzing the function $f^*[x(t)] + g^*[y(t)]$.

The stereo decoder is therefore optimized according to said criterion for example for $|\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}|$ and for $|\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}|$, namely

$$\operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} \cdot 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1,$$

where $0 \leq a \leq 1$ stipulates the desired definition range.

This method has proven itself to be particularly advantageous, since a single parameter, namely a , takes optimum account of, in particular, the different nature of the output signals from a device or a method according to WO/2009/138205 or EP1850639. The parameter may preferably be dependent on the type of the audio signal, for example in order to process speech or music differently on a manual or automatic

basis. In the case of speech, unlike music recordings, the definition range determined by the parameter a preferably needs to be restricted significantly due to disturbing artifacts such as high frequency sidetones during the articulation.

In addition, given the limitation to a single parameter a , any optimum function range can be chosen for $f^*[x(t)] + g^*[y(t)]$ based on the unit circle resp. the imaginary axis.

If the signals $x(t)$, $y(t)$ do not satisfy the aforementioned conditions, the invention involves optimization being carried out by re-determining the parameters φ or f (resp. n) or α or β or newly s - according to an iterative procedure that is matched with the function values $x[t(\varphi, f, \alpha, \beta, s)]$ and $y[t(\varphi, f, \alpha, \beta, s)]$ resp. $x[t(\varphi, n, \alpha, \beta, s)]$ and $y[t(\varphi, n, \alpha, \beta, s)]$ - whilst executing steps described so far until $x(t)$ and $y(t)$ meet the aforementioned conditions.

In a further step, the relief of the function $f^*[x(t)] + g^*[y(t)]$ for example is now analyzed for the purpose of maximizing the function values thereof. It is possible to show that this procedure is equivalent to the maximization of

$$(6B) \quad \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt$$

this expression, for its part, remains less than or equal to the value of

$$(7aB) \quad \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)]\}}\} dt$$

$$+ g^*[y(t)]\}}\} dt.$$

In this case too, the user is provided with a tool insofar as he has a free choice of the limit value R^* (or the deviation Δ defined by the inequality (8aB), see below) for this maximization within the context of (8aB). Overall, the following condition must be met for the total number of possible signal variants $x_j(t)$, $y_j(t)$:

$$\begin{aligned}
 (8aB) \quad & 0 \leq R^* - \Delta \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt \\
 & \leq \max \int_{-T}^T |f^*[x_j(t)] \{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi \\
 & + g^*[y_j(t)]| dt \\
 & \leq R^* + \Delta \\
 & \leq \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)] \\
 & + g^*[y(t)]\}}\} dt
 \end{aligned}$$

R^* and Δ are directly related to the loudness of the output signal that is to be attained (i.e. to those parameters which the listener also takes as a basis for assessing the validity of a stereophonic mapping).

If the neighborhood of the limit value R^* , defined by Δ , or the maximum of all possible integrated reliefs is not reached, optimization in terms of the limit value R^* and the deviation Δ or in terms of the

aforementioned maximum – in accordance with an iterative procedure that is matched with the function values $x[t(\varphi, f, \alpha, \beta, s)]$ and $y[t(\varphi, f, \alpha, \beta, s)]$ resp. $x[t(\varphi, n, \alpha, \beta, s)]$ and $y[t(\varphi, n, \alpha, \beta, s)]$ – involves new parameters φ resp. f resp. α resp. β resp. newly s being determined, and all steps described so far being executed until signals $x(t)$, $y(t)$ resp. parameters φ resp. λ resp. ρ resp. f (resp. n) resp. α resp. β resp. newly s result, which correspond to optimum stereophonization.

With an appropriate choice of the degree of correlation r , of the parameter a – stipulating the desired respective definition range – and of the limit value R^* and also deviation Δ thereof, it is possible to configure optimum systems for the respective area of application (for example speech or music reproduction) for the respective nature of the input signals.

On the basis of the algebraic invariants presented in PCT/EP2011/063322, see below, it is possible, as part of the invention, to define a new weighting as follows:

For this purpose, a first optimization according to W02011/009650, FIG. 15D, 2B, 3B to 5B, is performed on a signal section of the length t_1 . The outputs of FIG. 5B are connected for example to a module 6001 according to FIG. 16D and the invariants (constituted in the intersection points ξ_{h1} of the sum of the complex transfer functions $f^*[x(t_1)] = [x(t_1)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_1)] = [y(t_1)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$ or also $(-1, -1, 2)$ and $(1, 1, 1)$ and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1 , u_1 of the represented algebraic model coincides here with the real axis, the axis x_2 , u_2 with the imaginary axis) are analyzed in

terms of their statistical distribution. All ξ_{h1} with the total number k_1 are stored in a memory ("stack") valid for all further described operation sequences; the mean value

$$\begin{aligned} &5 \quad \xi_1^o := \left(\sum_{h_1=1}^{k_1} \xi_{h1} \right) / k_1 \\ &\quad h_1 = 1 \end{aligned}$$

is likewise calculated. The latter is stored together with the parameterization ϕ_1, f_1 (resp. n_1), α_1, β_1 and
 10 henceforth newly s_1 , determined by means of said first optimization, in a further dictionary valid for all further described operation sequences.

According to the function command 6004, in a second step a second optimization according to
 15 W02011/009650, FIG. 15D, 2B, 3B to 5B is then performed on a signal section t_2 of any length. The outputs of FIG. 5B are in turn connected to the module 6001 of FIG. 16D and the invariants (constituted in the intersection points ξ_{h2} of the sum of the complex
 20 transfer functions $f^*[x(t_2)] = [x(t_2)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_2)] = [y(t_2)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$ or also $(-1, -1, 2)$ and $(1, 1, 1)$ and located in the 1st or also 3rd quadrants of the complex number plane –
 25 the axis of x_1, u_1 of the represented algebraic model coincides here with the real axis, the axis x_2, u_2 with the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{h2} from the total number k_2 are stored in a memory ("stack") valid for all
 30 further described operation sequences; the mean value

$$\begin{aligned} &\xi_2^o := \left(\sum_{h_2=1}^{k_2} \xi_{h2} \right) / k_2 \\ &\quad h_2 = 1 \end{aligned}$$

is likewise calculated. The latter is in turn added together with the parameterization φ_2, f_2 (resp. n_2), α_2, β_2 and henceforth newly s_2 , determined by means of said second optimization, to the first mean value ξ_1° as well as its parameterization φ_1, f_1 (resp. n_1), α_1, β_1, s_1 in the dictionary valid for all further described operation sequences. Since the memory ("stack") now contains more than one mean value, the module 6002 of FIG. 16D is then activated.

10 The latter calculates the mean value ξ^*_2 of all intersection points ξ_{h1}, ξ_{h2} stored in the stack:

$$\xi^*_2 := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} \right) / (k_1 + k_2)$$

15 and selects from the dictionary among the mean values ξ_1°, ξ_2° with their associated parameterization the mean value closest to ξ^*_2 . If this is the case for both mean values ξ_1°, ξ_2° , ξ_1° resp. the parameterization φ_1, f_1 (resp. n_1), α_1, β_1, s_1 is selected from the dictionary.
20 The mean value selected from the dictionary is then transmitted together with ξ^*_2 to the module 6003. The latter verifies whether the mean value selected by the module 6002 is within the interval $[-\sigma + \xi^*_2, \xi^*_2 + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely
25 selectable by the user, of the Gauss distribution fictitiously set as zero in ξ^*_2 :

$$f^\sim(z_2^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_2^* - \xi^*_2)^2)/\sigma^2}$$

30 If the mean value selected by the module 6002 of FIG. 16D is within the interval $[-\sigma + \xi^*_2, \xi^*_2 + \sigma]$, the parameterization selected by the module 6002 according to 6010 in the arrangement of FIG. 6D resp. FIG 15D (which shows again for the sake of clarity the amplifier 717 and the MS matrix, wherein both of which

are to be passed through only once) resp. the outputs 6006 and 6007 of FIG. 15D are activated, likewise the outputs 6008 and 6009 of FIG. 2B. The output 6006 runs into the input 6006 of FIG. 16D, the output 6007 runs
 5 into the input 6007 of FIG. 16D, the output 6008 runs into the input 6008 of FIG. 16D and the output 6009 runs into the input 6009 of FIG. 16D. Reference 6006 of FIG. 16D directly represents the output signal $x(t)$ from the module 6003 of FIG. 16D, reference 6007 of
 10 FIG. 16D directly represents the output signal $y(t)$ from the module 6003 of FIG. 16D, 6008 of FIG. 16D directly represents the output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D, 6009 of FIG. 16D directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D. These
 15 signals, in the signal processing described above, are to be treated as if they represented the output signals of FIG. 5B, which with FIG. 16D then constitutes, in this example of application, an inseparable unit.

20 If the mean value selected by the module 6002 is outside the interval $[-\sigma + \xi_2^*, \xi_2^* + \sigma]$, in a q^{th} step a q^{th} optimization is performed according to the extention, described here, of W02011/009650, FIG. 15D, 2B, 3B to 5B on a signal section t_q of any length. The
 25 outputs of FIG. 5B are in turn connected to the module 6001 of FIG. 16D and the invariants (constituted at the intersection points ξ_{hq} of the sum of the complex transfer functions $f^*[x(t_q)] = [x(t_q)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_q)] = [y(t_q)/\sqrt{2}] * (1 + i)$ with the half-
 30 plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$ or also $(-1, -1, 2)$ and $(1, 1, 1)$ and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1, u_1 of the represented algebraic model coincides here with the real axis, the axis x_2, u_2 with
 35 the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{hq} with the total number k_q are added to $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq-1}$ in a memory ("stack")

valid for all further described operation sequences;
the mean value

$$\xi_{\circ q}^{\circ} := \left(\sum_{h_q=1}^{k_q} \xi_{hq} \right) / k_q$$

is likewise calculated. The latter is in turn added together with the parameterization φ_q, f_q (resp. n_q), α_q, β_q , and henceforth newly s_q determined by means of said q^{th} optimization, to the first mean values $\xi_{\circ 1}^{\circ}, \xi_{\circ 1}^{\circ},$
10 $\dots, \xi_{\circ q-1}^{\circ}$ as well as to their parameterizations φ_1, f_1 (resp. n_1), α_1, β_1, s_1 ; φ_2, f_2 (resp. n_2), α_2, β_2, s_2 ; $\dots$; φ_{q-1}, f_{q-1} (resp. n_{q-1}), $\alpha_{q-1}, \beta_{q-1}, s_{q-1}$, in the dictionary valid for all further described operation sequences. Since the memory ("stack") now contains more than one
15 mean value, the module 6002 of FIG. 16D is then activated.

The latter calculates the mean value $\xi_{\circ q}^{*}$ of all intersection points $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq}$ stored in the stack:

$$\xi_{\circ q}^{*} := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} + \dots + \sum_{h_q=1}^{k_q} \xi_{hq} \right) / (k_1 + k_2 + \dots k_q)$$

and selects from the dictionary among the mean values $\xi_{\circ 1}^{\circ}, \xi_{\circ 2}^{\circ}, \dots, \xi_{\circ q}^{\circ}$ with their associated parameterization
25 φ, f (resp. n), α, β , and henceforth newly s , the mean value closest to $\xi_{\circ q}^{*}$. If this is the case for different parameterizations, the parameterization that appears most often in the dictionary is selected. If several parameterizations appear the same number of times, the
30 one that exhibits the widest scattering in the dictionary is selected, i.e. the one for which the difference $d - c$ is maximum, where d represents the last and c the first index number of the optimization

steps respectively undergone. If this too applies to several parameterizations, the first one that appears is selected. If two mean values from $\xi_1^o, \xi_2^o, \dots, \xi_q^o$ are closest to ξ_q^* , insofar as in a $q-1^{\text{th}}$ step one of the
 5 two mean values resp. their associated parameterization is selected from the dictionary, the very same one resp. its associated parameterization is retained. The mean value selected from the dictionary is then transmitted together with ξ_q^* to the module 6003 of FIG.
 10 16D. The latter verifies whether the mean value selected by the module 6002 of FIG. 16D is within the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire process illustrated here,
 15 of the Gauss distribution fictitiously set as zero in ξ_q^* :

$$f^{\sim}(z_q^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_q^* - \xi_q^*)^2)/\sigma^2}$$

If the mean value selected by the module 6002 of FIG. 16D is within the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$,
 20 the parameterization selected by the module 6002 according to 6010 in the arrangement of FIG. 6D resp. FIG 15D resp. the outputs 6006 and 6007 of FIG. 15D are activated, likewise the outputs 6008 and 6009 of FIG. 2B as well as the corresponding inputs and outputs of
 25 FIG. 16D. Reference 6006 of FIG. 16D again directly represents the output signal $x(t)$ from the module 6003 of FIG. 16D, reference 6007 of FIG. 16D directly represents the output signal $y(t)$ from the module 6003 of FIG. 16D, 6008 of FIG. 16D directly represents the
 30 output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D, 6009 of FIG. 16D directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D. These signals, in the signal processing described further below, are again to be treated as if
 35 they represented the output signals of FIG. 5B, which

with FIG. 16D constitutes, in the present example of application, an inseparable unit.

If the mean value selected by the module 6002 of FIG. 16D is outside the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$,
 5 in a $q+1^{\text{th}}$ step a $q+1^{\text{th}}$ optimization is performed in the same form as for the q^{th} step and for the q^{th} optimization. The process is repeated as long as necessary until one element of the dictionary fulfills the above requirements or a maximum number of allowed
 10 optimization steps has been reached.

FIG. 5C shows the convergence behavior of the just established weighting function for three optimization steps: 5001 represents in this case the first mean value ξ_1^o , 5002 the second mean value ξ_2^o ,
 15 5003 the first Gauss distribution fictitiously set as zero in ξ_2^* :

$$f^{\sim}(z_2^*) = (1 / (\sqrt{(2\pi)} * \sigma)) * e^{-(1/2)*((z_2^* - \xi_2^o)^2 / \sigma^2)},$$

where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire
 20 process illustrated here, 5004 the third mean value ξ_3^o , which remains within the inflexion points defined by σ of the Gauss distribution 5005 of same standard deviation, fictitiously set as zero in ξ_3^* , and thus fulfills the convergence criterion.

25 In each case, the result is a parameterization ϕ , f (resp. n), α , β and henceforth newly s which supplies a pseudostereophonic function that on average is optimum in relation to all algebraic invariants.

30 As the number of signal sections increases, the distribution of the intersection points ξ of the algebraic invariants on the half-plane respectively analyzed with the complex number plane approximates the

Gauss distribution. The smaller the chosen standard deviation σ is selected, the closer to ideal the resulting parameterization will be. However, as an only finite number of signal sections are available, σ
 5 should not be chosen too small.

Nevertheless, the method presented in FIG. 16D in terms of its convergence is considerably faster for sufficiently long signal sections than mentioned simulation models, since algebraic invariants are
 10 available for the first time to serve as valid "points of reference" for a weighting of already determined parameterizations.

If in an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or
 15 W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 only the special case $\varphi = 0$ (for EP1850639, the parameters in the figures below are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta =1$) or the special case $L_\alpha = L_\beta$ (for W02009/138205) or $L_A =$
 20 L_B (for EP1850639) are analyzed, this will yield the simplified circuits of the form of FIG. 17D resp. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10
 25 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D. In the case of EP1850639, the pseudostereophonic signal thus depends in terms of the
 30 time differences only on one constant $(\sqrt{5} - 1)/2$ resp. on s resp. in terms of the amplifications of constants resp. of the attenuation λ resp. ρ (see below), in the case of W02009/138205 it depends, in addition to newly on s , on $L_\alpha = L_\beta$ resp. in terms of the amplifications
 35 on P_α resp. P_β or also P_M' resp. P_β' or also P_M'' resp. P'_α (see W02009/138205) resp. of the attenuation λ resp.

ρ or the attenuation factors $1/\lambda$ or $1/\tau$ resp. λ' (see below).

For an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or
 5 W02011/009650 or PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also
 10 FIG. E2 or also the just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11
 15 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D, alternative systems to W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 for determining
 20 the optimum parameters φ resp. f (resp. n) resp. α resp. β resp. λ resp. ρ resp. s resp. the optimum time delay resp. combinations of these parameters can be obtained. If for example one sets $\varphi = 0$ and thus $\alpha = \beta$, and if furthermore one assumes $\lambda = \rho$, for given
 25 directional pattern f optimum values for α , β and λ can be determined as follows: define a weighting that ensures values for $0 \leq \lambda \leq 1$ as low as possible (since the virtualization of several microphones should appear natural) and ensure with the same weighting likewise
 30 for delay times that are as short as possible (in order to avoid artifacts). Both criteria are mutually exclusive: low values of λ foster great fictitious opening angles α and β ; and conversely short delay times correspond to small fictitious opening angles α
 35 and β .

Thus, a target correlation k , previously stipulated or chosen at will by the user, a weighting p previously stipulated or chosen at will by the user (for example $0 \leq p \leq 10$) can be introduced for the size
 5 of the fictitious opening angles α and β as well as a variable $g(\alpha)$ which overall balance this antagonistic behavior. Thanks to the high stability of the total system, the fictitious opening angles $\alpha = \beta$ can for example be analyzed in steps of 5° each, which for an
 10 algorithm implemented accordingly results in considerably reduced computation times.

$g(\alpha)$ can be defined on the basis of the time differences $L_\alpha = L_\beta$ (see above) for example as follows:

$$g(\alpha) := 2^{(-20 * (L_\alpha + L_\beta))}$$

15 A corresponding weighting function $h(\alpha)$ depending on α could then be stipulated for example as follows:

$$h(\alpha) := \lambda(\alpha)^p * g(\alpha)^{(10 - p)},$$

where $\lambda(\alpha)$ corresponds to the respective value
 20 determined for example according to the logic element 125 resp. to the feedback 126 of FIG. 1B and which supplies a stereo signal with the target correlation k for a specific fictitious opening angle $\alpha = \beta$.

For $p = 0$, it is exclusively $g(\alpha)$ that has an
 25 influence on the subsequent calculation of the optimum fictitious opening angle $\alpha_{\text{opt}} = \beta_{\text{opt}}$ determined on the basis of the weight p previously stipulated or chosen at will by the user; for $p = 10$, it is exclusively $\lambda(\alpha)$ that has the same influence.

30 The optimum opening angles α_{opt} and β_{opt} are then calculated in the present example according to the

following formula, wherein in the present practical example the integration is over the interval $[5^\circ; 90^\circ]$ or in the radiant over the interval $[\pi/36; \pi/2]$ as follows (wherein for practical considerations the
 5 integrals mentioned below can be determined as sums calculated in 5° steps):

$$\alpha_{\text{opt}} = \frac{\int_{\pi/36}^{\pi/2} \alpha * h(\alpha) d\alpha}{\int_{\pi/36}^{\pi/2} h(\alpha) d\alpha}$$

10 N.B. Due to the symmetry of α and β and because of the fact that φ is equal to 0, $h(\alpha)$ can be represented exclusively as a function of α .

Finally, for $\alpha_{\text{opt}} = \beta_{\text{opt}}$, which can also take on an intermediate value, the value $\lambda(\alpha_{\text{opt}})$ is again
 15 determined for example according to the logic element 125 resp. to the feedback 126 of FIG. 1B and which supplies a stereo signal with the exact target correlation k for $\alpha_{\text{opt}} = \beta_{\text{opt}}$.

The same principle (which allows a host of
 20 possible defined weightings and thus cannot be represented comprehensively) can be extended with L_α and L_β resp. with L'_α and L'_β also to the parameter s , which determines in the overall system the optimum spaciality, and furthermore to other parameters in
 25 connection with an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D
 30 or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or auch FIG. 21D resp. FIG. 23D or

simplified FIG. 24D or auch FIG. 25D resp. FIG. 27D or
 simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or
 simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or
 simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or
 5 simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp.
 FIG. 26D.

The interaction of the parameters f (resp. n)
 which describe the directional pattern of the signal
 that is to be rendered stereophonic, the angle φ - to
 10 be ascertained manually or by metrology - enclosed by
 the main axis and the sound source, the fictitious left
 opening angle α and the fictitious right opening angle
 β , the attenuations λ or else ρ for forming the
 resulting stereo signal in the case of W02009/138205
 15 resp. the angle φ enclosed by the main axis and the
 sound source as well as the attenuation λ or else ρ for
 forming the resulting stereo signal in the case of
 EP1850639 or also the parameters of the arrangements
 according to W02011/009649 or W02011/009650 or
 20 CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or
 also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG.
 E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp.
 FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A
 resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also
 25 FIG. E2 or also the just described arrangements of the
 form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D
 or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D
 or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D
 resp. FIG. E9 or also FIG. E10 or simplified FIG. E11
 30 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14
 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17
 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D and
 altogether of the time parameter s overall cause a
 spatial impression that on the one hand can be matched
 35 to the spatial impression (the acoustic parameters) of
 an existing stereophonic or pseudostereophonic signal,
 and on the other hand is to be stipulated with accuracy

precisely on the basis of the spatial impression (the acoustic parameters). For example, this spatial impression depends on the 1st resp. 2nd main reflection resp. on the diffuse sound. In particular the parameter
 5 s is given in this context a vital importance.

Inverse problems as well as their solutions are known from the theory of mathematical filters, in particular in connection with so-called wavelets. These are systems that despite the noise of a measuring
 10 system (for example of an optical camera or of a magnetic resonance system for generating images) manage to obtain a high-resolution signal. The resulting measured signal can be written as follows:

$$Y[q] = Uf[q] + W[q]$$

15 The operator U in this case contains the specific transfer function of the measuring system, f represents our high-resolution signal, W the noise of the measured signal, q the time. If the number of available measurements is considerably below the
 20 dimension n of the analyzed complex space (whose element is the high-resolution signal to be obtained), this is called an ill-posed inverse problem.

Main reflections can be derived from the base signal, which in our case represents the input signal
 25 to be stereophonized. Interestingly, the theory of inverse problems, with a few elements changed, can be transposed to the problem mentioned above of determining the spatial parameters of a stereophonic function; in this case (as the base signal is known),
 30 it is not a new ill-posed inverse problem and therefore knows unambiguous solutions.

The above equation is first reframed in the equation:

$$Y[q] = UY[q - t^*] + W[q] + D[q]$$

where $Y[q]$ represents the resulting stereo signal at the point in time q , $Y[q - t^*]$ the same stereo signal at the point in time $q - t^*$, $t^* \geq 0$, wherein t^* is the delay with which the 1st main reflection occurs, $W[q]$ is the signal without reverberation and $D[q]$ is the reverberation without the 1st main reflection, which can incidentally be easily estimated statistically. The operator U henceforth contains the specific transfer functions for the stereo signal $Y[q - t^*]$, so that the latter exhibits the acoustic properties of the 1st main reflection.

This decomposition proves optimal, since the listener judges the acoustic parameters first and foremost on the basis of the 1st main reflection.

Two application cases can now be distinguished:

The first case is an optimization problem wherein a pseudostereophonic signal $Y^*[q]$ is to be newly generated on the basis of the acoustic parameters of an already existing stereo signal $Y[q]$. In a first step, the specific t^* is sought that maximises our "high-resolution" signal, in fact the 1st main reflection of the signal $Y[q]$, and subsequently that parameterization of f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left opening angle α and the fictitious right opening angle β , as well as the attenuations λ or else ρ for forming the resulting stereo signal in the case of W02009/138205 resp. the angle φ enclosed by the main axis and the sound source as well as the attenuation λ or else ρ for forming the resulting stereo signal in the case of EP1850639 or

also the parameterization of the arrangements according to W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8
 5 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also the just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D
 10 resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D
 15 resp. FIG. 22D resp. FIG. 26D as overall the time coefficient s optimized according to the following consideration:

$$Y^*[q] = U^*Y^*[q - t^*] + W^*[q] + D^*[q]$$

represents our second unambiguously solvable
 20 inverse problem for the pseudostereophonic signal $Y^*[q]$ to be formed, wherein U^* resp. D^* is directly dependent functionally on said parameters f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be
 25 ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left opening angle α and the fictitious right opening angle β , as well as the attenuations λ or else ρ or the time parameter s for forming the resulting stereo signal in
 30 the case of W02009/138205 resp. the angle φ enclosed by the main axis and the sound source as well as the attenuation λ or else ρ or the time parameter s for forming the resulting stereo signal in the case of EP1850639 or also the parameters of the arrangements
 35 according to W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or

also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A bzw. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or auch
 5 FIG. E2 or also the just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11
 10 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D. If one now substitutes in the original equation

$$Y[q] = UY[q - t^*] + W[q] + D[q]$$

15 U by U^* resp. D by D^* , the sought optimized parameters parameters f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the
 20 sound source, the fictitious left opening angle α and the fictitious right opening angle β , as well as the attenuations λ or else ρ or the time parameter s for forming the resulting stereo signal in the case of W02009/138205 resp. the angle φ enclosed by the main
 25 axis and the sound source as well as the attenuation λ or else ρ or the time parameter s for forming the resulting stereo signal in the case of EP1850639 or also the parameters of the arrangements according to W02011/009649 or W02011/009650 or CH01264/10 resp.
 30 PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also
 35 just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D

bzw. FIG. 23D or simplified FIG. 24D or also FIG. 25D
 resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or
 also FIG. E10 or simplified FIG. E11 resp. FIG. E12
 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or
 5 also FIG. E16 or simplified FIG. E17 resp. FIG. 18D
 resp. FIG. 22D resp. FIG. 26D can be determined ideally
 or approximately on the basis of the following
 equation:

$$Y[q] - U^*Y[q - t^*] - W[q] - D^*[q] = 0.$$

10 In the second case, no stereophonic signal $Y[q]$
 is available for optimizing the pseudostereophonic
 signal $Y^*[q]$. Rather, the user is to be provided with a
 tool for directly optimizing the parameters f (resp. n)
 which describe the directional pattern of the signal
 15 that is to be rendered stereophonic, the angle φ - to
 be ascertained manually or by metrology - enclosed by
 the main axis and the sound source, the fictitious left
 opening angle α and the fictitious right opening angle
 β , as well as the attenuations λ or else ρ or the time
 20 parameter s for forming the resulting stereo signal in
 the case of W02009/138205 resp. the angle φ enclosed by
 the main axis and the sound source as well as the
 attenuation λ or else ρ or the time parameter s for
 forming the resulting stereo signal in the case of
 25 EP1850639 or also the parameters of the arrangements
 according to W02011/009649 or W02011/009650 or
 CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or
 also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG.
 E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp.
 30 FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A
 resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also
 FIG. E2 or also just described arrangements of the form
 of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or
 also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or
 35 also FIG. 25D resp. FIG. 27D or simplified FIG. 28D
 resp. FIG. E9 or also FIG. E10 or simplified FIG. E11

resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14
 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17
 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D, thus de
 facto to directly influence the spatial parameters of
 5 the pseudostereophonic mapping. This occurs by directly
 influencing the parameters t^* resp. U^* resp. D^* , that
 are optimized according to the following equation:

$$Y^*[q] - U^*Y^*[q - t^*] - W^*[q] - D^*[q] = 0$$

until an ideal or approximately satisfactory result is
 10 achieved. In particular, $W^*[q]$ can be expressed
 directly by the monophonic base signal to be rendered
 stereophonic.

If U^* resp. D^* are to be matched with an
 existing dictionary of available operators U bzw. D , we
 15 have again the described solvable first case of our
 optimization problem with the equation

$$Y^*[q] - UY^*[q - t^*] - W^*[q] - D[q] = 0$$

A further criterion for the first case of the
 optimization problem, wherein a pseudostereophonic
 20 signal $Y^*[q]$ is to be newly formed on the basis of the
 acoustic parameters of an already existing signal $Y[q]$,
 is supplied by the Spatial Audio Object Coding (SAOC)
 belonging to the state of the art. In this case, in
 addition to the spatial operators U, D bzw. U^*, D^* , the
 25 sinusoidal models of $Y^*[q]$ and $Y[q]$ are compared, i.e.
 those spectral components that are responsible within
 SAOC for localization. In particular the deviation of
 these sinusoidal models from one another can be
 quantized. Those parameters f (resp. n) which describe
 30 the directional pattern of the signal that is to be
 rendered stereophonic, the angle φ - to be ascertained
 manually or by metrology - enclosed by the main axis
 and the sound source, the fictitious left opening angle

α and the fictitious right opening angle β , as well as the attenuations λ or else ρ or the time parameter s for forming the resulting stereo signal in the case of W02009/138205 resp. the angle φ enclosed by the main
 5 axis and the sound source as well as the attenuation λ or else ρ or the time parameter s for forming the resulting stereo signal in the case of EP1850639 or also the parameters of the arrangements according to W02011/009649 or W02011/009650 or CH01264/10 resp.
 10 PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also
 15 just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D bzw. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11 resp. FIG. E12
 20 bzw. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D are then selected for which on the one hand this deviation is minimal and for which on the other hand the following equation

$$25 \quad Y[q] - U^*Y[q - t^*] - W[q] - D^*[q] = 0$$

is ideally or approximately solved.

In the prior art, so-called all-pass filters can be used for eliminating the mid sound sources amplified by 3dB in a mono signal as well as for
 30 varying a stereophonic or pseudostereophonic sound image, by being connected downstream to the left and/or right channel of a stereophonic or pseudostereophonic output signal. In principle, see for example US5671287 (Gerzon), the output signals of such all-pass filters
 35 can be combined to new stereophonic or

pseudostereophonic signals for example by means of addition or subtraction. Their use on stereophonic resp. pseudostereophonic signals with more than two channels is likewise possible. Such all-pass filters, which are described in literature as all-pass filters of first, second or n^{th} order and whose overall application on monophonic, stereophonic or pseudostereophonic signals is also state of the art, work outstandingly well with our systems described herein resp. cited herein. They can be used not only for post-processing of the stereophonic or pseudostereophonic signals obtained on the basis of own circuit schemata described herein resp. cited herein, but moreover enable their integration directly and in a versatile manner into the described or cited own circuit schemata and optimization processes, this being likewise according to the state of the art. The circuit principle of an all-pass filter of the first order is represented by way of example in FIG. 1F, the circuit principle of an all-pass filter of the second order is represented by way of example in FIG. 2F.

Similarly, phase regulators can be used that additionally enable the phase difference of the stereophonic or pseudostereophonic signal to be adjusted. Such phase regulators are also suitable not only for post-processing of the stereophonic or pseudostereophonic signals obtained on the basis of own circuit schemata described herein resp. cited herein, but moreover enable their integration directly and in a versatile manner into the described or cited own circuit schemata and optimization processes, this being likewise according to the state of the art. Their use on stereophonic resp. pseudostereophonic signals with more than two channels is likewise possible. The basic circuit principle of a phase regulator, here represented in a simplified manner for sinusoidal signals, is represented by way of example in FIG. 3F.

The simplest example of numerous possibilities
 is represented for example by FIG. 4F, wherein an all-
 pass filter is connected downstream for example in the
 left resp. in the right channel of a pseudostereophonic
 5 output signal of an inventive arrangement according to
 EP1850639 or W02009/138205 or W02011/009649 or
 W02011/009650 or PCT/EP2011/063322 resp. FIG. E3 or
 also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG.
 E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp.
 10 FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A
 resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also
 FIG. E2 or also just described arrangements of the form
 of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or
 also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or
 15 also FIG. 25D bzw. FIG. 27D or simplified FIG. 28D
 resp. FIG. E9 or also FIG. E10 or simplified FIG. E11
 resp. FIG. E12 bzw. FIG. E13 or simplified FIG. E14
 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17
 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D. Whilst
 20 the pseudostereophonic output signal of an inventive
 arrangement according to EP1850639 or W02009/138205 or
 W02011/009649 or W02011/009650 or CH01264/10 resp.
 PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also
 FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8
 25 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also
 FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D
 resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also
 just described arrangements of the form of FIG. 17D
 resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D
 30 resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D
 bzw. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or
 also FIG. E10 or simplified FIG. E11 resp. FIG. E12
 bzw. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or
 also FIG. E16 or simplified FIG. E17 resp. FIG. 18D
 35 resp. FIG. 22D resp. FIG. 26D exhibits a good emphasis
 of the mid sound sources, for example FIG. 4F provides
 for a corresponding dispersion with good source
 localization.

The simplest example of numerous possibilities for the use of phase shifters is represented for example by FIG. 5F, wherein a phase shifter is connected downstream for example in the left channel of the pseudostereophonic output signal of an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 resp. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D bzw. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11 resp. FIG. E12 bzw. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D and thus provides for an adjustable dispersion.

Short description of the figures

Various embodiments of the present invention are described hereinafter by way of example, with reference being made to the following drawings:

FIG.1A shows the circuit principle of a known panoramic potentiometer.

FIG.1B shows an example of a circuit for two logic elements for normalizing the level and for normalizing the degree of correlation of the output signals from a stereo decoder (for example a stereo decoder according to W02009/138205 or EP1850639), wherein the input signal M and S can (before passing through an

amplifier upstream to the MS matrix) optionally be fed to a circuit according to FIG. 7B, which is optionally also connected downstream to FIG.10B.

5 FIG.1C shows the apolarity constraint for the functions S , S' and Σ' .

FIG.1D shows a circuit according to EP1850639 which according to the invention has been extended by the parameter s .

10 FIG.1F shows the circuit schema of a prior art all-pass filter of first order.

FIG.2A shows the attenuation curve for the left and right channels of a panoramic potentiometer without an extended stereo width range and corresponding mapping angles.

15

FIG.2B shows an example of a circuit which maps given signals $x(t)$, $y(t)$, by using the transfer functions $f^*[x(t)]$ and $g^*[y(t)]$, on the complex number plane, resp. ascertains the argument of the sum thereof $f^*[x(t)] + g^*[y(t)]$.

20

FIG.2C shows the functions S , S' and Σ' for the Cartesian coordinate system $x_1 = u_1$, $x_2 = u_2$, $x_3 = u_3$ from the perspective of the first quadrant of the corresponding complex number plane.

25

FIG.2D shows a first circuit according to W02009/138205 resp. W02011/009649 which according to the invention has been extended by the parameter s .

FIG.2F shows the circuit schema of a prior art all-pass filter of second order.

5 FIG.3A shows a first embodiment of a device or of a method according to W02011/009649, wherein the left channel L' resp. right channel R' resulting from the stereo decoding are each fed to a panoramic potentiometer for collective buses L and R.

10 FIG.3B shows an example of a circuit for selecting the definition range by using the parameter a.

15 FIG.3C shows the functions S , S' and Σ' for the Cartesian coordinate system $x_1 = u_1$, $x_2 = u_2$, $x_3 = u_3$ likewise from the perspective of the first quadrant of the corresponding complex number plane.

20 FIG.3D shows a second circuit according to W02009/138205 resp. W02011/009649 which according to the invention has been extended by the parameter s.

FIG.3F shows the circuit schema of a prior art phase shifter.

25 FIG.4A shows a second embodiment of a device or of a method according to W02011/009649, wherein the left channel L' resp. the right channel R' resulting from the stereo decoding are each fed to a panoramic potentiometer for collective buses L and R.

30 FIG.4B shows an example of a circuit for a third logic element which checks the signals, which are generated in FIG. 1B and which are

mapped on the complex number plane as shown in
 FIG. 2B, in terms of the admissible definition
 range, defined in a new fashion according to
 FIG. 3B by the parameter a , according to the
 5 constraint $\text{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 +$
 $\text{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1$.

FIG.4C shows the functions S , S' and Σ' for
 the Cartesian coordinate system $x_1 = u_1$, $x_2 =$
 u_2 , $x_3 = u_3$ from the perspective of the fourth
 10 quadrant of the corresponding complex number
 plane.

FIG.4D shows a third circuit according to
 W02009/138205 resp. W02011/009649 which
 according to the invention has been extended by
 15 the parameter s .

FIG.4F shows a simple example of the
 downstream connection of an all-pass filter
 each in the left resp. in the right channel of
 a stereophonic or pseudostereophonic output
 20 signal of an inventive arrangement according to
 EP1850639 or W02009/138205 or W02011/009649 or
 W02011/009650 or CH01264/10 resp.
 PCT/EP2011/063322.

FIG.5A shows a third embodiment of a device
 25 or of a method according to W02011/009649,
 wherein the left channel L' resp. right channel
 R' resulting from the stereo decoding are each
 fed to a panoramic potentiometer for collective
 buses L and R .

FIG.5B shows an example of a circuit for a
 30 fourth logic element which finally analyzes the
 relief of the function $f^*[x(t)] + g^*[y(t)]$ for
 the purpose of maximizing the function values

thereof, wherein the user has a free choice of the limit value R^* defined by the inequality (8aB) (resp. by the deviation Δ likewise defined by the inequality (8aB)) for this maximization.

5

FIG.5C shows the convergence behavior of a weighting function, which here optimizes the parameters ϕ , f (resp. n), α , β for example on the basis of the mean values of the inter-section points in the 1st or also 3rd quadrants of three pseudostereophonic signal sections of the same length mapped on the complex number plane with the plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1).

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FIG.5D shows a first variant embodiment of the first circuit according to W02009/138205 resp. W02011/009649 which according to the invention has been extended by the parameter s .

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FIG.5F shows a simple example of the downstream connection of a phase shifter in the left channel of a stereophonic or pseudostereophonic output signal of an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322. The right input signal is retained without change as output signal. This results in a modified stereophonic or pseudostereophonic signal.

25

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FIG.6A shows a fourth embodiment of a device or of a method according to W02011/009649 with a circuit that is equivalent to FIG. 3A having a slightly modified MS matrix, which renders

superfluous a direct downstream connection of panoramic potentiometers.

5 FIG.6B shows an input circuit for an already existing stereo signal prior to transfer to a circuit as shown in FIG. 10B for determining the localization of the signal.

10 FIG.6C shows an example of the circuit described below for optimizing pseudostereophonic signals on the basis of algebraic invariants, which can be directly connected downstream of FIG. 5B, and with which it then constitutes in this example an inseparable unit. In this case, the outputs of FIG. 6C within the entire circuit schema are to be
15 treated in such a way as if they were those of FIG. 5B. The circuit of FIG. 6C has the effect that the elements connected upstream thereto are now passed through for different sections of audio signals. The result is an optimized
20 parameterization φ , f , α , β on the basis of the mean values of the intersection points in the 1st or also 3rd quadrant of these signal sections mapped on the complex number plane with the plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1).
25

FIG.6D shows a second variant embodiment of the first circuit according to W02009/138205 resp. W02011/009649 which according to the invention has been extended by the parameter s .

30 FIG.6F shows the principle of the coding of a five-channel signal according to Rec. ITU-R BS.775-1, which after being downmixed to the stereo 2/0 format is approximately identical to the (amplified) signal L , R , and after being

downmixed to the mono 1/0 format is approximately identical to the (amplified) signal M.

5 FIG.7A shows a circuit which is equivalent to FIG. 3A resp. FIG. 6A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 3A.

10 FIG.7B shows a further example of a circuit for normalizing stereophonic or pseudostereophonic signals which, when connected downstream to FIG. 10B, is activated as soon as the parameter z is present as an input signal. In this case, the initial value of the gain factor λ corresponds to the final value of the gain factor λ in FIG. 1B when the parameter z is transferred.

20 FIG.7C shows an example of a circuit which, on the basis of the determination of the *mean square energy* of the input signals $s_1(t_i)$, $s_2(t_i)$, ..., $s_8(t_i)$ and of the definable weightings G_1 , G_2 , ... G_8 , performs a normalization of these input signals and subsequently determines the invariants of a combination $\hat{f}(t)$ or of several combinations $\hat{f}_1(t)$, $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of these input signals.

30 FIG.7D shows a circuit equivalent to FIG. 4A resp. FIG. 13A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 4A, extended according to the invention by the parameter s .

FIG.7F shows the principle of the coding of a five-channel signal according to Rec. ITU-R BS.775-1, which after being downmixed to the stereo 2/0 format does not correspond to the (amplified) signal L, R, however, after being downmixed to the mono 1/0 format is exactly identical to the (amplified) signal M.

FIG.8A shows an enhanced circuit based on FIG. 7A for normalizing the level of the output signals from the stereo decoder.

FIG.8B shows an example of a circuit which maps given signals $x(t)$, $y(t)$ on the complex number plane by using the transfer functions $f^*[x(t)]$ and $g^*[y(t)]$.

FIG.8D shows a circuit equivalent to FIG. 5A resp. FIG. 14A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 5A, extended according to the invention by the parameter s .

FIG.8F shows the principle of the coding of a five-channel signal according to Rec. ITU-R BS.775-1, which after being downmixed to the stereo 2/0 format is approximately identical to the (amplified) signal L, R, and after being downmixed to the mono 1/0 format is approximately identical to the (amplified) signal $L + R$.

FIG.9A shows an example of a circuit which, as an enhancement to FIG. 8A, maps given signals $x(t)$, $y(t)$ as the sum of the transfer functions $f^*[x(t)] = [x(t)/\sqrt{\Sigma 2}] * (-1 + i)$ and

$g^*[y(t)] = [y(t)/\sqrt{\Sigma 2}] * (1 + i)$ on the complex number plane.

FIG.9B shows an example of a circuit for adjusting the image width of an audio signal.

5 FIG.9D shows a circuit equivalent to FIG. 7D, based on the equations (3AA) and (4AA) hereafter and which according to the invention has been extended by the parameter s .

10 FIG.9F shows the loudspeaker array for a five-channel signal according to Rec. ITU-R BS.775-1.

FIG.10A shows the example of a circuit that, as an enhancement to FIG. 9A, stipulates the image width of a stereo signal.

15 FIG10B shows a circuit for determining the localization of the signal, whose inputs are connected with the outputs of FIG. 5B resp. with the outputs of FIG. 6B.

20 FIG.10D shows a circuit equivalent to FIG. 7D, based on the equations (3AAA) and (4AAA) hereafter and which according to the invention has been extended by the parameter s .

25 FIG.10F shows the standardized mix ratio of a downmix for a five-channel signal according to Rec. ITU-R BS.775-1.

30 FIG.11A shows an example of an input circuit for an already existing stereo signal L° , R° prior to transfer to a circuit as shown in FIG. 12A (for determining the localization of the signal) which maps L° , i.e. $l(t)$, and R° , i.e.

$r(t)$, as the sum of the transfer functions
 $f*[l(t)] = [l(t)/\sqrt{2}] * (-1 + i)$ and $g*[r(t)] = [r(t)/\sqrt{2}] * (1 + i)$ on the complex number plane.

5 FIG.11D shows a circuit equivalent to FIG. 8D, based on the equations (3AA) and (4AA) hereafter and which according to the invention has been extended by the parameter s .

10 FIG.11F shows the total possible parameter combinations for the parameter f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis of the
 15 signal that is to be rendered stereophonic and the sound source, the fictitious left opening angle α and the fictitious right opening angle β .

20 FIG.12A shows a circuit for determining the localization of the signal, whose inputs can be connected to the outputs of FIG. 10A resp. to the outputs of FIG. 11A.

25 FIG.12D shows a circuit equivalent to FIG. 8D, based on the equations (3AAA) and (4AAA) hereafter and which according to the invention has been extended by the parameter s .

30 FIG.13A shows a fifth embodiment of a device or of a method according to W02011/009649 with a circuit that is equivalent to FIG. 4A having a slightly modified MS matrix, which renders superfluous a direct downstream connection of panoramic potentiometers.

FIG.13D shows the circuit according to FIG. 13A extended according to the invention by the parameter s .

5 FIG.14A shows a sixth embodiment of a device or of a method according to W02011/009649 with a circuit that is equivalent to FIG. 5A having a slightly modified MS matrix, which renders superfluous a direct downstream connection of panoramic potentiometers.

10 FIG.14D shows the circuit according to FIG. 14A extended according to the invention by the parameter s .

15 FIG.15D shows the example, henceforth extended according to the invention by the parameter s , of a circuit for two logic elements for normalizing the level and for normalizing the degree of correlation of the output signals from a stereo decoder (for example a stereo decoder according to W02009/138205 or
20 EP1850639), wherein the input signal M and S can (before passing through an amplifier upstream to the MS matrix) optionally be fed to a circuit according to FIG. 7B, which is optionally also connected downstream to FIG.
25 10B.

FIG.16D shows an example, henceforth extended according to the invention by the parameter s , of the circuit described below for optimizing pseudostereophonic signals on the basis of
30 algebraic invariants, which can be directly connected downstream of FIG. 5B, and with which it then constitutes in this example an inseparable unit. In this case, the outputs of FIG. 16D within the entire circuit schema are

to be treated in such a way as if they were those of FIG. 5B. The circuit of FIG. 16D has the effect that the elements connected upstream thereto are now passed through for different
 5 sections of audio signals. The result is an optimized parameterization φ , f , α , β , s on the basis of the mean values of the intersection points in the 1st or also 3rd quadrant of these signal sections mapped on the complex number
 10 plane with the plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1).

FIG.17D shows the first simplification, extended according to the invention by the
 15 parameter s , of a circuit according to W02009/138205 for uniform time differences $L'_\alpha = L'_\beta$.

FIG.18D shows a further simplification, extended according to the invention by the
 20 parameter s , of a circuit according to W02009/138205 for uniform time differences $L'_\alpha = L'_\beta$.

FIG.19D shows the simplified first variant embodiment, extended according to the invention
 25 by the parameter s , of the circuit of FIG. 6D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$.

FIG.20D shows the second variant embodiment, extended according to the invention by the
 30 parameter s and further simplified by integrating the parameter λ , of the circuit of FIG. 6D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$.

FIG.21D shows the second simplification,
 extended according to the invention by the
 parameter s , of the circuit according to
 W02009/138205 for uniform time differences L'_α
 5 $= L'_\beta$. For EP1850639, the parameters are to be
 set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin$
 $\alpha = \sin \beta = 1$.

FIG.22D shows a further simplification,
 extended according to the invention by the
 10 parameter s , of the circuit according to
 W02009/138205 for uniform time differences L'_α
 $= L'_\beta$. For EP1850639, the parameters are to be
 set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin$
 $\alpha = \sin \beta = 1$.

FIG.23D shows the simplified first variant
 embodiment, extended according to the invention
 by the parameter s , of the circuit of FIG. 3D
 according to W02009/138205 resp. W02011/009649
 for uniform time differences $L'_\alpha = L'_\beta$. For
 15 EP1850639, the parameters are to be set as
 follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha =$
 20 $\sin \beta = 1$.

FIG.24D shows the second variant embodiment,
 extended according to the invention by the
 25 parameter s and further simplified by
 integrating the parameter λ , of the circuit of
 FIG. 3D according to W02009/138205 resp.
 W02011/009649 for uniform time differences L'_α
 $= L'_\beta$. For EP1850639, the parameters are to be
 set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin$
 30 $\alpha = \sin \beta = 1$.

FIG.25D shows the third simplification,
 extended according to the invention by the
 parameter s , of a circuit according to

W02009/138205 for uniform time differences L'_α
 $= L'_\beta$.

FIG.26D shows a further simplification,
 extended according to the invention by the
 5 parameter s , of a circuit according to
 W02009/138205 for uniform time differences L'_α
 $= L'_\beta$.

FIG.27D shows the simplified first variant
 embodiment, extended according to the invention
 10 by the parameter s , of the circuit of FIG. 4D
 according to W02009/138205 resp. W02011/009649
 for uniform time differences $L'_\alpha = L'_\beta$. For
 EP1850639, the parameters are to be set as
 follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha =$
 15 $\sin \beta = 1$.

FIG.28D shows the second variant embodiment,
 extended according to the invention by the
 parameter s and further simplified by
 integrating the parameter λ , of the circuit of
 20 FIG. 4D according to W02009/138205 resp.
 W02011/009649 for uniform time differences L'_α
 $= L'_\beta$. For EP1850639, the parameters are to be
 set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin$
 $\alpha = \sin \beta = 1$.

FIG.E1 shows a circuit which is equivalent to
 25 FIG. 3A resp. FIG. 6A, provided that the
 relation $\lambda = \rho$ applies for the inversely
 proportional attenuations λ and ρ of the
 panoramic potentiometers shown in FIG. 3A.

FIG.E2 shows a circuit which is equivalent to
 30 FIG. 3A resp. FIG. 6A, provided that the
 relation $\lambda = \rho$ applies for the inversely

proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 3A.

FIG.E3 shows a circuit which is equivalent to FIG. 4A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 4A.

FIG.E4 shows a circuit equivalent to FIG. 4A according to the equations (3AA) and (4AA), provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 4A.

FIG.E5 shows a circuit equivalent to FIG. 4A according to the equations (3AAA) and (4AAA), provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 4A.

FIG.E6 shows a circuit equivalent to FIG. 5A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 5A.

FIG.E7 shows a circuit equivalent to FIG. 5A according to the equations (3AA) and (4AA), provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 5A.

FIG.E8 shows a circuit equivalent to FIG. 5A according to the equations (3AAA) and (4AAA),

provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 5A.

5 FIG.E9 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 6D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$.

10 FIG.E10 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 6D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$.

15 FIG.E11 shows a simplified second variant embodiment, extended according to the invention by the parameter s and further simplified by integrating the parameter λ' , of the circuit of FIG. E10 according to W02009/138205 resp.
20 W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$.

FIG.E12 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 3D
25 according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi) = f(\alpha) = f(\beta) = 1$, $\sin \varphi = 0$, $\sin \alpha = \sin \beta = 1$.

30 FIG.E13 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 3D according to W02009/138205 resp. W02011/009649

for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta = 1$.

5 FIG.E14 shows a new variant embodiment, extended according to the invention by the parameter s and further simplified by integrating the parameter λ' , of the circuit of FIG. 3D according to W02009/138205 resp.

10 W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta = 1$.

15 FIG.E15 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 4D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta = 1$.

20

25 FIG.E16 shows a new simplified variant embodiment, extended according to the invention by the parameter s , of the circuit of FIG. 4D according to W02009/138205 resp. W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta = 1$.

30 FIG.E17 shows a new variant embodiment, extended according to the invention by the parameter s and further simplified by integrating the parameter λ' , of the circuit of FIG. 4D according to W02009/138205 resp.

W02011/009649 for uniform time differences $L'_\alpha = L'_\beta$. For EP1850639, the parameters are to be set as follows: $f(\varphi)=f(\alpha) = f(\beta)=1$, $\sin \varphi=0$, $\sin \alpha = \sin \beta =1$.

5 Detailed description

It is general knowledge that audio signals that are emitted via two or more loudspeakers provide the listener with a spatial impression, provided that they show different amplitudes, frequencies, time resp.
10 phase differences or are reverberated appropriately

Such decorrelated signals can firstly be generated by differently positioned sound transducer systems, the signals from which are optionally post-processed, or can be generated by means of so-called
15 pseudostereophonic techniques, which – on the basis of a mono signal – produce such suitable decorrelation.

Hereinafter, the contents of W02011/009649 will be reproduced in full for a better understanding of the following examples of application of the present
20 invention:

Some pseudostereophonic signals show increased “phasiness”, that is to say distinctly perceptible time differences between both channels. Frequently, the degree of correlation between both channels also is too
25 low (lack of compatibility) or too high (undesirable convergence towards a mono sound). Pseudostereophonic signals, but also stereophonic signals, may therefore show deficiencies due to lacking or excessive decorrelations between the emitted signals.

30 It is thus an aim of W02011/009649 to solve this problem and to align or inversely to differentiate

more strongly stereophonic (including pseudostereophonic) signals.

It is another aim to improve, generate, transmit, convert and reproduce stereophonic and
5 pseudostereophonic audio signals.

In W02011/009649, these problems are solved inter alia by means of the ostensibly unprofitable downstream connection of a panoramic potentiometer in a device for pseudostereo conversion.

10 Panoramic potentiometers (also called pan pots or panoramic controls or panoramic dials) are known per se and are used for intensity stereophonic signals, i.e. for stereo signals that differ exclusively in terms of their levels but not in terms of timing resp.
15 phase differences or different frequency spectra. The circuit principle of a known panoramic potentiometer is represented in Figure 1A. The device has an input 101 and two outputs 202, 203 which are connected to the buses 204, 205 for the group channels L (left audio
20 channel) and R (right audio channel). In the center position (M), both buses receive the same level; in the side positions to the left (L) and to the right (R), the signal is routed only to the left bus or to the right bus, respectively. In the intermediate positions,
25 a panoramic potentiometer produces level differences corresponding to the different positions of the phantom source on the loudspeaker base.

FIG. 2A shows the attenuation curve for the left channel and the right channel of a panoramic
30 potentiometer without an extended stereo width range, and corresponding mapping angles. In the center position, the attenuation in each channel is 3 dB, the acoustic convolution thereby producing the same

perception of volume as would be if there were only one channel in the L or R position.

Panoramic potentiometers are able as voltage dividers for example to distribute the left channel in
5 a different, selectable ratio to the resulting left output resp. right output (these outputs are also called buses) or, in the same way, to distribute the right channel in a different, selectable ratio to the same left output resp. right output (the same buses).
10 Therefore, in the case of intensity stereophonic signals, the image width can be narrowed and the direction of such signals can be shifted.

In the case of pseudostereophonic signals, which make use of timing resp. phase differences,
15 different frequency spectra or reverberation (and also in the case of stereo signals of such kind in general), such narrowing of the image width resp. shifting of the localisation direction are not possible by using a panoramic potentiometer. The application of panoramic
20 potentiometers to such signals is therefore deliberately abstained from.

However, as represented in W02011/009649 it has been observed that the previously unknown downstream connection of a panoramic potentiometer downstream to a
25 circuit for pseudostereo conversion affords unexpected advantages. Although such downstream connection cannot result in the aforementioned narrowing of the image width or in the shifting of the localisation direction of the stereo signals obtained, the degree of
30 correlation between the left signal and the right signal can however be increased or also decreased in this way by using such a panoramic potentiometer.

In a preferred embodiment, a panoramic potentiometer is connected downstream respectively to

the left output and to the right output of the circuit for obtaining a pseudostereophonic signal. In this case, the buses of both panoramic potentiometers are preferably used collectively and preferably
5 identically.

In this arrangement, each panoramic potentiometer has an input and two outputs. The input of a first panoramic potentiometer is connected to a first output of the circuit, and the input of a second
10 panoramic potentiometer is connected to a second output of this circuit. The first output of the first panoramic potentiometer is connected to the first output of the second panoramic potentiometer. The second output of the first panoramic potentiometer is
15 connected to the second output of the second panoramic potentiometer.

Alternatively and equivalently, rather than using panoramic potentiometers, the degree of correlation can also be adjusted by using a first
20 circuit for pseudostereo decoding with a stereo decoder and an amplifier connected upstream of the stereo decoder for amplifying an input signal of the stereo decoder, this being achieved without panoramic potentiometer. An equivalent adjustment of the degree
25 of correlation can therefore be implemented with fewer components.

Alternatively and equivalently, rather than using a panoramic potentiometer, the degree of correlation can also be varied by using a second
30 circuit, this being achieved with a modified stereo decoder which contains an adder and a subtractor in order to add respectively subtract input signals (M, S), which are respectively amplified by predetermined factors, in order to generate signals which are
35 identical to the bus signals from the panoramic

potentiometers. An equivalent adjustment of the degree of correlation can therefore be implemented with even fewer components.

5 The invention can also be applied to devices or methods that generate signals which are reproduced by more than two loudspeakers (for example surround sound systems belonging to the prior art).

Figures 3A to 5A show various embodiments of the circuit principle described above, in which a
 10 panoramic potentiometer 311 and 312, 411 and 412, 511 and 512, respectively, is connected directly downstream to a pseudo conversion circuit 309, 409 resp. 509. In each example shown here, the pseudo conversion circuit 309, 409 resp. 509 comprises a circuit having an MS
 15 matrix 310, 410 resp. 510, as described in W02009/138205 as well as in EP1850639.

This panoramic potentiometer 311 and 312, 411 and 412, 511 and 512 can be used to increase or decrease the degree of correlation of the resulting
 20 buses L 304, 404, 504 and R 305, 405, 505. Accordingly, the left channel L' 302, 402, 502 and the right channel R' 303, 403, 503 resulting from the stereo decoding (after passing through the MS matrix) are fed each to a panoramic potentiometer for collectively used buses L
 25 and R.

If the attenuation λ for the left input signal L' of the panoramic potentiometer 311, 411 or 511 and the attenuation ρ for the right input signal R' of the panoramic potentiometer 312, 412, 512 for a stereo
 30 signal 302 and 303, 402 and 403, 502 and 503 resulting from a device 309, 409 or 509 is limited to the range between 0 and 3 dB, the inversely proportional relations

$$1 \geq \lambda \geq 0$$

and

$$1 \geq \rho \geq 0$$

may be introduced (where 1 corresponds to the value 0
5 dB and 0 corresponds to the value 3 dB).

λ and ρ therefore correspond to the inversely proportional attenuations of the panoramic potentiometers shown in FIG. 3A to FIG. 5A, limited to the range between 0 and 3 dB.

10 Therefore, the following relations are obtained for the resulting stereo signals (buses) L and R (304 and 305, 404 and 405, 504 and 505) resp. the output signals L'' 313, 413, 513 and R'' 314, 414, 514 from the panoramic potentiometer 311, 411, 511 and the
15 output signals L''' 315, 415, 515 and R''' 316, 416, 516 from the panoramic potentiometer 312, 412, 512:

$$(1A) \quad L = L'' + L''' = \frac{1}{2} * L' (1 + \lambda) + \frac{1}{2} * R' (1 - \rho)$$

and

$$(2A) \quad R = R'' + R''' = \frac{1}{2} * L' (1 - \lambda) + \frac{1}{2} * R' (1 + \rho)$$

20 Figure 6A shows a further embodiment with a circuit equivalent to FIG. 3A having a slightly modified MS matrix, which renders direct downstream connection of panoramic potentiometers superfluous. Taking into account the equivalences of the stereo
25 decoding (MS matrixing)

$$L' = (M + S) * 1/\sqrt{2}$$

and

$$R' = (M - S) * 1/\sqrt{2}$$

the following relations are obtained:

$$(1A) \quad L = [M (2 + \lambda - \rho) + S (\lambda + \rho)] * 1/2\sqrt{2}$$

$$(2A) \quad R = [M (2 - \lambda + \rho) - S (\lambda + \rho)] * 1/2\sqrt{2}$$

5 This allows the signals on the buses L and R to be also derived directly from the input signals M and S of the stereo decoding circuit.

 If $\lambda = \rho$ (same attenuation in the left channel and in the right channel), the following relations
10 apply:

$$(3A) \quad L = (M + \lambda * S) * 1/\sqrt{2}$$

$$(4A) \quad R = (M - \lambda * S) * 1/\sqrt{2}$$

 i.e. the variation in the amplitude of the signal S is equivalent to the downstream connection of
15 a respective panoramic potentiometer for identical attenuation in the left channel and in the right channel. Under these assumptions, the output signals L and R correspond to the bus signals L and R in FIG. 3A.

 This will therefore yield a circuit or a method
20 showing for example the form in FIG. 6A (trivial modifications being possible), which forms a composite signal from the M signal, amplified by the factor $(2 + \lambda - \rho)$, and the S signal, amplified by the factor $(\lambda + \rho)$, as well as a difference signal which is compiled
25 from the M signal, amplified by the factor $(2 - \lambda + \rho)$, minus the S signal, amplified by the factor $(\lambda + \rho)$, with correction by the factor $1/2\sqrt{2}$ needing to be performed overall in order to obtain signals L and R equivalent to the formulae (1A) and (2B).

FIG. 7A shows a circuit equivalent to FIG. 3A resp. FIG. 6A, provided that the relation $\lambda = \rho$ applies for the inversely proportional attenuations λ and ρ of the panoramic potentiometers shown in FIG. 3A. This
 5 circuit should not be confused with the arrangement known from intensity stereophony (MS microphone technique) for altering the pickup or aperture angle (which does not take place here!).

In this case, it is assumed that uniform
 10 attenuation for proposed panoramic potentiometers or modified MS matrix, as just illustrated, is frequently sufficient for levelling or differentiating stereo signals. When $\lambda = \rho$, the device just illustrated is then simplified on the basis of the above formulae (3A)
 15 and (4A) according to:

$$(3A) \quad L = (M + \lambda * S) * 1/\sqrt{2}$$

$$(4A) \quad R = (M - \lambda * S) * 1/\sqrt{2}$$

which is equivalent to a simple amplitude correction of the S signal (717).

20 Such an amplitude correction for the S signal has been known to date only for the classical MS microphone technique and in the ideal range results in the alteration of the pickup or aperture angle in that case, which does not take place here. A transfer of the
 25 same operating principle is not possible (and an application of the MS microphone technique to the present circuit is accordingly not obvious).

In FIG. 7A, the S signal is therefore additionally amplified by the factor λ ($1 \geq \lambda \geq 0$)
 30 prior to finally passing through the MS matrix. The resulting stereo signal is equivalent to the bus signals 304 and 305 in FIG. 3A, 404 and 405 in FIG. 4A

and 504 and 505 in FIG. 5A for uniform attenuation and also to the output signal L and R in FIG. 6A, provided that $\lambda = \rho$ is true in that case.

In practice, this circuit resp. method can be
 5 used to exactly stipulate the degree of correlation, i.e. there is a direct functional relationship between the attenuation λ and the degree of correlation r , for which ideally

$$0,2 \leq r \leq 0,7$$

10 is true. For λ , a series of experiments has found

$$0,07 \leq \lambda \leq 0,46$$

to be favorable for most applications.

In particular, artifacts (such as disturbing
 15 timing differences, phase shifts, or the like) can be eliminated without difficulty by using this device or method, whether manually or automatically (algorithmically).

On the basis of the equivalence of downstream
 20 panoramic potentiometers with uniform attenuation and amplitude correction of the S signal by the factor λ ($1 \geq \lambda \geq 0$) prior to final MS matrixing, it is therefore possible to achieve convincing pseudostereophony which, on the basis of the original mono signal, grants the
 25 listener a comprehensive, albeit extremely simple, post-processing option, while fundamentally maintaining the compatibility and avoiding disturbing artifacts.

Alternatively, the M signal can also be amplified by the factor $1/\lambda$. The equations (3A) and
 30 (4A) are then to be replaced by the equations

$$(3AA) \quad (1/\lambda) * L = [(1/\lambda) * M + S] * 1/\sqrt{2}$$

resp.

$$(4AA) \quad (1/\lambda) * R = [(1/\lambda) * M - S] * 1/\sqrt{2}$$

in order to obtain signals equivalent to (3A) resp.

5 (4A). FIG. 7A in this case is to be replaced by FIG. E1, FIG. E3 described hereafter is to be replaced by FIG. E4, FIG. E6 described hereafter is to be replaced by FIG. E7.

10 If alternatively both the M signal as well as the S signal are amplified, the relations $1 \geq \tau + \lambda' \geq 0$ and $\lambda = \tau + \lambda'$ result for the new amplification factor $1/\tau$ for the M signal and for the new amplification factor λ' for the S signal. The equations (3A) and (4A) are then to be replaced by the equations

$$15 \quad (3AAA) \quad (1/\tau) * L = [(1/\tau) * M + \lambda' * S] * 1/\sqrt{2}$$

resp.

$$(4AAA) \quad (1/\tau) * R = [(1/\tau) * M - \lambda' * S] * 1/\sqrt{2}$$

in order to obtain signals equivalent to (3A) resp.

(4A). FIG. 7A in this case is to be replaced by FIG. E2, FIG. E3 described hereafter is to be replaced by FIG. E5, FIG. E6 described hereafter is to be replaced by FIG. E8. Likewise, the new amplification factors $1/\lambda$ resp. $1/\tau$ resp. λ' can be used in a similar way as the inversely proportional attenuations λ or ρ for
 25 optimizing pseudostereophonic signals according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 or also according to the present application.

Overall, these devices and methods or systems can be used for example in telephony, in the field of professional post-processing of audio signals or else in the area of high-quality electronic consumer goods, the aim of which is the most simple yet efficient
5 handling.

For narrowing or expanding the image width:

For this application, the additional use of prior art compression algorithms or data reduction
10 methods or the analysis of characteristic features, such as the minima or maxima for the pseudostereophonic signals obtained is recommended in order to speed up evaluation thereof in accordance with the invention.

Of particular interest (for example for
15 reproducing stereophonic signals in automobiles) is the subsequent narrowing or expanding of the image width of the stereo signal obtained by using the specific variation of the degree of correlation r of the resulting stereo signal resp. the attenuations λ or
20 else ρ (for forming the resulting stereo signal). The previously determined parameters f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the
25 main axis and the sound source, the fictitious left opening angle α and the fictitious right opening angle β can be retained in this case, and it makes sense that now only a final amplitude correction is necessary, for example as per the logic element 120 in FIG. 8A,
30 provided that this narrowing or expanding of the image width is performed manually.

If this is to be automated, series of psychoacoustic experiments show that a constant image

width for stereophonic output signals $x(t)$, $y(t)$ resp. complex transfer functions thereof

$$(5A) \quad f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

$$(6A) \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

5 is essentially dependent on the criterion

$$(7A) \quad 0 \leq S^* - \varepsilon \leq \max |\operatorname{Re} \{ f^*[x(t)] + g^*[y(t)] \}| \\ \leq S^* + \varepsilon \leq 1$$

and also on the criterion

$$(8A) \quad 0 \leq U^* - \kappa \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt \leq U^* + \kappa$$

(where S^* and ε or, respectively, U^* and κ need to be stipulated differently for telephone signals, for example, than for music recordings). Accordingly, it is
15 now necessary to determine only suitable function values $x(t)$, $y(t)$ which are dependent on the degree of correlation r of the resulting stereo signal respectively on the attenuations λ or else ρ (for the generation of the resulting stereo signal) resp. on a
20 logic element 120 in FIG. 8A, in accordance with an iterative operating principle which is based on feedback.

The represented arrangement can accordingly be enhanced as follows within the context of an
25 arrangement for example in the form shown in FIG. 8A to 10A:

An output signal resulting from an arrangement as shown in FIG. 1A to 7A is in this case amplified

uniformly by a factor ρ^* (amplifiers 118, 119 in FIG. 8) such that the maximum of both signals has a level of exactly 0 dB (normalization on the unit circle of the complex number plane). By way of example, this is
 5 achieved by the downstream connection of a logic element 120 which varies or corrects the gain factor ρ^* of the amplifiers 118 and 119 via the feedback loops 121 and 122 until the maximum level for the left channel resp. for the right channel is 0 dB.

10 In a further step, the resulting signals $x(t)$ (123) and $y(t)$ (124) are now fed to a matrix in which, following respective amplification by the factor $1/\sqrt{2}$ (amplifiers 229, 230 in FIG. 9A), they are split into a respective identical real and imaginary part, with the
 15 real part formed from the signal $x(t)$, amplified by means of 229, additionally passing through the amplifier 231 with the gain factor -1. This therefore yields the transfer functions

$$(5A) \quad f^*[x(t)] = [x(t) / \sqrt{2}] * (-1 + i)$$

20 and

$$(6A) \quad g^*[y(t)] = [y(t) / \sqrt{2}] * (1 + i) .$$

The respective real and imaginary parts are now summed and therefore produce the real part resp. the imaginary part of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$.
 25

An arrangement, for example based on the logic element 640 in FIG. 10A, now needs to be connected downstream, which arrangement checks, for a limit value S^* - suitably selected by the user with respect to the
 30 image width of the stereo signal that is to be achieved - or a suitably selected deviation ε - both defined by the inequality (7A) - whether the condition

$$(7A) \quad 0 \leq S^* - \varepsilon \leq \max |\operatorname{Re} \{f^*[x(t)] + g^*[y(t)]\}| \leq S^* + \varepsilon \leq 1$$

is met. If this is not the case, a feedback loop 641 is used to determine a new optimized value for the degree of correlation r resp. for the attenuations λ or else ρ (for the generation of the resulting stereo signal), and the previous steps just described, as illustrated in FIG. 8A to 10A, are performed until the above condition (7A) is fulfilled.

The input signals for the logic element 640 are now transferred to an arrangement, for example based on the logic element 642 in FIG. 10A. Such arrangement finally analyzes the relief of the function $f^*[x(t)] + g^*[y(t)]$ for the purpose of optimizing the function values in terms of the image width of the stereo signal that is to be achieved, the user being able to suitably select the limit value U^* as well as the deviation κ , both defined by the inequality (8A), with respect to the image width of the stereo signal that is to be achieved. Overall, the condition

$$(8A) \quad 0 \leq U^* - \kappa \leq \int |f^*[x(t)] + g^*[y(t)]| dt \leq U^* + \kappa$$

must be met. If this is not the case, a feedback loop 643 is used to determine a new optimized value for the degree of correlation r resp. for the attenuations λ or else ρ (for the generation of the resulting stereo signal), and the previous steps just described, as illustrated in FIG. 8A to 10A, are performed until the relief of the function $f^*[x(t)] + g^*[y(t)]$ satisfies the desired optimization of the function values with respect to the image width taking into account the limit value U^* and the deviation κ , both suitably chosen by the user.

In terms of the image width – determined by the degree of correlation r resp. the attenuations λ or else ρ (for the generation of the resulting stereo signal) – the signals $x(t)$ (123) and $y(t)$ (124) therefore correspond to the specifications by the user and represent the output signals L^{**} and R^{**} from the arrangement which has just been described.

The present considerations remain valid as an entity even if a different reference system than the unit circle of the imaginary plane is chosen. By way of example, instead of normalizing function values, it is also possible to normalize the axis length in order to reduce the computation complexity accordingly.

Stipulation of the localisation direction:

Occasionally, it is also important to mirror the stereophonic mapping obtained about the main axis of the directional pattern on which the stereophonic rendition is based, since, for instance, mirror inverted mapping in relation to the main axis occurs. This can be achieved manually by swapping the left channel and the right channel.

If an already existing stereo signal L° , R° is to be mapped by the present system, the correct localisation direction can also be ascertained automatically by means of the phantom sources generated using the illustrated pseudostereophonic methodology, as is shown by way of example in FIG. 12A (which is directly connected downstream with FIG. 10A, whereas FIG. 11A may likewise be added to FIG. 12A for determining the sum of the complex transfer functions $f^*(l(t_i)) + g^*(r(t_i))$ for the already existing stereo signal L° , R° ; cf. the explanations relating to FIG. 9A). In this case, at suitably chosen times t_i (for which not all of the subsequently cited correlating

function values of the transfer functions $f^*(x(t_i)) + g^*(y(t_i))$ resp. $f^*(l(t_i)) + g^*(r(t_i))$ may be equal to zero for at least one case), the already ascertained transfer function $f^*(x(t_i)) + g^*(y(t_i))$ as shown in FIG. 5 9A is compared with the transfer function $f^*(l(t_i)) + g^*(r(t_i))$ of the left signal $l(t)$ resp. right signal $r(t)$ of the original stereo signal L°, R° . If these transfer functions range in the same quadrant or in the diagonally opposite quadrant of the complex number 10 plane, the total number m of function values from said transfer functions which are located in the same quadrant or in the diagonally opposite quadrant of the complex number plane is increased by 1 in each case.

An empirically (or statistically determined) 15 specifiable number b , which should be less than or equal to the number of correlating function values of the transfer functions $f^*(x(t_i)) + g^*(y(t_i))$ resp. $f^*(l(t_i)) + g^*(r(t_i))$ unequal to zero, now stipulates the number of necessary matches. Below this number, the 20 left channel $x(t)$ and the right channel $y(t)$ of the stereo signal resulting, for example, from an arrangement as shown in FIG. 8A - 10A are swapped.

If an originally stereophonic signal is to be encoded into a mono signal plus the function f 25 describing the directional pattern (resp. the simplifying parameter n of said function) and likewise plus the parameters $\varphi, \alpha, \beta, \lambda$ or ρ (for example for the purpose of data compression) (for an exemplary output 640a which may be extended by the parameter z , 30 see below), it makes sense to jointly encode the information regarding whether the resulting left channel and the resulting right channel need to be swapped (for example expressed by the parameter z , which takes the value 0 or 1).

With slight modifications, similar circuits to the circuits shown in FIGURES 11A and 12A can be constructed which can be connected directly downstream with FIGURE 3A or 4A or 5A or 6A or 7A or else can be
 5 used at another location within the electrical circuit or algorithm.

For obtaining stable FM stereo signals by using W02011/009649 by way of example for the evaluation of an existing stereo signal which can be reproduced by
 10 two or more loudspeakers:

W02011/009649 is also of particular importance in connection with obtaining stable FM stereo signals under bad reception conditions (for example in automobiles). In this case, it is possible to achieve
 15 stable stereophony by simply using the main channel signal ($L + R$) as an input signal, which is the sum of the left and right channel of the original stereo signal. The complete or incomplete sub-channel signal ($L - R$), which is the result of subtracting the right
 20 channel from the left channel of the original stereo signal, can also be used in this case in order to form a useable S signal or in order to determine or optimize the parameters f (resp. n), which describe the directional pattern of the signal that is to be
 25 rendered stereophonic as well as the angle φ that is to be ascertained manually or by metrology and is enclosed by the main axis and the sound source, the fictitious left opening angle α , the fictitious right opening angle β , the attenuations λ or else ρ for the
 30 generation of the resulting stereo signal or, resulting therefrom, the gain factor ρ^* for normalizing the left channel and the right channel, resulting from the MS matrixing (for example determined in a similar fashion to the logic element 120 as shown in FIG. 8A) or from
 35 another arrangement according to the invention on the unit circle (in this case 1 corresponds to the maximum

level of 0 dB which has been normalized by using ρ^* ,
 where $x(t)$ is the left output signal resulting from
 this normalization and $y(t)$ is the right output signal
 resulting from this normalization) or the degree of
 5 correlation r of the resulting stereo signal or the
 parameter a , for example defined by the inequality
 (9aA) below for defining the admissible range of values
 for the sum of the transfer functions of the resulting
 output signals (for example said complex transfer
 10 functions

$$(5A) \quad f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

and

$$(6A) \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

where, for $0 \leq a \leq 1$ for example the following is true:

$$15 \quad (9aA) \quad \operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1)$$

or the limit value R^* , defined by the inequality (11aA)
 below, or the deviation Δ , likewise defined by the
 inequality (11aA) below for stipulating or maximizing
 20 the absolute value of the function values of the sum of
 these transfer functions (where, for this stipulation
 or maximization and for the time interval $[-T, T]$ resp.
 the total number of possible output signals $x_j(t)$,
 $y_j(t)$, the following for example is true:

$$25 \quad (11aA) \quad 0 \leq R^* - \Delta \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt$$

$$\leq \max \int_{-T}^T |f^*[x_j(t)] \{f^*[x_j(t)], g^*[y_j(t)]\}| dt \in \Phi$$

30

$$\begin{aligned}
& + g^*[y_j(t)] \big| dt \\
& \leq R^* + \Delta \\
& \int_{-T}^T a^* \{ 1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{ f^*[x(t)] \\
5 \quad & + g^*[y(t)] \}} \} dt)
\end{aligned}$$

or the limit value S^* defined above or the deviation ε defined above (for which, by way of example, it must be true that

$$\begin{aligned}
10 \quad (7A) \quad 0 \leq S^* - \varepsilon \leq \max | \operatorname{Re} \{ f^*[x(t)] + g^*[y(t)] \} | \\
\leq S^* + \varepsilon \leq 1)
\end{aligned}$$

or the limit value U^* defined above or the deviation κ defined above (for which, by way of example, it must be true that

$$\begin{aligned}
15 \quad (8A) \quad 0 \leq U^* - \kappa \leq \int_{-T}^T | f^*[x(t)] + g^*[y(t)] | dt \leq U^* + \kappa,
\end{aligned}$$

all for determining the image width of the stereo signal to be achieved, or the localisation direction of the reproduced sound sources in accordance with the arrangement described above. In any case, the result is stereophonic mapping which is constant in respect of the FM signal.

In particular, the use of compression algorithms or data reduction methods which belong to the prior art resp. the analysis of characteristic features, such as the minima or maxima, is also recommended in this case in order to speed up the

evaluation of stereophonic or pseudostereophonic signals according to the criteria described above.

Hereinafter, the contents of W02011/009650 will
5 be reproduced in full for a better understanding of the following examples of application of the present invention:

In the case of the configuration according to W02009/138205, according to EP1850639 and/or according
10 to W02011/009649, different parameters may be chosen in the stereo decoder and which are used to generate pseudostereophonic signals. Though often several parameters or several sets of parameters may be used in order to obtain pseudostereophonic audio signals, the
15 choice of such parameters has an impact on the perceived spatial sound image. The choice of the parameters which are optimum in a certain condition or for a particular audio signal is however not trivial.

Furthermore, the adjustment of the parameters
20 also frequently has an impact on the degree of correlation between the left channel and the right channel. In the context of W02011/009650, however, it has been found that it would make sense to stipulate a uniform degree of correlation for the evaluation of
25 different parameterizations for φ resp. f (resp. the simplifying parameter n), α , β .

An aim therein is to provide a new method and a new device for obtaining pseudostereophonic signals resp. a new method and a new device for automatically
30 and optimally choosing such parameters which form the basis for the generation of stereophonic or pseudostereophonic signals, resp. a method and a device for optimally and automatically determining

particularly the parameters (ϕ , λ , ρ resp. f (resp. n), α , β) while generating said stereophonic or pseudostereophonic signals.

Such a method resp. such a device are intended
 5 to be used to select, from a plurality of decorrelated, in particular pseudostereophonic, signal variants, those whose decorrelation is found to be particularly advantageous.

In particular, it should be possible to
 10 influence the selection criteria themselves in a form as efficient and compact as possible in order to be able to convert signals of different nature (for example speech in contrast to music recordings) into the optimized reproduction thereof.

According to one aspect, W02011/009650 proposes
 15 a device and a method for obtaining pseudostereophonic output signals $x(t)$ and $y(t)$ by using a stereo decoder, wherein $x(t)$ is the function value of the resulting left output channel at the time t , and $y(t)$ is the
 20 function value of the resulting right output channel at the time t , in which the obtainment is iteratively optimized until $\langle x(t), y(t) \rangle$ is within a predetermined definition range.

If there are dropouts or similar defects,
 25 however, an insignificant quantity of single points may lie outside the definition range. In this case, the obtainment is iteratively optimized until a portion of $\langle x(t), y(t) \rangle$ is within the predetermined definition range.

30 The desired definition range is preferably stipulated by a single numerical parameter a , where preferably $0 \leq a \leq 1$. This parameter and hence the

definition range can be usefully stipulated for example by the inequality

$$\operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1$$

5 wherein the relations

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

and

$$g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

10 apply for the complex transfer functions
 $f^*[x(t)]$ and $g^*[y(t)]$ of the output signal $x(t)$, $y(t)$.

15 The user can arbitrarily stipulate such a definition range, on the basis of the unit circle of the complex number plane resp. of the imaginary axis (if the maximum level of the output signal $x(t)$, $y(t)$ has been normalized on the unit circle), by using the parameter a , $0 \leq a \leq 1$.

20 This principle also remains valid when a reference system other than the unit circle of the complex number plane is chosen and a different new definition range is defined. "Definition range" is therefore understood generally to mean an admissible range of values for $\langle x(t), y(t) \rangle$ of the output signal $x(t)$, $y(t)$, which, overall, is intended to contain $\langle x(t), y(t) \rangle$ in full or in part (for example in the case of defective sound recordings which show what are known as dropouts).

 In a preferred variant embodiment, the degree of correlation of the output signals ($x(t)$ and $y(t)$) is normalized. In a preferred variant embodiment, the

level of the maximum of the resulting left and right channel is normalized. In this way, certain parameters can be iteratively optimized in order to attain the desired definition range, without said parameters
 5 affecting the degree of correlation or the level of the maximum of the resulting left channel and right channel.

It also makes sense if – for extremely different parameterizations for φ resp. f (resp. n), α ,
 10 β – criteria which are dependent on $|\langle x(t), y(t) \rangle|$ are used for the stipulation. For this purpose, according to the invention, a corresponding range of values dependent on $|\langle x(t), y(t) \rangle|$ is normalized, so as to constitute a criterion for the optimization of the
 15 parameters.

In one embodiment, a method for obtaining pseudostereophonic output signals $x(t)$ and $y(t)$ by using a converter is therefore proposed, wherein $x(t)$ is the function value of the resulting left output
 20 channel at the time t , wherein $y(t)$ is the function value of the resulting right output channel at the time t , wherein the complex transfer functions $f^*[x(t)]$ and $g^*[y(t)]$ of the output signals are defined:

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

25
$$g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

in which the obtaining is iteratively optimized until the following criterion is satisfied:

$$\operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1,$$

where $0 \leq a \leq 1$ stipulates the desired definition
 30 range.

A remarkable aspect of the methods for obtaining pseudostereophonic signals according to W02009/138205 or according to EP1850639 is the fact that they always provide a perfect center signal. For
 5 this reason, the short time cross correlation

$$(1B) \quad r = (1/2T) * \int_{-T}^T x(t)y(t) dt$$

$$* (1/x(t)_{eff}y(t)_{eff})$$

10 is introduced here for the time interval $[-T, T]$ and the output signals $x(t)$ from the left channel and $y(t)$ from the right channel.

As already mentioned, it makes sense if a uniform degree of correlation is attained for extremely
 15 different parameterizations for φ resp. f (resp. n), α , β . For this purpose, according to the invention, the degree of correlation between the output signals ($x(t)$ and $y(t)$) is normalized. This normalization can preferably be stipulated by means of the specific
 20 variation of λ (left attenuation) or ρ (right attenuation).

On the basis of the uniform degree of correlation, the signal attained can now be systematically subjected to evaluation criteria that
 25 can be influenced by the user.

It also makes sense if a uniform level for the maximum of the resulting left and right channel is being attained for extremely different parameteriza-
 tions for φ resp. f (resp. n), α , β . For this purpose,
 30 in the represented system, the level of the maximum of the resulting left and right channel is normalized, so

that this level is not influenced by the optimization of the parameters.

It makes sense, for example, for the level control for the maximum of the left signal L and of the right signal R to initially be uniformly confined for example to 0 dB by means of a first logic element.

It also makes sense if - for extremely different parameterizations for φ resp. f (resp. n), α , β - criteria which are dependent on $|\langle x(t), y(t) \rangle|$ are used for the stipulation. For this purpose, according to the invention, a corresponding range of values is normalized, so as to constitute a criterion for the optimization of the parameters.

$x(t)$ and $y(t)$ are mapped within the unit circle of the complex number plane. The function $f^*[x(t)] + g^*[y(t)]$ can now be analyzed in more detail in order to draw conclusions concerning the quality of the respective output signal from a device according to W02009/138205 or EP1850639, for example. Any decorrelation between the two signals $f^*[x(t)]$ and $g^*[y(t)]$ is in this case equivalent to a deflection on the real axis when analyzing the function $f^*[x(t)] + g^*[y(t)]$.

The stereo decoder is therefore optimized according to said criteria for example for $|\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}|$ and for $|\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}|$.

This method has proven itself to be particularly advantageous, since a single parameter, namely a , takes optimum account of, in particular, the different nature of the output signals from a device or a method according to W0/2009/138205 or EP1850639. The parameter may preferably be dependent on the type of the audio signal, for example in order to process

speech or music differently on a manual or automatic basis. In the case of speech, unlike music recordings, the definition range determined by the parameter a preferably needs to be restricted significantly due to
 5 disturbing artifacts such as high frequency sidetones during the articulation.

In addition, given the limitation to a single parameter a , any optimum function range can be chosen for $f^*[x(t)] + g^*[y(t)]$ based on the unit circle resp.
 10 the imaginary axis.

If the signals $x(t)$, $y(t)$ do not satisfy the aforementioned conditions, the invention involves optimization being carried out by re-determining the parameters φ or f (resp. n) or α or β – according to an
 15 iterative procedure that is matched with the function values $x[t(\varphi, f, \alpha, \beta)]$ and $y[t(\varphi, f, \alpha, \beta)]$ resp. $x[t(\varphi, n, \alpha, \beta)]$ and $y[t(\varphi, n, \alpha, \beta)]$ – whilst executing steps described so far until $x(t)$ and $y(t)$ meet the aforementioned conditions.

20 In a further step, the relief of the function $f^*[x(t)] + g^*[y(t)]$ for example is now analyzed for the purpose of maximizing the function values thereof. It is possible to show that this procedure is equivalent to the maximization of

25

$$(6B) \quad \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt$$

this expression, for its part, remains less than or equal to the value of

30

$$(7aB) \quad \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)]\}}\} dt$$

$$+ g^*[y(t)]\}}\} dt.$$

In this case too, the user is provided with a tool insofar as he has a free choice of the limit value R^* (or the deviation Δ defined by the inequality (8aB), see below) for this maximization within the context of (8aB). Overall, the following condition must be met for the total number of possible signal variants $x_j(t)$, $y_j(t)$:

$$\begin{aligned}
 (8aB) \quad 0 &\leq R^* - \Delta \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt \\
 &\leq \max \int_{-T}^T |f^*[x_j(t)] \{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi \\
 &\quad + g^*[y_j(t)]| dt \\
 &\leq R^* + \Delta \\
 &\leq \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)] \\
 &\quad + g^*[y(t)]\}}\} dt
 \end{aligned}$$

R^* and Δ are directly related to the loudness of the output signal that is to be attained (i.e. to those parameters which the listener also takes as a basis for assessing the validity of a stereophonic function).

If the neighborhood of the limit value R^* , defined by Δ , or the maximum of all possible integrated reliefs is not reached, optimization in terms of the limit value R^* and the deviation Δ or in terms of the

aforementioned maximum – in accordance with an iterative procedure that is matched with the function values $x[t(\varphi, f, \alpha, \beta)]$ and $y[t(\varphi, f, \alpha, \beta)]$ resp. $x[t(\varphi, n, \alpha, \beta)]$ and $y[t(\varphi, n, \alpha, \beta)]$ – involves new
 5 parameters φ resp. f resp. α resp. β being determined, and all steps described so far being executed until signals $x(t)$, $y(t)$ resp. parameters φ resp. λ resp. ρ resp. f (resp. n) resp. α resp. β result, which correspond to optimum stereophonization.

10 With an appropriate choice of the degree of correlation r , of the parameter a – stipulating the desired respective definition range – and of the limit value R^* and also the deviation Δ thereof, it is possible to configure optimum systems for the
 15 respective area of application (for example speech or music reproduction) for the respective nature of the input signals.

The present considerations remain valid as an entity even if a different reference system than the
 20 unit circle of the imaginary plane is chosen. By way of example, instead of normalizing individual function values, it is also possible to normalize the axis length in order to reduce the computing time accordingly.

25 According to one aspect, it is recommended practice to use (inherently known) compression algorithms or data reduction methods or to analyze characteristic features such as the minima or maxima for the pseudostereophonic signals obtained according
 30 to W02009/138205 or EP1850639, for the purpose of speeding up the evaluation thereof.

Instead of the proposed analysis of $|\langle x(t), y(t) \rangle|$, it is also possible to use $|\langle x(t), y(t) \rangle|^2$ for

optimizing the stereophonization. The computational complexity is significantly reduced as a result.

W02011/009650 can incidentally be applied to devices or methods that generate stereophonic signals which are reproduced by more than two loudspeakers (for example surround sound systems belonging to the prior art).

According to one aspect, W02011/009650 proposes the cascaded downstream connection of a plurality of means (for example logic elements), some of the parameters of which can be aligned, with a stereo decoder (for example according to W0/2009/138205 resp. EP2124486 or EP1850639), wherein feedback for said devices or methods involves the parameters ϕ resp. λ resp. ρ resp. f (resp. n) esp. α resp. β being changed in an optimized way until all conditions of the logic elements are met.

These means (logic elements) can incidentally be arranged differently, and can even - with restrictions - be omitted completely or in part.

For a stereo decoder, for example in a device according to W0/2009/138205 or EP1850639 - for the case of identical inversely proportional attenuations λ and ρ - optimized parameters ϕ , λ , f (resp. the simplifying parameter n), α , β are to be determined in order to convert a mono signal into corresponding pseudostereophonic signals which have optimum decorrelation and loudness (the two criteria according to which the listener assesses the quality of a stereo signal). The intent is to achieve such determination with as few technical means as possible.

FIG. 1B shows the circuit principle for the first two logic elements, as described, for normalizing

the level and for normalizing the degree of correlation of the output signals from a stereo decoder with an MS matrix 110 (for example a stereo decoder according to WO/2009/138205 or EP1850639), wherein the input signal
 5 M and S can (prior to passing through an amplifier connected upstream to the MS matrix) optionally be fed to a circuit according to FIGURE 7B, which is optionally and ideally connected downstream to FIG. 10B, and is activated as soon as the parameter z
 10 resulting from FIG. 10B has been determined (see below).

The first logic element 120 for normalizing the level is in this case coupled to two identical amplifiers having the gain factor ρ^* and ensures a
 15 level control, showing the maximum of 0 dB, of the left channel L and the right channel R.

The signals L and R resulting from the arrangement 110 (for example an MS matrix according to WO/2009/138205 or EP1850639) are amplified uniformly by
 20 the factor ρ^* (amplifiers 118, 119) in such a way that the maximum of both signals has a level of exactly 0 dB (normalization on the unit circle of the complex number plane). This is achieved for example by the downstream connection of a logic element 120 which uses the
 25 feedbacks 121 and 122 and variation or correction of the gain factor ρ^* of the amplifiers 118 and 119 to cause a level control of the maximum value of L and R to reach 0 dB.

The resulting stereo signals $x(t)$ (123) and
 30 $y(t)$ (124), the amplitudes of which are directly proportional to L and R, are fed in a second step to a further logic element 125 which determines the degree of correlation r by using the short time cross relation:

$$(1B) \quad r = (1/2T) * \int_{-T}^T x(t)y(t) dt * (1/x(t)_{eff}y(t)_{eff})$$

5 r can be stipulated by the user in the interval $-1 \leq r \leq 1$ and ideally ranges in the interval $0.2 \leq r \leq 0.7$.

Any deviation from r results in optimized adjustment of the gain factor λ of the amplifier 117 for the S signal via the feedback 126.

10 The resulting signals L and R again pass through the amplifiers 118 and 119 and also the logic element 120, which in turn causes a fresh level control of the maximum value of L and R to reach 0 dB again via the feedbacks 121 and 122, and said signals are then fed to the logic element 125 again.

15 This procedure is performed in an optimized way until the degree of correlation r stipulated by the user has been attained.

20 The result is a stereo signal $x(t)$, $y(t)$ normalized in relation to the unit circle of the complex number plane.

FIG. 2B clarifies the circuit principle which maps the input signals $x(t)$, $y(t)$ on the complex number plane resp. determines the argument of the sum thereof $f^*[x(t)] + g^*[y(t)]$. With this circuit, the resulting
25 signals $x(t)$ and $y(t)$ from the output of FIG. 1B are fed to a matrix in which, following respective amplification by the factor $1/\sqrt{2}$ (amplifiers 229, 230), said signals are broken down into respective identical real and imaginary parts, with the real part
30 formed from the signal $x(t)$, amplified by means of 229, additionally passing through the amplifier 231 with the gain factor -1 . Therefore, the transfer functions:

$$(2B) \quad f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

and

$$(3B) \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i).$$

are obtained.

5 The respective real and imaginary parts are now summed and therefore produce the real part and the imaginary part of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$.

10 The element 232 determines the argument for $f^*[x(t)] + g^*[y(t)]$.

FIG. 3B enables the definition range to be selected by means of the parameter a , $0 \leq a \leq 1$, wherein continuous regulation is made possible by means of the parameter a , on the basis of the unit circle of the complex number plane resp. of the imaginary axis. 15 The user can therefore freely determine the definition range determined by the parameter a on the complex number plane within the unit circle. For this, the squared real part (333a) and the squared imaginary part (334a) of $f^*[x(t)] + g^*[y(t)]$ are calculated. The signal 20 resulting from 333a is then fed to an amplifier 335a and is amplified by the gain factor $1/a^2$ freely selectable by the user. Additionally, the squared sine of the argument of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ is calculated. 25

FIG. 4B, which is to be connected downstream at the output of FIG. 3B, shows the circuit principle for a new third logic element, which checks the signals generated in FIG. 1B and mapped on the complex number plane according to FIG. 2B, according to the condition 30

$$(4aB) \quad \operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1$$

The squared real part and squared imaginary part of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ and the signals resulting from 334a and 335a are in this case fed to a further logic element 436a, which checks whether the above criterion is satisfied, hence whether the values of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ are within the new range of values defined by the user by means of a .

If this is not the case, a feedback 437a is used to determine new optimized values ϕ resp. f (resp. n) resp. α resp. β , and the entire system described so far is passed through again until the values of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ are within the new range of values defined by the user by means of a . The output signals for the logic element 436a are now transferred to the last logic element 538a (FIG. 5B).

The latter finally analyzes the relief of the function $f^*[x(t)] + g^*[y(t)]$ for the purpose of maximizing the function values, wherein the user has a free choice of limit value R^* determined by the inequality (8aB) (and of deviation Δ , likewise determined by the inequality (8aB)) for this maximization. Overall, the condition:

$$(8aB) \quad 0 \leq R^* - \Delta \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt$$

$$\leq \max \int_{-T}^T |f^*[x_j(t)] \{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi$$

$$\begin{aligned}
& + g^*[y_j(t)] \big| dt \\
& \leq R^* + \Delta \\
& \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)] \\
& + g^*[y(t)]\}}\} dt
\end{aligned}$$

must be met. If this is not the case, a feedback 539a is used to iteratively determine new optimized values φ resp. f (resp. n) resp. α resp. β , and the entire
10 system described so far is passed through again until the relief of the function $f^*[x(t)] + g^*[y(t)]$ satisfies the desired maximization of the function values taking into account the limit value R^* resp. the deviation Δ , both defined by the user.

15 Hence, the original pseudostereo decoder, for example according to one of the embodiments in WO/2009/138205 or EP1850639 (in this case assuming the instance of identical inversely proportional
attenuations λ and ρ), is used to iteratively determine
20 new parameters φ resp. f (resp. n) resp. α resp. β until $x(t)$ and $y(t)$ meet the aforementioned conditions (4aB) and (8aB).

In terms of compatibility (determined by the selectable degree of correlation r), definition range
25 (determined by the selectable gain factor a) and loudness (determined by the selectable limit value R^* resp. the selectable deviation Δ), the signals $x(t)$ (123) and $y(t)$ (124) therefore correspond to the selections by the user and are the output signals L^*
30 and R^* from the arrangement described.

Stipulation of the localisation direction:

Occasionally, it is also important to mirror the stereophonic function obtained about the main axis of the directional pattern on which the stereophonic processing is based, since for example mirror inverted mapping in relation to the main axis occurs. This can be achieved manually by swapping the left channel and the right channel.

If an already existing stereo signal L° , R° is to be mapped by the present system, the correct localisation direction can also be ascertained automatically by means of the phantom sources generated using the illustrated pseudostereophonic methodology, as is shown by way of example in FIG. 10B (which is directly connected downstream to FIG. 5B, wherein FIGURE 6B may likewise be added to FIG. 10B for determining the sum of the complex transfer functions $f^*(l(t_i)) + g^*(r(t_i))$ for the already existing stereo signal L° , R°). In this case, at suitably chosen times t_i (for which not all of the subsequently cited correlating function values of the transfer functions $f^*(x(t_i)) + g^*(y(t_i))$ resp. $f^*(l(t_i)) + g^*(r(t_i))$ may be equal to zero for at least one case), the already ascertained transfer function $f^*(x(t_i)) + g^*(y(t_i))$ as shown in FIG. 2B is compared with the transfer function $f^*(l(t_i)) + g^*(r(t_i))$ of the left signal $l(t)$ resp. of the right signal $r(t)$ of the original stereo signal L° , R° (which transfer function is ascertained by using the circuit shown in FIG. 6B, the design of which corresponds to the first part of the circuit for the input signals $x(t)$, $y(t)$ in FIG. 2B). If these transfer functions range in the same quadrant or in the diagonally opposite quadrant of the complex number plane, the total number m of the function values from the cited transfer functions which are located in the same quadrant or in the diagonally opposite quadrant of

the complex number plane is increased by 1 in each case.

An empirically (or statistically determined) specifiable number b , which should be less than or equal to the number of correlating function values of the transfer functions $f^*(x(t_i)) + g^*(y(t_i))$ resp. $f^*(l(t_i)) + g^*(r(t_i))$ unequal to zero, now stipulates the number of necessary matches. Below this number, the left channel $x(t)$ and the right channel $y(t)$ of the stereo signal resulting for example from an arrangement as shown in FIG. 1B, 2B, 3B to 5B are swapped.

If an originally stereophonic signal is to be encoded into a mono signal plus the function f describing the directional pattern (resp. the simplifying parameter n of said function) and likewise the parameters ϕ , α , β , λ or ρ (for example for the purpose of data compression) (for an exemplary output 640a which may be enhanced by the parameter z , see below), it makes sense to jointly encode the information regarding whether the resulting left channel is to be swapped with the resulting right channel (for example expressed by the parameter z , which takes the value 0 or 1, and, if desired, can simultaneously activate a circuit as shown in FIG. 7B).

With slight modifications, similar circuits to the circuits shown in FIG. 6B and 10B can be constructed which can also be used at another location within the electrical circuit or algorithm.

For narrowing or expanding the image width:

For this application too, the additional use of prior art compression algorithms or data reduction methods or the analysis of characteristic features, such as the minima or maxima for the pseudostereophonic

signals obtained is recommended in order to speed up evaluation thereof in accordance with the invention.

Of particular interest (for example for reproducing stereophonic signals in automobiles) is the subsequent narrowing or expanding of the image width of the stereo signal obtained by using the specific variation of the degree of correlation r of the resulting stereo signal resp. the attenuations λ or else ρ (for forming the resulting stereo signal). The previously determined parameters f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left opening angle α and the fictitious right opening angle β can be retained in this case, and it makes sense that now only a final amplitude correction is necessary, for example as per the logic element 120 in FIG. 1B, provided that this narrowing or expanding of the image width is performed manually.

If this is to be automated, series of psychoacoustic experiments show that a constant image width is essentially dependent on the criterion

$$(9B) \quad 0 \leq S^* - \varepsilon \leq \max |\operatorname{Re} \{ f^*[x(t)] + g^*[y(t)] \}| \leq S^* + \varepsilon \leq 1$$

as well as on the criterion

$$(10B) \quad 0 \leq U^* - \kappa \leq \int_{-T}^T |\operatorname{Re} \{ f^*[x(t)] + g^*[y(t)] \}| dt \leq U^* + \kappa$$

(where S^* and ε resp. U^* and κ need to be stipulated differently for telephone signals, for example, than for music recordings). Accordingly, it is now necessary

to determine only suitable function values $x(t)$, $y(t)$ which are dependent on the degree of correlation r of the resulting stereo signal respectively on the attenuations λ or else ρ (for the generation of the
 5 resulting stereo signal) or, where necessary, on a logic element which is identical to the logic element 120 in FIG. 1B, in accordance with an iterative operating principle which is based on feedback.

The arrangement according to FIG. 1B, 2B, 3B to
 10 5B, 6B, 10B can accordingly be enhanced within the context of an arrangement, for instance, of the form shown in FIG. 7B, 8B and/or 9B. FIG. 7B in this case shows a further example of a circuit for normalizing stereophonic or pseudostereophonic signals which, when
 15 connected downstream to FIG. 10B, is activated as soon as the parameter z is present as an input signal. In this case, the initial value of the gain factor λ corresponds to the final value of the gain factor λ in FIG. 1B when the parameter z is transferred, and the
 20 input signals in FIG. 1B are transferred directly as input signals to FIG. 7B at the time of this transfer.

The circuits shown in FIG. 7B to 9B can incidentally also be used autonomously in other circuits or algorithms.

25 In the present arrangement, the left channel and the right channel are swapped in the MS matrix 110 by using a logic element 110a (which also activates this MS matrix as soon as the parameter z is present as an input signal), provided that the parameter z is
 30 equal to 1, otherwise such a swap does not take place.

The resulting output signals L and R from the MS matrix 110 are now amplified (amplifiers 118, 119) uniformly by the factor ρ^* such that the maximum of both signals has a level of exactly 0 dB (normalization

on the unit circle of the complex number plane). This is achieved for example by the downstream connection of a logic element 120 which uses the feedbacks 121 and 122 and variation resp. correction of the gain factor ρ^* of the amplifiers 118 and 119 to cause a level control of the maximum value of L and R to reach 0 dB.

In a further step, the resulting signals $x(t)$ (123) and $y(t)$ (124) are now fed to a matrix as shown in FIG. 8B in which, following respective amplification by the factor $1/\sqrt{2}$ (amplifiers 229, 230), they are split into respective identical real and imaginary parts, with the real part formed from the signal $x(t)$, amplified by means of 229, additionally passing through the amplifier 231 with the gain factor -1. The complex transfer functions $f^*[x(t)] + g^*[y(t)]$ already mentioned in connection with FIG. 2B are thus obtained. The respective real and imaginary parts are now summed and thus result in the real part and the imaginary part of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$.

An arrangement, for example based on the logic element 640 in FIG. 9B, now needs to be connected downstream, which arrangement checks, for a limit value S^* - suitably chosen by the user with respect to the image width of the stereo signal that is to be attained - or a suitably chosen deviation ε - both defined by the inequality (9B) - whether the condition

$$(9B) \quad 0 \leq S^* - \varepsilon \leq \max |\operatorname{Re} \{f^*[x(t)] + g^*[y(t)]\}| \leq S^* + \varepsilon \leq 1$$

is met. If this is not the case, a feedback 641 is used to determine a new optimized value for the degree of correlation r resp. for the attenuations λ or else ρ (for the generation of the resulting stereo signal), and the previous steps just described, as illustrated in FIGURES 7B to 9B, are performed until the above condition (9B) is fulfilled.

The output signals for the logic element 640 are now transferred to an arrangement, for example based on the logic element 642 in FIG. 9B. This arrangement finally analyzes the relief of the function
 5 $f^*[x(t)] + g^*[y(t)]$ for the purpose of optimizing the function values in terms of the image width of the stereo signal that is to be achieved, the user being able to suitably select the limit value U^* and the deviation κ , both defined by the inequality (10B), with
 10 respect to the image width of the stereo signal that is to be achieved. Overall, the condition:

$$(10B) \quad 0 \leq U^* - \kappa \leq \int |f^*[x(t)] + g^*[y(t)]| dt \leq U^* + \kappa$$

must be met. If this is not the case, a feedback 643 is used to determine a new optimized value for the degree
 15 of correlation r resp. for the attenuations λ or else ρ (for the generation of the resulting stereo signal), and the previous steps just described, as illustrated in FIGURES 7B to 9B, are performed until the relief of the function $f^*[x(t)] + g^*[y(t)]$ satisfies the desired
 20 optimization of the function values with respect to the image width taking into account the limit value U^* and the deviation κ , both suitably chosen by the user.

In terms of the image width - determined by the degree of correlation r resp. the attenuations λ or
 25 else ρ (for the generation of the resulting stereo signal) - the signals $x(t)$ (123) and $y(t)$ (124) therefore correspond to the selections by the user and represent the output signals L^{**} and R^{**} from the arrangement which has just been described.

30 The arrangement just described, or portions of this arrangement, can be used as an encoder for a full-fledged stereo signal that is limited to a mono signal plus the parameters φ , f (resp. the simplifying parameter n), α , β , λ resp. ρ).

An already existing stereo signal can be evaluated in respect of r resp. a resp. R^* resp. Δ resp. the localisation direction (resp. parameters S^* resp. ε resp. U^* resp. κ described below) and can then
 5 likewise be encoded anew as a mono signal by using the parameters φ , f (resp. n), α , β , λ resp. ρ in view of a device or a method according to W0/2009/138205 or EP1850639.

Similarly, the arrangement just described, to
 10 which the elements below may possibly be added, can be used as a decoder for mono signals. If φ , f (resp. n), α , β , λ resp. ρ resp. the localisation direction (for example expressed by the parameter z , which can assume the value 0 or 1) are known, such a decoder is reduced
 15 to an arrangement according to W02009/138205 or EP1850639 or W02011/009649 or W02011/009650.

Overall, such encoders or decoders can be used wherever audio signals are recorded, transduced/
 converted, transmitted or reproduced. They provide an
 20 excellent alternative to multichannel stereophonic techniques.

Specific areas of application are telecommu-
 nications (hands-free devices), global networks,
 computer systems, broadcasting and transmission
 25 devices, particularly satellite transmission devices, professional audio technology, television, film and broadcasting and also electronic consumer goods.

The invention is also of particular importance in connection with the obtaining of stable FM stereo
 30 signals under bad reception conditions (for example in automobiles). In this case, it is possible to achieve stable stereophony by simply using the main channel signal ($L + R$) as an input signal, which is the sum of the left channel and of the right channel of the

original stereo signal. The complete or incomplete sub-channel signal $(L - R)$, which is the result of subtracting the right channel from the left channel of the original stereo signal, can also be used in this case in order to form a useable S signal resp. in order to determine or optimize the parameters f (resp. n), which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left opening angle α , the fictitious right opening angle β , the attenuations λ or else ρ for the generation of the resulting stereo signal or, resulting therefrom, the gain factor ρ^* of FIG. 1B for normalizing the left and right channel, resulting from the MS matrixing or from another arrangement according to the invention, on the unit circle (in this case 1, for example, corresponds to the maximum level of 0 dB which has been normalized by using ρ^* , where $x(t)$ is the left output signal resulting from this normalization and $y(t)$ is the right output signal resulting from this normalization) or the degree of correlation r of the resulting stereo signal or the gain factor a for defining the admissible range of values for the sum of the transfer functions of the resulting output signals (for example the complex transfer functions

$$(2B) \quad f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

and

$$(3B) \quad g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i),$$

where, for $0 \leq a \leq 1$, for example, the following is true:

$$(4aB) \quad \operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1)$$

or the limit value R^* or the deviation Δ for stipulating or maximizing the absolute value of the function values of the sum of these transfer functions (where, for this stipulation or maximization and for
 5 the time interval $[-T, T]$ resp. the total number of possible output signals $x_j(t)$, $y_j(t)$, the following for example is true:

$$\begin{aligned}
 (8aB) \quad & 0 \leq R^* - \Delta \leq \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt \\
 & \leq \max \int_{-T}^T |f^*[x_j(t)] \{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi \\
 & + g^*[y_j(t)]| dt \\
 15 \quad & \leq R^* + \Delta \\
 & \leq \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg \{f^*[x(t)] \\
 & + g^*[y(t)]\}}\} dt)
 \end{aligned}$$

20 or the localisation direction of the reproduced sound sources, for example by determining the corresponding quadrants for the function values of the sum, determined for example according to FIG. 6B, of the transfer functions for the left and right channel of
 25 the original stereo signal (which can be optimized for example by virtue of subsequent swapping of the resulting left resp. right channel, see above), or the limit value S^* or the deviation ε (for which, by way of example, it must be true that

$$30 \quad (9B) \quad 0 \leq S^* - \varepsilon \leq \max |\operatorname{Re} \{f^*[x(t)] + g^*[y(t)]\}| \leq S^* + \varepsilon \leq 1)$$

or the limit value U^* or the deviation κ (for which, by way of example, it must be true that

$$(10B) \quad 0 \leq U^* - \kappa \leq \int_{-T}^T \|f^*[x(t)] + g^*[y(t)]\| dt \leq U^* + \kappa,$$

all for determining resp. optimizing the image width of the stereo signal to be attained. In any case, the result is stereophonic function which is constant in respect of the FM signal.

10 In this case too, it is additionally possible to use prior art compression algorithms, data reduction methods or the analysis of characteristic features, such as the minima and maxima, in order to speed up the evaluation of existing or obtained signals or signal
15 components.

In each embodiment and in each figure resp. each element, the circuits, converters, arrangements or logic elements described can be implemented for example by equivalent software programs and programmed
20 processors or DSP or FPGA solutions.

List of symbols used

φ (phi)	angle of incidence
α (alpha)	left fictitious opening angle
β (beta)	right fictitious opening angle
λ	attenuation for the left input signal
ρ	attenuation for the right input signal
The attenuations λ and ρ can be used to adjust the degree of correlation of the stereo signal.	
ψ	polar angle
f	polar distance, which describes the directional pattern of the M signal
P_α, P_β	gain factor for α resp. β
L_α, L_β	time difference for α resp. β
S_α	simulated left signal component of the S signal
S_β	simulated right signal component of the S signal
$x(t)$	left output signal
$y(t)$	right output signal
$f^*[x(t)]$	complex transfer function
$g^*[y(t)]$	complex transfer function
a	gain factor for the definition of the admissible range of values for the sum of the transfer functions of the resulting output signals $x(t)$, $y(t)$

r	degree of correlation, derived from the short-time cross correlation
R^*	limit value for the loudness of the resulting output signals $x(t)$, $y(t)$
Δ	deviation
S^*	1 st limit value for the image width of the resulting output signals $x(t)$, $y(t)$
ε	deviation
U^*	2 nd limit value for the image width of the resulting output signals $x(t)$, $y(t)$
κ	deviation

CH01264/10 resp. PCT/EP2011/063322, at the time
 5 of the present application, have not been published.
 Hereinafter, their contents will therefore be
 reproduced in full for a better understanding of the
 following examples of application of the present
 invention:

10 CH01264/10 resp. PCT/EP2011/063322 also relate
 to signals (for example audio signals) and devices or
 methods for generating, transmitting, processing,
 converting and reproducing them – and relate in
 particular to a method and a device or a system
 15 allowing conclusions to be drawn on the basis of any
 mapping or mappings of one or more signals or also of a
 combination or combinations of two or more signals. In
 the exemplary case of a stereophonic audio signal $x(t)$,

y(t), where x(t) represents the function value of the left input signal at the point in time t, y(t) represents the function value of the right input signal at the point in time t, it is possible to observe for
 5 example the sum of the transfer functions

$$f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$$

$$g^*[y(t)] = [y(t)/\sqrt{2}] * (1 + i)$$

10

in order to be able to draw conclusions as to the properties of the signals.

These conclusions should be reachable in particular on the basis of common properties of two
 15 different signals that appear to be completely random (such as for example audio signals).

Methods so far have attempted – with comparably great difficulty – to simulate this randomization principle and thus make it useful for the signals being
 20 analyzed. For example in the case of DAB (Digital Audio Broadcasting), a Gaussian process is simulated with a so-called Tapped Delay Line model or a Monte Carlo method (colored, complex Gaussian noise in two dimensions) is also used for the simulation of the
 25 mobile radio channel.

Altogether, it can be said about the state of the art that algebraic invariants so far, given the lack of corresponding basis, have never been used for analyzing or optimizing sound events or similar
 30 processes.

Although it has been generally presumed for over 100 years since David Hilbert's groundbreaking work on algebraic invariants that such algebraic invariants exist in particular for Gaussian processes

(an in particular for audio signals), it was never successfully proven.

Not only do CH01264/10 resp. PCT/EP2011/063322 demonstrate such algebraic invariants, the latter are
 5 thus made practically usable commercially for signal technology, for example for calibrating devices or methods for obtaining, improving or optimizing stereophonic or pseudostereophonic audio signals.

In CH01264/10 resp. PCT/EP2011/063322, one
 10 first analyzes a combination $f^\wedge(t)$ or several combinations $f_1^\wedge(t)$, $f_2^\wedge(t)$, ..., $f_p^\wedge(t)$ of at least two signals $s_1(t)$, $s_2(t)$, ..., $s_m(t)$ resp. of their transfer functions $t_1(s_1(t))$, $t_2(s_2(t))$, ..., $t_m(s_m(t))$ – or also the freely definable function $f_1^\#(t)$ or the freely
 15 definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of one signal $s^\#(t)$ or several signals $s_1^\#(t)$, $s_2^\#(t)$, ..., $s_\Omega^\#(t)$ – on the complex number plane resp. their projection on the relief defined by the norm of all points of the complex number plane (the standard cone
 20 whose tip lies in the origin of the complex number plane and whose axis of symmetry is perpendicular to the complex number plane).

The real axis, the imaginary axis and the axis of symmetry of the cone are henceforth represented as a
 25 Cartesian coordinate system with coordinates (x_1, x_2, x_3) . The change in the opening angle of the circle yields the cone equation

$$x_1^2 + x_2^2 - (1/g^{*2}) * x_3 = 0$$

resp. the coefficients $[1 \ 1 \ -1/g^{*2}]$. Two cone
 30 equations are now analyzed:

$$S := a_x^2 := 1 * x_1^2 + 1 * x_2^2 - (1/g^2) * x_3^2 = 0$$

and

$$S' := a'_x{}^2 := 1 * x_1^2 + 1 * x_2^2 - (1/g'^2) * x_3^2 = 0.$$

As known, one invariant is thus

$$a_a{}'^2 := 1 * 1^2 + 1 * 1^2 - (1/g^2) * (1/g'^4).$$

5 Both cones S , S' are non polar if

$$(1/g^2) * (1/g'^4) = 2.$$

S is then inscribed harmonically in S' .

If, for example, one analyzes the above combination $\hat{f}(t)$ or several combinations $\hat{f}_1(t)$,
 10 $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of two or more signals $s_1(t)$, $s_2(t)$,
 ..., $s_m(t)$ resp. of their transfer functions $t_1(s_1(t))$,
 $t_2(s_2(t))$, ..., $t_m(s_m(t))$ for two time intervals t_1 , t_2 -
 or also the freely definable function $f^\#(t)$ or the
 freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of
 15 one signal $s^\#(t)$ or of several signals $s_1^\#(t)$, $s_2^\#(t)$,
 ..., $s_\Omega^\#(t)$ for two time intervals t_1 , t_2 - as well as
 the functions S , S' and Σ' with

$$\begin{aligned} \Sigma' &:= u_a{}'^2 := A'u_1^2 + B'u_2^2 + C'u_3^2 + 2F'u_2u_3 \\ &+ 2G'u_3u_1 + 2H'u_1u_2 \\ 20 \quad &= 1 * u_1^2 + 1 * u_2^2 + (1/g''^2) * u_3^2 + 2 * 1 \\ &* u_2u_3 + 2 * 1 * u_3u_1 + 2 * 1 * u_1u_2 \\ &= 0. \end{aligned}$$

The following is true

$$aA' + bB' + cC' + 2fF' + 2gG' + 2hH' = 0,$$

25 and S and Σ' should be non polar:

$$1 * 1 + 1 * 1 - (1/g^2) * (1/g''^2) = 0$$

or

$$(1/g^2) * (1/g'^2) = 2.$$

Thus, provided $g' = g'' = 1$ and $g = 1/\sqrt{2}$ applies, the non-polarity of S with S' and Σ' is
5 ensured.

Considering the standard cone

$$S' = 1 * x_1^2 + 1 * x_2^2 - 1 * x_3^2 = 0$$

thus simultaneously enables the analysis of identical vanishing invariants relative to S

10
$$S = 1 * x_1^2 + 1 * x_2^2 - 2 * x_3^2 = 0$$

resp.

$$\begin{aligned} \Sigma' &= 1 * u_1^2 + 1 * u_2^2 + 1 * u_3^2 + 2 * 1 * u_2 u_3 \\ &+ 2 * 1 * u_3 u_1 + 2 * 1 * u_1 u_2 \\ &= 0. \end{aligned}$$

15 The relation

$$a_a'^2 := 1 * 1^2 + 1 * 1^2 - 2 * 1^2 = 0$$

is thus linear in the coefficients of the equations

$$S = 1 * x_1^2 + 1 * x_2^2 - 2 * x_3^2 = 0$$

and

20
$$\begin{aligned} \Sigma' &= 1 * u_1^2 + 1 * u_2^2 + 1 * u_3^2 + 2 * 1 * u_2 u_3 \\ &+ 2 * 1 * u_3 u_1 + 2 * 1 * u_1 u_2 \\ &= 0. \end{aligned}$$

According to Hilbert's famous theorem on invariants (Hilbert, page 291, § 2), in our system, the linear combination

$$\begin{aligned} & \phi [1, 1, -2] * [1, 1, -1]^2 + \\ 5 \quad & \Theta [1, 1, -2] * [1, 1, 1]^2 = 0 \end{aligned}$$

in turn represents an invariant. Thus, for example, any piercing straight lines of $f^*(t_1)$ and $f^*(t_2)$, considered on the plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$, ξ_1 and ξ_2 , correspond to an
10 infinite number of invariants of S and S' resp. of S and Σ' .

When considering the standard cone reflected on the complex number plane, the change of opening angle of the cone yields the cone equation

$$15 \quad -x_1^2 - x_2^2 + (1/g^{*2}) * x_3$$

resp. the coefficients $[-1 \ -1 \ 1/g^{*2}]$. Two cone equations are then considered:

$$S := a_x^2 := -1 * x_1^2 - 1 * x_2^2 + (1/g^2) * x_3^2 = 0$$

and

$$\begin{aligned} 20 \quad S' &:= a'_x{}^2 := -1 * x_1^2 - 1 * x_2^2 + (1/g'^2) * x_3^2 \\ &= 0. \end{aligned}$$

It is well known that one invariant is thus:

$$a_a'^2 := -1 * (-1)^2 - 1 * (-1)^2 + (1/g^2) * (1/g'^4).$$

25 Both cones S , S' are non polar if

$$(1/g^2) * (1/g'^4) = 2.$$

S is then inscribed harmonically in S'.

If, for example, one analyzes the above combination $f^\wedge(t)$ or several combinations $f_1^\wedge(t)$,
 5 $f_2^\wedge(t)$, ..., $f_p^\wedge(t)$ or two or more signals $s_1(t)$, $s_2(t)$,
 ..., $s_m(t)$ resp. of their transfer functions $t_1(s_1(t))$,
 $t_2(s_2(t))$, ..., $t_m(s_m(t))$ for two time intervals t_1 , t_2 -
 or also the freely definable function $f^\#(t)$ or the
 freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of
 10 one signal $s^\#(t)$ or of several signals $s_1^\#(t)$, $s_2^\#(t)$,
 ..., $s_\Omega^\#(t)$ for two time intervals t_1 , t_2 - as well as
 the functions S, S' and Σ' with

$$\begin{aligned} \Sigma' &:= u_a'^2 := A'u_1^2 + B'u_2^2 + C'u_3^2 + 2F'u_2u_3 \\ &+ 2G'u_3u_1 + 2H'u_1u_2 \\ 15 \quad &= 1 * u_1^2 + 1 * u_2^2 + (1/g''^2) * u_3^2 + 2 * 1 \\ &* u_2u_3 + 2 * 1 * u_3u_1 + 2 * 1 * u_1u_2 \\ &= 0. \end{aligned}$$

The following is true

$$aA' + bB' + cC' + 2fF' + 2gG' + 2hH' = 0,$$

20 and S and Σ' should be non polar:

$$-1 * 1 - 1 * 1 + (1/g^2) * (1/g''^2) = 0$$

or

$$(1/g^2) * (1/g''^2) = 2.$$

Thus, again provided $g' = g'' = 1$ and $g = 1/\sqrt{2}$
 25 applies, the non-polarity of S with S' and Σ' is
 ensured.

Considering the standard cone

$$S' = -1 * x_1^2 - 1 * x_2^2 + 1 * x_3^2 = 0$$

thus simultaneously enables the consideration of identical vanishing invariants relative to S

$$S = -1 * x_1^2 - 1 * x_2^2 + 2 * x_3^2 = 0$$

5 resp.

$$\begin{aligned} \Sigma' &= 1 * u_1^2 + 1 * u_2^2 + 1 * u_3^2 + 2 * 1 * u_2 u_3 \\ &+ 2 * 1 * u_3 u_1 + 2 * 1 * u_1 u_2 \\ &= 0. \end{aligned}$$

The relation

$$\begin{aligned} 10 \quad a_{a'}^2 &:= -1 * (-1)^2 - 1 * (-1)^2 + 2 * 1^2 \\ &= -1 * 1 - 1 * 1 + 2 * 1 = 0 \end{aligned}$$

is thus linear in the coefficients of the equations

$$S = -1 * x_1^2 - 1 * x_2^2 + 2 * x_3^2 = 0$$

and

$$\begin{aligned} 15 \quad \Sigma' &= 1 * u_1^2 + 1 * u_2^2 + 1 * u_3^2 + 2 * 1 * u_2 u_3 \\ &+ 2 * 1 * u_3 u_1 + 2 * 1 * u_1 u_2 \\ &= 0. \end{aligned}$$

According to Hilbert's theorem on invariants (Hilbert, page 291, § 2), in our system, the linear
20 combination

$$\begin{aligned} &\phi [-1, -1, 2] * [-1, -1, 1]^2 + \\ &\oplus [-1, -1, 2] * [1, 1, 1]^2 = 0 \end{aligned}$$

in turn represents an invariant. Thus, for example, any piercing straight lines of $f^*(t_1)$ and
25 $f^*(t_2)$, considered on the plane spanned by the vectors

$(-1, -1, 2)$ und $(1, 1, 1)$, ξ_1 and ξ_2 , correspond to an infinite number of invariants of S and S' resp. of S and Σ' .

5 All combinatory possibilities for the situation of S , S' and Σ' , as can easily be seen, are exhausted in terms of the result in the same plane.

The practical application of this fact in signal technology allows for example the analysis of a combination $\hat{f}(t)$ or of several combinations $\hat{f}_1(t)$,
 10 $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of at least two signals $s_1(t)$, $s_2(t)$, ..., $s_m(t)$ resp. of their transfer functions $t_1(s_1(t))$, $t_2(s_2(t))$, ..., $t_m(s_m(t))$ – or also of the freely definable function $f^\#(t)$ or the freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of one signal $s^\#(t)$
 15 or of several signals $s_1^\#(t)$, $s_2^\#(t)$, ..., $s_\Omega^\#(t)$ by determining said invariants. Doing this enables the function of this combination $\hat{f}(t)$ or of these combinations $\hat{f}_1(t)$, $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of at least two signals $s_1(t)$, $s_2(t)$, ..., $s_m(t)$ resp. of their transfer
 20 functions $t_1(s_1(t))$, $t_2(s_2(t))$, ..., $t_m(s_m(t))$ – or also the freely definable function $f^\#(t)$ or the freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of one signal $s^\#(t)$ or of several signals $s_1^\#(t)$, $s_2^\#(t)$, ..., $s_\Omega^\#(t)$ for example on the complex number plane – the x_1
 25 axis then coincides for example with the real axis, the x_2 axis then coincides for example with the imaginary axis – and subsequently enables the piercing points of these functions in the present example with the plane spanned by the vectors $(1, 1, -2)$ und $(1, 1, 1)$ or $(-1, -1, 2)$ and $(1, 1, 1)$ to be analyzed and which now
 30 constitute absolutely or also in terms of their statistical distribution precise reference points for further analysis, processing or optimization. For example, according to W02011/009649 or also
 35 W02011/009650, an optimization of pseudostereophonic audio signals can be carried out and subsequently the

piercing points of the sum of the transfer functions
 $f^*[x(t)] = [x(t)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t)] =$
 $[y(t)/\sqrt{2}] * (1 + i)$, see further below, with the plane
 spanned by the vectors (1, 1, -2) and (1, 1, 1) or (-1,
 5 -1, 2) and (1, 1, 1) can be determined. When these
 piercing points are weighted by using an appropriate
 method, see the detailed description, it will result in
 parameterizations according W02011/009649 or also
 W02011/009650, which prove particularly advantageous
 10 for the audio signals being analyzed.

According to one aspect, it is recommended
 practice to use (inherently known) compression
 algorithms or data reduction methods or to analyze
 characteristic features such as the minima or maxima
 15 for the signals or transfer functions or combinations
 or functions being analyzed, for the purpose of
 speeding up the evaluation thereof in accordance with
 the invention.

20 The algebraic principles of the present
 invention will first be explained with the aid of
 Figures 1C to 4C.

FIG. 1C shows the non-polarity constraint for S
 and S' resp. S and Σ' . Reference number 1001
 25 illustrates that for S and S', expressed by $f^{\sim}(g')$,
 1002 illustrates that for S and Σ' , expressed by
 $f^{\sim}(g'')$. The intersection point 1004 of 1001 with the
 diagonal of the 1st quadrant 1003 illustrates the
 coinciding of S and S', the intersection point 1005 of
 30 1001 and 1002 illustrates the sought non-polarity
 constraint itself; $g' = g'' = 1$ can be read directly.

FIG. 2C shows the functions S (2001), S' (2002)
 and Σ' (2003) as well as the plane 2004 spanned by the

vectors $(1, 1, -2)$ and $(1, 1, 1)$, on which the sought algebraic invariants of S and S' resp. of S and Σ' are located, from the perspective of the 1st quadrant of the corresponding complex number plane. 2005, 2006 and
 5 2007 show the planes defined by the Cartesian coordinate system $x_1 = u_1$, $x_2 = u_2$, $x_3 = u_3$.

FIG. 3C shows the functions S (2001), S' (2002) and Σ' (2003) as well as the plane 2004 spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$, on which the sought
 10 algebraic invariants of S and S' resp. of S and Σ' are located, likewise from the perspective of the 1st quadrant of the corresponding complex number plane. 2005, 2006 and 2007 show the planes defined by the Cartesian coordinate system $x_1 = u_1$, $x_2 = u_2$, $x_3 = u_3$.

FIG. 4C shows the functions S (2001), S' (2002) and Σ' (2003) as well as the plane 2004 spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$, on which the sought algebraic invariants of S and S' resp. of S and Σ' are located, from the perspective of the 4th quadrant of
 20 the corresponding complex number plane. 2005, 2006 and 2007 show the planes defined by the Cartesian coordinate system $x_1 = u_1$, $x_2 = u_2$, $x_3 = u_3$.

The practical-commercial application of the algebraic invariants just developed covers nearly the
 25 entire signal processing field. The stochastic analysis of audio signals, as known for example from the field of digital audio broadcasting (DAB), is of particular interest; in that field, so far, for the simulation of Gaussian processes, techniques such as the so-called
 30 Tapped Delay Line model or Monte Carlo methods (colored, complex Gaussian noise in two dimensions) were used, see bibliographical references. The transfer of the operating principles applied there for stabilizing optimization processes, such as described

in W02011/009650, would be conceivable, yet not very efficient in practice.

On the basis of the present algebraic invariants, it is however possible to define for
 5 example a weighting as follows:

For this purpose, a first optimization according to W02011/009650, FIG. 1B, 2B, 3B to 5B, is performed on a signal section of the length t_1 . The outputs of FIG. 5B are connected for example to a
 10 module 6001 according to FIG. 6C and the invariants (constituted at the intersection points ξ_{h1} of the sum of the complex transfer functions $f^*[x(t_1)] = [x(t_1)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_1)] = [y(t_1)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1) and
 15 located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1 , u_1 of the represented algebraic model coincides here with the real axis, the axis x_2 , u_2 with the imaginary axis) are analyzed in
 20 terms of their statistical distribution. All ξ_{h1} from the total number k_1 are stored in a memory ("stack") valid for all further described operation sequences; the mean value

$$\xi^{\circ}_1 := \left(\sum_{h_1=1}^{k_1} \xi_{h1} \right) / k_1$$

is likewise calculated. The latter is stored together with the parameterization φ_1 , f_1 (resp. n_1), α_1 , β_1 , determined by means of said first optimization, in a
 30 further dictionary valid for all further described operation sequences.

According to the function command 6004, in a second step a second optimization according to

W02011/009650, FIG. 1B, 2B, 3B to 5B, is then performed on a signal section t_2 of any length. The outputs of FIG. 5B are in turn again connected to the module 6001 and the invariants (constituted at the intersection points ξ_{h2} of the sum of the complex transfer functions $f^*[x(t_2)] = [x(t_2)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_2)] = [y(t_2)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$ or also $(-1, -1, 2)$ and $(1, 1, 1)$ and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1 , u_1 of the represented algebraic model coincides here with the real axis, the axis x_2 , u_2 with the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{h2} from the total number k_2 are stored in a memory ("stack") valid for all further described operation sequences; the mean value

$$\xi_{2}^{\circ} := \left(\sum_{h_2=1}^{k_2} \xi_{h2} \right) / k_2$$

is likewise calculated. The latter is in turn again added together with the parameterization φ_2 , f_2 (resp. n_2), α_2 , β_2 , determined by means of said second optimization, to the first mean value ξ_1° as well as its parameterization φ_1 , f_1 (resp. n_1), α_1 , β_1 , in the dictionary valid for all further described operation sequences. Since the memory ("stack") now contains more than one mean value, the module 6002 of FIG. 6C is then activated.

The latter calculates the mean value ξ_2^* of all intersection points ξ_{h1} , ξ_{h2} stored in the stack:

$$\xi_2^* := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} \right) / (k_1 + k_2)$$

and selects from the dictionary among the mean values ξ_1^o, ξ_2^o with their associated parameterization the mean value closest to ξ_2^* . If this is the case for both mean values ξ_1^o, ξ_2^o , ξ_1^o resp. the parameterization ϕ_1, f_1 (resp. n_1), α_1, β_1 is selected from the dictionary. The mean value selected from the dictionary is then transmitted together with ξ_2^* to the module 6003 of FIG. 6C. The latter verifies whether the mean value selected by the module 6002 of FIG. 6C is within the interval $[-\sigma + \xi_2^*, \xi_2^* + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely selectable by the user, of the Gauss distribution fictitiously set as zero in ξ_2^* :

$$f^{\sim}(z_2^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_2^* - \xi_2^*)^2)/\sigma^2}$$

If the mean value selected by the module 6002 of FIG. 6C is within the interval $[-\sigma + \xi_2^*, \xi_2^* + \sigma]$, the parameterization selected by the module 6002 of FIG. 6C according to 6010 of FIG. 6C in the arrangement of FIG. 7A resp. FIG 1B (which shows again for the sake of clarity the amplifier 717 and the MS matrix, wherein both of which are to be passed through only once) resp. the outputs 6006 and 6007 of FIG. 1B are activated, likewise the outputs 6008 and 6009 of FIG. 2B. The output 6006 of FIG. 1B runs into the input 6006 of FIG. 6C, the output 6007 of FIG. 1B runs into the input 6007 of FIG. 6C, the output 6008 of FIG. 2B runs into the input 6008 of FIG. 6C and the output 6009 of FIG. 2B runs into the input 6009 of FIG. 6C. Reference 6006 of FIG. 6C directly represents the output signal $x(t)$ from the module 6003 of FIG. 6C, reference 6007 of FIG. 6C directly represents the output signal $y(t)$ from the module 6003 of FIG. 6C, 6008 of FIG. 6C directly represents the output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 6C, 6009 of FIG. 6C directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 6C. These signals, in the signal processing described above, are to be treated as

if they represented the output signals of FIG. 5B, which with FIG. 6C then constitutes, in this example of application, an inseparable unit.

If the mean value selected by the module 6002 of FIG. 6C is outside the interval $[-\sigma + \xi_{*2}, \xi_{*2} + \sigma]$, in a q^{th} step a q^{th} optimization is performed according to W02011/009650, FIG. 1B, 2B, 3B to 5B, on a signal section t_q of any length. The outputs of FIG. 5B are in turn connected to the module 6001 of FIG. 6C and the invariants (constituted at the intersection points ξ_{hq} of the sum of the complex transfer functions $f^*[x(t_q)] = [x(t_q)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_q)] = [y(t_q)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1) and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1, u_1 of the represented algebraic model coincides here with the real axis, the axis x_2, u_2 with the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{hq} from the total number k_q are added to $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq-1}$ in a memory ("stack") valid for all further described operation sequences; the mean value

$$\xi_q^{\circ} := \left(\sum_{h_q=1}^{k_q} \xi_{hq} \right) / k_q$$

is likewise calculated. The latter is in turn added together with the parameterization φ_q, f_q (resp. n_q), α_q, β_q , determined by means of said q^{th} optimization, to the first mean values $\xi_1^{\circ}, \xi_1^{\circ}, \dots, \xi_{q-1}^{\circ}$ as well as to their parameterizations φ_1, f_1 (resp. n_1), α_1, β_1 ; φ_2, f_2 (resp. n_2), α_2, β_2 ; $\dots$; φ_{q-1}, f_{q-1} (resp. n_{q-1}), $\alpha_{q-1}, \beta_{q-1}$, in the dictionary valid for all further described operation sequences. Since the memory ("stack") now contains more than one mean value, the module 6002 of FIG. 6C is then activated.

The latter calculates the mean value ξ^*_{q} of all intersection points $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq}$ stored in the stack:

$$\xi^*_{\text{q}} := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} + \dots + \sum_{h_q=1}^{k_q} \xi_{hq} \right) / (k_1 + k_2 + \dots k_q)$$

and selects from the dictionary among the mean values $\xi^{\circ}_1, \xi^{\circ}_2, \dots, \xi^{\circ}_q$ with their associated parameterization ϕ, f (resp. n), α, β , the mean value closest to ξ^*_{q} . If this is the case for different parameterizations, the parameterization that appears most often in the dictionary is selected. If several parameterizations appear the same number of times, the one that exhibits the widest scattering in the dictionary is selected, i.e. the one for which the difference $d - c$ is maximum, where d represents the last and c the first index number of the optimization steps respectively undergone. If this too applies to several parameterizations, the first one that appears is selected. If two mean values from $\xi^{\circ}_1, \xi^{\circ}_2, \dots, \xi^{\circ}_q$ are close to ξ^*_{q} , insofar as in a $q-1^{\text{th}}$ step one of the two mean values resp. their associated parameterizations is selected from the dictionary, the very same one resp. its associated parameterizations is retained. The mean value selected from the dictionary is then transmitted together with ξ^*_{q} to the module 6003 of FIG. 6C. The latter verifies whether the mean value selected by the module 6002 of FIG. 6C is within the interval $[-\sigma + \xi^*_{\text{q}}, \xi^*_{\text{q}} + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire process illustrated here, of the Gauss distribution fictitiously set as zero in ξ^*_{q} :

$$f^{\sim}(z_q^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_q^* - \xi^*_{\text{q}})^2)/\sigma^2}$$

If the mean value selected by the module 6002 of FIG. 6C is within the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$, the parameterization selected by the module 6002 of FIG. 6C according to 6010 of FIG. 6C in the arrangement of FIG. 7A resp. FIG 1B resp. the outputs 6006 and 6007 of FIG. 2B are activated, likewise the outputs 6008 and 6009 of FIG. 2B as well as the corresponding inputs and outputs of FIG. 6C. Reference 6006 of FIG. 6C again directly represents the output signal $x(t)$ from the module 6003 of FIG. 6C, reference 6007 of FIG. 6C directly represents the output signal $y(t)$ from the module 6003 of FIG. 6C, 6008 of FIG. 6C directly represents the output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 6C, 6009 of FIG. 6C directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 6C. These signals, in the signal processing described above, are again to be treated as if they represented the output signals of FIG. 5B, which with FIG. 6C constitutes, in the present example of application, an inseparable unit.

If the mean value selected by the module 6002 of FIG. 6C is outside the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$, in a $q+1^{\text{th}}$ step a $q+1^{\text{th}}$ optimization is performed in the same form as for the q^{th} step and for the q^{th} optimization. The process is repeated as long as necessary until one element of the dictionary fulfills the above requirements or a maximum number of allowed optimization steps has been reached.

FIG. 5C shows the convergence behavior of the just established weighting function for three optimization steps: 5001 represents in this case the first mean value ξ_1^o , 5002 the second mean value ξ_2^o , 5003 the first Gauss distribution fictitiously set as zero in ξ_2^* :

$$f^{\sim}(z_2^*) = (1 / (\sqrt{(2\pi)} * \sigma)) * e^{-(1/2)*((z_2^* - \xi_2^*)^2)/\sigma^2},$$

where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire process illustrated here, 5004 the third mean value ξ^0_3 , which remains within the inflexion points defined
 5 by σ of the Gauss distribution 5005 of same standard deviation, fictitiously set as zero in ξ^*_3 , and thus fulfills the convergence criterion.

In each case, the result is a parameterization φ , f (resp. n), α , β , which supplies a
 10 pseudostereophonic function that on average is optimum in relation to all algebraic invariants.

As the number of signal sections increases, the distribution of the intersection points ξ of the algebraic invariants on the half-plane respectively
 15 analyzed with the complex number plane approximates the Gauss distribution. The smaller the chosen standard deviation σ is, the closer to ideal the resulting parameterization will be. However, as an only finite number of signal sections are available, σ should not
 20 be chosen too small.

Nevertheless, the method in terms of its convergence is considerably faster for sufficiently long signal sections than mentioned simulation models, since algebraic invariants are available for the first
 25 time to serve as valid "points of reference" for a weighting of already determined parameterizations.

In principle, use of the described invariants is however not compulsorily bound to a system as in FIG. 3A to 12A resp. 1B, 2B, 3B, 4B, 5B, 6B, 10B, 7B to
 30 9B resp. 5C and 6C, but can on the contrary be applied to nearly any part of the entire field of signal technology. The described normalization on the unit circle of the complex number plane, wherein for example two signals $x(t)$ and $y(t)$ are amplified uniformly by a

factor ρ^* (amplifiers 118, 119 in FIG. 1B) such that the maximum of both signals has a level of exactly 0 dB, is not necessary in this case. The value range for the analyzed combination or combinations – or also for
 5 any above function or functions of one or several signals – can thus according to the invention comprise the entire value range of the real or complex number plan and therefore does not remain limited to the unit circle.

10 If it is intended to normalize a combination $\hat{f}(t)$ or several combinations $\hat{f}_1(t)$, $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of at least two signals $s_1(t)$, $s_2(t)$, ..., $s_m(t)$ resp. of their transfer functions $t_1(s_1(t))$, $t_2(s_2(t))$, ..., $t_m(s_m(t))$ – or also the above freely definable
 15 function $f^\#(t)$ or freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of one signal $s^\#(t)$ or of several signals $s_1^\#(t)$, $s_2^\#(t)$, ..., $s_\Omega^\#(t)$ – although there is no imperative necessity of doing so, this normalization can be freely definable.

20 Instead of the effective level control of the maximum value of L and R in the present example to 0 db (amplifiers 118 and 119 as well as logic element 120 in FIG. 1B), it is thus possible for example to achieve a normalization on the basis of the sums of the *mean*
 25 *square energy* of each of the two respective channels, i.e. for L equal to $x_\#(t_i)$ and R equal to $y_\#(t_i)$ on the basis of the sum $z_{Li} + z_{Ri}$ of

$$z_{Li} = (1/T_i) \int_0^{T_i} x_\#(t_i) dt$$

30

and

$$Z_{Ri} = (1/T_i) * \int_0^{T_i} y_{\#}(t_i) dt$$

to introduce a normalization with respect to a
 5 reference value z_{ref} according to the principle that $x_{\#}(t_i)$ and $y_{\#}(t_i)$ each are to be multiplied by the factor

$$z_{ref} / (z_{Li} + z_{Ri})$$

If this principle is generalized for example
 according to FIG. 7C, this principle can be extended to
 10 any number of signals $s_j(t_i)$ of the total number δ (7001), for each of which respectively the *mean square energy* is calculated (7002):

$$Z_{sj}(t_i) = (1/T_i) * \int_0^{T_i} s_j(t_i) dt,$$

15

where again T_i represents the time span of the time interval t_i , and which signals are subsequently multiplied (7003) with a weighting G_j defined for each signal $s_j(t)$.

20 Following this, the products $G_j * Z_{sj}(t_i)$ thus obtained are summed according to 7004. This sum is transmitted to the amplifiers of 7005 that are individually connected to the original signal inputs $s_1(t_i)$, $s_2(t_i)$, ..., $s_{\delta}(t_i)$, and the signals $s_1(t_i)$,
 25 $s_2(t_i)$, ..., $s_{\delta}(t_i)$ are then uniformly amplified by the factor

$$z_{ref} / \left(\sum_{j=1}^{\delta} G_j * Z_{sj}(t_i) \right)$$

and for example transmitted to the module 7006, which -
 according to the disclosure of the invention -
 determines the invariants of the combination $\hat{f}(t)$ or
 of several combinations $\hat{f}_1(t)$, $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of
 5 at least two signals $s_1(t)$, $s_2(t)$, ..., $s_\delta(t)$ resp. of
 their transfer functions $t_1(s_1(t))$, $t_2(s_2(t))$, ...,
 $t_\delta(s_\delta(t))$ - or also the freely definable function $f^\#(t)$
 or the freely definable functions $f_1^\#(t)$, $f_2^\#(t)$, ...,
 $f_\mu^\#(t)$ of one signal $s^\#(t)$ or of several signals $s_1^\#(t)$,
 10 $s_2^\#(t)$, ..., $s_\delta^\#(t)$.

Similar considerations can in particular extend
 also for example to audio signals according for example
 to ITU-R BS.1770; the modules 7002 to 7005 are then
 omitted and the signals can be forwarded directly to
 15 the module 7006.

Even when analyzing a single, sufficiently long
 time span for a combination $\hat{f}(t)$ or several
 combinations $\hat{f}_1(t)$, $\hat{f}_2(t)$, ..., $\hat{f}_p(t)$ of at least two
 signals $s_1(t)$, $s_2(t)$, ..., $s_m(t)$ resp. of their transfer
 20 functions $t_1(s_1(t))$, $t_2(s_2(t))$, ..., $t_m(s_m(t))$ - or also
 for the freely definable function $f^\#(t)$ or the freely
 definable functions $f_1^\#(t)$, $f_2^\#(t)$, ..., $f_\mu^\#(t)$ of one
 signal $s^\#(t)$ or of several signals $s_1^\#(t)$, $s_2^\#(t)$, ...,
 $s_\Omega^\#(t)$ - the invariants according to the above can be
 25 determined and used specifically in an industrial-
 technical context (for example for evaluating
 individual signals or processing or optimizing any
 signal parameter or transmission parameter). The
 application of the object of the invention it thus not
 30 limited to the examples given above, but is oriented in
 principle towards the described determination of
 invariants for any signals or signal sections of any
 length according to the disclosure of the invention.

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3. Rec. ITU-R BS.1770.

10

FIG. 1D shows a first inventive example of
application. The basic circuit of EP1850639 has been
expanded by a further time parameter s , which is
multiplied respectively with the time differences L_A
15 resp. L_B and thus yields the new time differences L'_A
resp. L'_B as follows:

$$(3D) \quad L'_A = L_A * s = [\sqrt{(5/4 - \sin\phi)} - 1/2] * s$$

and

$$(4D) \quad L'_B = L_B * s = [\sqrt{(5/4 + \sin\phi)} - 1/2] * s$$

20

The new circuit schema can be directly seen
from FIG. 1D. $s > 0$ can represent both a constant (an
ideal value for s for the present arrangement is for
example 100 ms) as well as be selected freely by the
user. In practice, the values of delay A' resp. delay
25 B' of FIG. 1D essential for determining the spatial
perception of the listener.

The transposition of this operating principle
to W02009/138205 results for example in the inventive
arrangements of FIG. 2D, 3D, 4D, 5D, 6D (to which
30 elements according to W02011/009649 are added for a
better illustration) as well as in FIG. 7D, FIG. 8D,

FIG. 9D, FIG. 10D, FIG. 11D, FIG. 12D, FIG. 13D, FIG. 14D. Again, the time differences L_α and L_β are each multiplied with the same parameter $s > 0$ and yield new time differences L'_α and L'_β . Consequently, the
 5 following relations apply:

$$(1D) \quad L'_\alpha = L_\alpha * s = \left\{ -\frac{f(\alpha)}{2\sin\alpha} + \sqrt{\frac{f^2(\alpha)}{4\sin^2\alpha} + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi) / \sin\alpha} \right\} * s$$

and

$$10 \quad (2D) \quad L'_\beta = L_\beta * s = \left\{ -\frac{f(\beta)}{2\sin\beta} + \sqrt{\frac{f^2(\beta)}{4\sin^2\beta} + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta} \right\} * s$$

The new circuit schema can be seen directly in FIG. 2D, 3D, 4D, 5D, 6D (to which elements according to
 15 W02011/009649 are added for a better illustration) as well as in FIG. 7D, FIG. 8D, FIG. 9D, FIG. 10D, FIG. 11D, FIG. 12D, FIG. 13D, FIG. 14D.

N.B. If stipulating $f(\varphi) = f(\alpha) = f(\beta) = 1$ and $\sin\alpha = \sin\beta = 1$ in the circuit schema 309 of FIG. 2D
 20 resp. in the circuit schema 409 of FIG. 3D resp. in the circuit schema 509 of FIG. 4D, variants of EP1850639 will result according to the invention that depend only on the parameter s and on the angle φ enclosed by the main axis and the sound source. In particular, for the
 25 circuit schema 309 of FIG. 2D, the following relations apply:

$$(3D) \quad L'_\alpha = L_\alpha * s = [\sqrt{5/4 - \sin\varphi} - 1/2] * s$$

and

$$(4D) \quad L'_\beta = L_\beta * s = [\sqrt{5/4 + \sin\varphi} - 1/2] * s$$

as well as

$$P_{\alpha} = 5/4 - \sin\varphi$$

and

$$P_{\beta} = 5/4 + \sin\varphi,$$

5 for the circuit schema 409 of FIG. 3D the relations

$$(3D) \quad L_{\alpha}' = L_{\alpha} * s = [\sqrt{(5/4 - \sin\varphi)} - 1/2] * s$$

and

$$(4D) \quad L_{\beta}' = L_{\beta} * s = [\sqrt{(5/4 + \sin\varphi)} - 1/2] * s$$

as well as

$$10 \quad P_M' = 1/(5/4 - \sin\varphi)$$

and

$$P_{\beta}' = (5/4 + \sin\varphi)/(5/4 - \sin\varphi),$$

where $1/P_M' = (5/4 - \sin\varphi)$ may not be equal to zero or
 an element of an environment of zero, and for the
 15 circuit schema 509 of FIG. 4D the following relations
 apply:

$$(3D) \quad L_{\alpha}' = L_{\alpha} * s = [\sqrt{(5/4 - \sin\varphi)} - 1/2] * s$$

and

$$(4D) \quad L_{\beta}' = L_{\beta} * s = [\sqrt{(5/4 + \sin\varphi)} - 1/2] * s$$

20 as well as

$$P_M'' = 1/(5/4 + \sin\varphi)$$

and

$$P_{\alpha}' = (5/4 - \sin\varphi)/(5/4 + \sin\varphi),$$

where $1/P_{\alpha}'' = (5/4 + \sin\varphi)$ may not be equal to zero or an element of an environment of zero.

5 N.B. It is also possible in the circuit schema 309 of FIG. 2D resp. in the circuit schema 409 of FIG. 3D resp. in the circuit schema 509 of FIG. 4D for $\alpha = \beta = \pi/2$ and thus to stipulate $\sin\alpha = \sin\beta = 1$, yet to keep $f(\varphi)$ resp. $f(\alpha)$ resp. $f(\beta)$, i.e. to create an
10 exclusive dependence on the angle φ enclosed by the sound source and the main axis of the mono signal that is to be rendered stereophonic, and on the directional pattern of the mono signal that is to be rendered
15 stereophonic (which can be represented in polar coordinates). For the circuit schema 309 of FIG. 2D in particular the following relations apply:

$$L_{\alpha}' = L_{\alpha} * s = \{- f(\alpha)/2 + \sqrt{[f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)]} * s$$

and

$$20 \quad L_{\beta}' = L_{\beta} * s = \{- f(\beta)/2 + \sqrt{[f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]} * s$$

as well as

$$P_{\alpha} = f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)$$

and

$$25 \quad P_{\beta} = f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)$$

and for the circuit schema 409 of FIG. 3D the relations

$$L_{\alpha}' = L_{\alpha} * s = \{-f(\alpha)/2 + \sqrt{[f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)]}\} * s$$

and

$$5 \quad L_{\beta}' = L_{\beta} * s = \{-f(\beta)/2 + \sqrt{[f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]}\} * s$$

as well as

$$P_M' = 1/[f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)]$$

and

$$10 \quad P_{\beta}' = [f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)] / [f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)],$$

where $1/P_M' = [f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)]$ may not be equal to zero or an element of an environment of zero, and for the circuit schema 509 of FIG. 4D the relations

$$15 \quad L_{\alpha}' = L_{\alpha} * s = \{-f(\alpha)/2 + \sqrt{[f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)]}\} * s$$

and

$$L_{\beta}' = L_{\beta} * s = \{-f(\beta)/2 + \sqrt{[f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]}\} * s$$

20 as well as

$$P_M'' = 1/[f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]$$

and

$$P_{\alpha}' = [f^2(\alpha)/4 + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)] / [f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]$$

where $1/P_M'' = [f^2(\beta)/4 + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)]$
 may not be equal to zero nor an element of an
 environment of zero. Under the condition $L_\alpha' = L_\beta'$ it
 is possible with $\sin\alpha = \sin\beta = 1$ to further simplify
 5 with the above formulae (see below) the FIG. 17D, FIG.
 18D, FIG. 19D, FIG. 20D, FIG. 21D, FIG. 22D, FIG. 23D,
 FIG. 24D, FIG. 25D, FIG. 26D, FIG. 27D, FIG. 28D resp.
 FIG. E9, FIG. E10, FIG. E11, FIG. E12, FIG. E13, FIG.
 E14, FIG. E15, FIG. E16, FIG. E17. In particular, see
 10 below, it is also possible here in FIG. E14 to
 integrate the amplification factor $1/\tau$ (which is then
 to be multiplied with the amplification factor P_M')
 into the gain M, resp. in FIG. E17 to integrate the
 amplification factor $1/\tau$ (which is then to be
 15 multiplied with the amplification factor P_M'') into the
 gain M.

The choice of s, as shown in practice, is not
 trivial. If s is chosen too small, the
 pseudostereophonic effect to be achieved disappears, if
 20 s is chosen too great, disturbing artifacts will
 result. If s is about 100 milliseconds, this will yield
 for a modified device or a method according to
 W02009/138205 or W02011/009649 ideal pseudostereophonic
 signals that exhibit the same quality as with a classic
 25 MS recording technique.

If the object of the invention is applied to
 W02011/009649 or W02011/009650, in particular to FIG.
 1B as well as FIG. 4B (feedback 437a) and FIG. 5B
 (feedback 539a), it is not only the parameters φ resp.
 30 f (resp. n) resp. α resp. β that are optimized
 iteratively but henceforth newly also the parameter $s > 0$
 introduced according to the invention, in a new
 manner and as represented in FIG. 15D, using the same
 evaluation method described above. The overall system
 35 presented in W02011/009649 or W02011/009650 resp.
 CH01264/10 resp. PCT/EP2011/063322 in this case remains

fully preserved with the addition of the just mentioned element (wherein FIG. 1B is hereinafter to be replaced with FIG. 15D).

If a system according to CH01264/10 resp.
 5 PCT/EP2011/063322 is to be added according to the invention, FIG. 6C is to be replaced by FIG. 16D. In more detail, a first optimization according to CH01776/09 resp. W02011/009650, FIG. 15D, 2B, 3B to 5B, is performed on a signal section of the length t_1 . The
 10 outputs of FIG. 5B are connected for example to a module 6001 according to FIG. 16D and the invariants (constituted at the intersection points ξ_{h1} of the sum of the complex transfer functions $f^*[x(t_1)] = [x(t_1)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_1)] = [y(t_1)/\sqrt{2}] * (1$
 15 $+ i)$ with the half-plane spanned by the vectors $(1, 1, -2)$ and $(1, 1, 1)$ or also $(-1, -1, 2)$ and $(1, 1, 1)$ and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1, u_1 of the represented algebraic model coincides here with the real axis, the
 20 axis x_2, u_2 with the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{h1} from the total number k_1 are stored in a memory ("stack") valid for all further described operation sequences; the mean value

$$\begin{aligned} & k_1 \\ 25 \quad \xi^{\circ}_1 &:= (\sum_{h_1=1} \xi_{h1}) / k_1 \\ & h_1 = 1 \end{aligned}$$

is likewise calculated. The latter is stored together with the parameterization φ_1, f_1 (resp. n_1), α_1, β_1, s_1
 30 determined by means of said first optimization, in a dictionary further valid for all further described operation sequences.

According to the function command 6004 of FIG. 16D, in a second step a second optimization according

to W02011/009650, FIG. 15D, 2B, 3B to 5B, is then performed on a signal section t_2 of any length. The outputs of FIG. 5B are in turn connected to the module 6001 of FIG. 16D and the invariants (constituted in the intersection points ξ_{h2} of the sum of the complex transfer functions $f^*[x(t_2)] = [x(t_2)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_2)] = [y(t_2)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors (1, 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1) and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1 , u_1 of the represented algebraic model coincides here with the real axis, the axis x_2 , u_2 with the imaginary axis) are analyzed in terms of their statistical distribution. All ξ_{h2} from the total number k_2 are stored in a memory ("stack") valid for all further described operation sequences; the mean value

$$\xi_2^{\circ} := \left(\sum_{h_2=1}^{k_2} \xi_{h2} \right) / k_2$$

is likewise calculated. The latter is in turn added together with the parameterization φ_2 , f_2 (resp. n_2), α_2 , β_2 , s_2 determined by means of said second optimization, to the first mean value ξ_1° as well as its parameterization φ_1 , f_1 (resp. n_1), α_1 , β_1 , s_1 in the dictionary valid for all further described operation sequences. Since the memory ("stack") now contains more than one mean value, the module 6002 of FIG. 16D is then activated.

The latter calculates the mean value ξ_2^* of all intersection points ξ_{h1} , ξ_{h2} stored in the stack:

$$\xi_2^* := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} \right) / (k_1 + k_2)$$

and selects from the dictionary among the mean values ξ_1^o, ξ_2^o with their associated parameterization the mean value closest to ξ_2^* . If this is the case for both mean values ξ_1^o, ξ_2^o , ξ_1^o resp. the parameterization ϕ_1, f_1 (resp. n_1), α_1, β_1, s_1 is selected from the dictionary. The mean value selected from the dictionary is then transmitted together with ξ_2^* to the module 6003 of FIG. 16D. The latter verifies whether the mean value selected by the module 6002 is within the interval $[-\sigma + \xi_2^*, \xi_2^* + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely selectable by the user, of the Gauss distribution fictitiously set as zero in ξ_2^* :

$$f^{\sim}(z_2^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_2^* - \xi_2^*)^2)/\sigma^2}$$

If the mean value selected by the module 6002 of FIG. 16D is within the interval $[-\sigma + \xi_2^*, \xi_2^* + \sigma]$, the parameterization selected by the module 6002 of FIG. 16D according to 6010 of FIG. 16D in the arrangement of FIG. 6D resp. FIG 15D (which shows again for the sake of clarity the amplifier 717 and the MS matrix, wherein both of which are to be passed through only once) resp. the outputs 6006 and 6007 of FIG. 15D are activated, likewise the outputs 6008 and 6009 of FIG. 2B. The output 6006 of FIG. 15D runs into the input 6006 of FIG. 16D, the output 6007 of FIG. 15D runs into the input 6007 of FIG. 16D, the output 6008 of FIG. 2B runs into the input 6008 of FIG. 16D and the output 6009 of FIG. 2B runs into the input 6009 of FIG. 16D. Reference 6006 of FIG. 16D directly represents the output signal $x(t)$ from the module 6003 of FIG. 16D, reference 6007 of FIG. 16D directly represents the output signal $y(t)$ from the module 6003 of FIG. 16D, 6008 of FIG. 16D directly represents the output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D, 6009 of FIG. 16D directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D. These signals, in the signal processing described

above, are to be treated as if they represented the output signals of FIG. 5B, which with FIG. 16D then constitutes, in this example of application, an inseparable unit.

5 If the mean value selected by the module 6002 is outside the interval $[-\sigma + \xi_{*2}, \xi_{*2} + \sigma]$, in a q^{th} step a q^{th} optimization is performed according to W02011/009650, FIG. 15D, 2B, 3B to 5B on a signal section t_q of any length. The outputs of FIG. 5B are in
 10 turn connected to the module 6001 of FIG. 16D and the invariants (constituted at the intersection points ξ_{hq} of the sum of the complex transfer functions $f^*[x(t_q)] = [x(t_q)/\sqrt{2}] * (-1 + i)$ and $g^*[y(t_q)] = [y(t_q)/\sqrt{2}] * (1 + i)$ with the half-plane spanned by the vectors (1,
 15 1, -2) and (1, 1, 1) or also (-1, -1, 2) and (1, 1, 1) and located in the 1st or also 3rd quadrants of the complex number plane – the axis of x_1, u_1 of the represented algebraic model coincides here with the real axis, the axis x_2, u_2 with the imaginary axis) are
 20 analyzed in terms of their statistical distribution. All ξ_{hq} from the total number k_q are added to $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq-1}$ in a memory ("stack") valid for all further described operation sequences; the mean value

$$25 \quad \xi_q^{\circ} := \left(\sum_{h_q=1}^{k_q} \xi_{hq} \right) / k_q$$

is likewise calculated. The latter is in turn added together with the parameterization φ_q, f_q (resp. n_q), α_q, β_q, s_q determined by means of said q^{th} optimization,
 30 to the first mean values $\xi_1^{\circ}, \xi_1^{\circ}, \dots, \xi_{q-1}^{\circ}$ as well as to their parameterizations φ_1, f_1 (resp. n_1), α_1, β_1, s_1 ; φ_2, f_2 (resp. n_2), α_2, β_2, s_1 ; $\dots$; φ_{q-1}, f_{q-1} (resp. n_{q-1}), $\alpha_{q-1}, \beta_{q-1}, s_{q-1}$, in the dictionary valid for all further described operation sequences. Since the memory

("stack") now contains more than one mean value, the module 6002 of FIG. 16D is then activated.

The latter calculates the mean value ξ^*_{q} of all intersection points $\xi_{h1}, \xi_{h2}, \dots, \xi_{hq}$ stored in the
5 stack:

$$\xi^*_{\text{q}} := \left(\sum_{h_1=1}^{k_1} \xi_{h1} + \sum_{h_2=1}^{k_2} \xi_{h2} + \dots + \sum_{h_q=1}^{k_q} \xi_{hq} \right) / (k_1 + k_2 + \dots k_q)$$

and selects from the dictionary among the mean values
10 $\xi^{\circ}_1, \xi^{\circ}_2, \dots, \xi^{\circ}_q$ with their associated parameterization φ, f (resp. n), α, β, s , the mean value closest to ξ^*_{q} . If this is the case for different parameterizations, the parameterization that appears most often in the dictionary is selected. If several parameterizations
15 appear the same number of times, the one that exhibits the widest scattering in the dictionary is selected, i.e. the one for which the difference $d - c$ is maximum, where d represents the last and c the first index number of the optimization steps respectively
20 undergone. If this too applies to several parameterizations, the first one that appears is selected. If two mean values from $\xi^{\circ}_1, \xi^{\circ}_2, \dots, \xi^{\circ}_q$ are close to ξ^*_{q} , insofar as in a $q-1^{\text{th}}$ step one of the two mean values resp. their associated parameterizations is
25 selected from the dictionary, the very same one resp. its associated parameterizations is retained. The mean value selected from the dictionary is then transmitted together with ξ^*_{q} to the module 6003 of FIG. 16D. The latter verifies whether the mean value selected by the
30 module 6002 of FIG. 16D is within the interval $[-\sigma + \xi^*_{\text{q}}, \xi^*_{\text{q}} + \sigma]$, where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire process illustrated here, of the Gauss distribution fictitiously set as zero in ξ^*_{q} :

$$\tilde{f}(z_q^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_q^* - \xi_q^*)^2)/\sigma^2}$$

If the mean value selected by the module 6002 of FIG. 16D is within the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$, the parameterization selected by the module 6002 of FIG. 16D according to 6010 of FIG. 16D in the arrangement of FIG. 6D resp. FIG 15D resp. the outputs 6006 and 6007 of FIG. 15D are activated, likewise the outputs 6008 and 6009 of FIG. 2B as well as the corresponding inputs and outputs of FIG. 16D. The output 6006 of FIG. 15D runs into the input 6006 of FIG. 16D, the output 6007 of FIG. 15D runs into the input 6007 of FIG. 16D, the output 6008 of FIG. 2B runs into the input 6008 of FIG. 16D and the output 6009 of FIG. 2B runs into the input 6009 of FIG. 16D. Reference 6006 of FIG. 16D again directly represents the output signal $x(t)$ from the module 6003 of FIG. 16D, reference 6007 directly represents the output signal $y(t)$ from the module 6003 of FIG. 16D, 6008 of FIG. 16D directly represents the output signal $\text{Re } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D, 6009 of FIG. 16D directly represents the output signal $\text{Im } f^*[x(t)] + g^*[y(t)]$ from the module 6003 of FIG. 16D. These signals, in the signal processing described above, are again to be treated as if they represented the output signals of FIG. 5B, which with FIG. 16D constitutes, in the present example of application, an inseparable unit.

If the mean value selected by the module 6002 of FIG. 16D is outside the interval $[-\sigma + \xi_q^*, \xi_q^* + \sigma]$, in a $q+1^{\text{th}}$ step a $q+1^{\text{th}}$ optimization is performed in the same form as for the q^{th} step and for the q^{th} optimization. The process is repeated as long as necessary until one element of the dictionary fulfills the above requirements or a maximum number of allowed optimization steps has been reached.

FIG. 5C shows the convergence behavior of the just established weighting function for three optimization steps: 5001 represents in this case the first mean value ξ°_1 , 5002 the second mean value ξ°_2 ,
 5 5003 the first Gauss distribution fictitiously set as zero in ξ^*_2 :

$$f^{\sim}(z_2^*) = (1 / (\sqrt{2\pi} * \sigma)) * e^{-(1/2)*((z_2^* - \xi^*_2)^2)/\sigma^2},$$

where $\sigma > 0$ represents the standard deviation, freely selectable by the user at the beginning of the entire
 10 process illustrated here, 5004 the third mean value ξ°_3 , which remains within the inflexion points defined by σ of the Gauss distribution 5005 of same standard deviation, fictitiously set as zero in ξ^*_3 , and thus fulfills the convergence criterion.

15 In each case, the result is a parameterization φ , f (resp. n), α , β and henceforth newly s , which supplies a pseudostereophonic function that on average is optimum in relation to all algebraic invariants.

As the number of signal sections increases, the
 20 distribution of the intersection points ξ of the algebraic invariants on the half-plane respectively analyzed with the complex number plane approximates the Gauss distribution. The smaller the chosen standard deviation σ is, the closer to ideal the resulting
 25 parameterization will be. However, as an only finite number of signal sections are available, σ should not be chosen too small.

Nevertheless, the method represented in FIG.
 16D in terms of its convergence is considerably faster
 30 for sufficiently long signal sections than mentioned simulation models, since algebraic invariants are available for the first time to serve as valid "points

of reference" for a weighting of already determined parameterizations.

Hereinafter two variant embodiments, represented in FIG. 13D and FIG. 14D, are presented
 5 using the modified MS matrix described by the equations (1A) and (2A).

Two further variant embodiments, illustrated in FIG. 7D and FIG. 8D, of the circuits of FIG. 3D and FIG. 4D, derived from FIG. E3 and FIG. E6, which in
 10 turn can be derived from the equations (3A) and (4A) resp. from FIG. 4A and FIG. 5A for the special case of identical inversely proportional attenuations $\lambda = \rho$, are also presented. The panorama potentiometers 411 and 412 of FIG. 3D are in this case replaced with an
 15 amplifier 717 with the amplification factor λ ; the same applies for the panorama potentiometers 511 and 512 of FIG. 4D.

According to the equations (3AA) and (4AA), the variant embodiments of FIG. 9D and FIG. 11D can be
 20 derived from FIG. 7D and FIG. 8D and, in the same way, according to the equations (3AAA) and (4AAA) the variants of FIG. 10D and FIG. 12D can be derived.

FIG. 17D illustrates a simplification of the circuit schema 309 of FIG. 2D for the case $L'_\alpha = L'_\beta$.
 25 This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies.

FIG. 18D illustrates a simplification of the circuit schema of FIG. 6A for the case $L'_\alpha = L'_\beta$. This
 30 case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies.

FIG. 19D illustrates a simplification of FIG. 6D, likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification already supplies convincing results that rival a first-class MS recording, and is thus not trivial.

FIG. 20D illustrates a circuit equivalent to FIG. 19D, wherein the amplification factor λ is directly integrated in gain S'_α . It represents the simplest circuit form that in its exact angle-dependent virtualization of a classical MS array is not trivial.

FIG. 21D illustrates a simplification of the circuit schema 409 of FIG. 3D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

may not be equal to zero nor an element of an environment of zero.

FIG. 22D illustrates a simplification of the circuit schema of FIG. 13D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

may not be equal to zero nor an element of an environment of zero.

FIG. 23D illustrates an inventive simplification derived from FIG. 3D, likewise for the case $L'_\alpha = L'_\beta$, wherein the elements 411 and 412 - provided that the attenuation for the left input signal λ is equal to the attenuation for the right input signal ρ - are replaced with the amplification of the S signal by the factor λ represented here (for deriving this fact, see the above formulae (3A) and (4A)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-class MS recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\alpha)}{(4 \sin^2 \alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin \varphi)} / \sin \alpha$$

may not be equal to zero nor an element of an environment of zero.

FIG. 24D illustrates a circuit equivalent to FIG. 23D, wherein the amplification factor λ is directly integrated in gain S'_β . It represents a simple circuit form that in its exact angle-dependent virtualization of a classical MS array is however not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\alpha)}{(4 \sin^2 \alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin \varphi)} / \sin \alpha$$

may not be equal to zero nor an element of an environment of zero.

FIG. 25D illustrates a simplification of the circuit schema 509 of FIG. 4D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\beta)}{(4 \sin^2 \beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin \varphi) / \sin \beta}$$

may not be equal to zero nor an element of an environment of zero.

FIG. 26D illustrates a simplification of the circuit schema of FIG. 14D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\beta)}{(4 \sin^2 \beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin \varphi) / \sin \beta}$$

may not be equal to zero nor an element of an environment of zero.

FIG. 27D illustrates an inventive simplification derived from FIG. 4D likewise for the case $L'_\alpha = L'_\beta$, wherein the elements 511 and 512 – provided that the attenuation for the left input signal λ is equal to the attenuation for the right input signal ρ – are replaced with the amplification of the S signal by the factor λ represented here (for deriving this fact, see the above formulae (3A) and (4A)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e.

for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-class MS recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification
 5 also the following expression applies:

$$\frac{f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta}{\sin\varphi}$$

may not be equal to zero nor an element of an environment of zero.

10 FIG. 28D illustrates a circuit equivalent to FIG. 27D, wherein the amplification factor λ is directly integrated in gain S'_α . It represents a simple circuit form that in its exact angle-dependent virtualization of a classical MS array is however not
 15 trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta}{\sin\varphi}$$

20 may not be equal to zero nor an element of an environment of zero.

When applying the considerations on FIG. E1 and FIG. E2, see above, the variant embodiments illustrated enable furthermore the following new variants according to FIG. E9, FIG. E10, FIG. E11, FIG. E12, FIG. E13,
 25 FIG. E14, FIG. E15, FIG. E16 and FIG. E17 for the special case of identical inversely proportional attenuations $\lambda = \rho$.

FIG. E9, taking into account FIG. E1 resp. the equations (3AA) and (4AA) illustrates a simplification
 30 of FIG. 6D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles

are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification already supplies convincing results that rival a first-class MS recording, and is thus not trivial.

5 FIG. E10, taking into account FIG. E2 resp. the equations (3AAA) and (4AAA), illustrates a simplification of FIG. 6D likewise for the case $L'_\alpha = L'_\beta$. This case arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification already supplies convincing results that rival a first-class MS recording, and is thus not trivial.

15 FIG. E11, taking into account FIG. E2 resp. the equations (3AAA) and (4AAA), illustrates a circuit equivalent to FIG. E10, wherein the amplification factor λ' is directly integrated in gain S'_α . It represents the simplest circuit form that in its exact angle-dependent virtualization of a classical MS array is not trivial.

20 FIG. E12, taking into account FIG. 23D resp. the equations (3AA) and (4AA), illustrates an inventive simplification derived from FIG. 3D, likewise for the case $L'_\alpha = L'_\beta$, wherein the elements 411 and 412 – provided that the attenuation of the left input signal λ is equal to the attenuation for the right input signal ρ – are replaced with the amplification of the M signal by the factor $1/\lambda$ represented here (for deriving this fact, see the above formulae (3AA) and (4AA)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-class MS recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

may not be equal to zero nor an element of an environment of zero.

5 FIG. E13, taking into account FIG. 23D resp. the equations (3AAA) and (4AAA), illustrates an inventive simplification likewise derived from FIG. 3D, likewise for the case $L'_\alpha = L'_\beta$, wherein the elements 411 and 412 - provided that the attenuation of the left
10 input signal λ is equal to the attenuation for the right input signal ρ - are replaced with the amplification of the M signal by the factor $1/\tau$ resp. the amplification of the S signal by the factor λ' represented here (for deriving this fact, see the above
15 formulae (3AAA) and (4AAA)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-class MS
20 recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

25 may not be equal to zero nor an element of an environment of zero.

 FIG. E14, taking into account FIG. 23D resp. the equations (3AAA) and (4AAA), illustrates a circuit equivalent to FIG. E13, wherein the amplification
30 factor λ' is directly integrated in gain S'_β . It represents a simple circuit form that in its exact angle-dependent virtualization of a classical MS array

is not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

- 5 may not be equal to zero nor an element of an environment of zero.

FIG. E15, taking into account FIG. 27D resp. the equations (3AA) and (4AA), illustrates an inventive simplification derived from FIG. 3D, likewise for the
 10 case $L'_\alpha = L'_\beta$, wherein the elements 511 and 512 - provided that the attenuation of the left input signal λ is equal to the attenuation for the right input signal ρ - are replaced with the amplification of the M signal by the factor $1/\lambda$ represented here (for deriving
 15 this fact, see the above formulae (3AA) and (4AA)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-
 20 class MS recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta$$

- 25 may not be equal to zero nor an element of an environment of zero.

FIG. E16, taking into account FIG. 27D resp. the equations (3AAA) and (4AAA), illustrates an inventive simplification derived from FIG. 4D, likewise
 30 for the case $L'_\alpha = L'_\beta$, wherein the elements 511 and 512 - provided that the attenuation of the left input signal λ is equal to the attenuation for the right

input signal ρ – are replaced with the amplification of the M signal by the factor $1/\tau$ resp. the amplification of the S signal by the factor λ' represented here (for deriving this fact, see the above formulae (3AAA) and (4AAA)). The case $L'_\alpha = L'_\beta$ arises for example when the fictitious opening angles are symmetrical with the main axis, i.e. for $\alpha = \beta$, wherein $\varphi = 0$ applies. Such a simplification also supplies convincing results that rival a first-class MS recording, and is thus not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta}{\sin\varphi)/\sin \beta}$$

may not be equal to zero nor an element of an environment of zero.

FIG. E17, taking into account FIG. 27D resp. the equations (3AAA) and (4AAA), illustrates a circuit equivalent to FIG. E16, wherein the amplification factor λ' is directly integrated in gain S'_α . It represents a simple circuit form that in its exact angle-dependent virtualization of a classical MS array is not trivial. As mentioned in W02009/138205, for this simplification also the following expression applies:

$$\frac{f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta}{\sin\varphi)/\sin \beta}$$

may not be equal to zero nor an element of an environment of zero.

In the same way, incidentally, in FIG. E14 the amplification factor $1/\tau$ (which is then to be multiplied with the amplification factor P_M') can be integrated into the gain M resp. in FIG. E17 the amplification factor $1/\tau$ (which is then to be

multiplied with the amplification factor P_M'') can be integrated into the gain M.

Similarly, in FIG. E12 the amplification factor $1/\lambda$ (which is then to be multiplied with the
 5 amplification factor P_M') can be integrated into the gain M resp. in FIG. E15 the amplification factor $1/\lambda$ (which is then to be multiplied with the amplification factor P_M'') can be integrated into the gain M.

N.B. FIG. 11F shows the total possible
 10 parameter combinations for the parameter f (resp. n) which describe the directional pattern of the signal that is to be rendered stereophonic, the angle φ - to be ascertained manually or by metrology - enclosed by the main axis and the sound source, the fictitious left
 15 opening angle α and the fictitious right opening angle β . These parameter combinations, which each include one line of FIG. 11F, can be derived from the following equations (1D) and (2D) as well as from the amplification factors P_α resp. P_β resp. P_α' resp. P_β'
 20 resp. P_M' resp. P_M'' formed from the discriminants of the equations (1D) and (2D) according to W02009/138205, wherein "X" designates those parameters of the following equations on which the respective circuit depends; other parameters yield constants or values
 25 equal to 1 or 0. The associated comments represent concrete cases in which the respective parameter combination is encountered. In particular, for FIG. 15 of W02009/138205, the following equations apply:

$$(1D) \quad L_\alpha' = L_\alpha * s = \{ - f(\alpha)/(2\sin\alpha) + \sqrt{[f^2(\alpha)/(4\sin^2\alpha) + f^2(\varphi) - (f(\alpha)*f(\varphi)*\sin\varphi)/\sin\alpha]} \} * s$$

30

and

$$(2D) \quad L_{\beta}' = L_{\beta} * s = \{-f(\beta)/(2\sin\beta) + \sqrt{[f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta]}\} * s$$

as well as

$$5 \quad P_{\alpha} = f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha$$

and

$$P_{\beta} = f^2(\beta)/(4 \sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi)/\sin \beta,$$

10 for the FIG. 16 of W02009/138205 the equations

$$(1D) \quad L_{\alpha}' = L_{\alpha} * s = \{-f(\alpha)/(2\sin\alpha) + \sqrt{[f^2(\alpha)/(4\sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi) / \sin\alpha]}\} * s$$

and

$$15 \quad (2D) \quad L_{\beta}' = L_{\beta} * s = \{-f(\beta)/(2\sin\beta) + \sqrt{[f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta]}\} * s$$

as well as

$$20 \quad P_{M'} = 1/[f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha]$$

and

$$P_{\beta}' = [f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta] / [f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin \alpha],$$

where the expression $[f^2(\alpha)/(4 \sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi)/\sin\alpha]$ may not be equal to zero nor an element of an environment of zero,

and for the FIG. 17 of W02009/138205 the equations

$$(1D) \quad L_{\alpha}' = L_{\alpha} * s = \{- f(\alpha)/(2\sin\alpha) + \sqrt{[f^2(\alpha)/(4\sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi) / \sin\alpha]} * s$$

and

$$(2D) \quad L_{\beta}' = L_{\beta} * s = \{- f(\beta)/(2\sin\beta) + \sqrt{[f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta]} * s$$

as well as

$$P_M'' = 1 / [f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta]$$

and

$$P_{\alpha}' = [f^2(\alpha)/(4\sin^2\alpha) + f^2(\varphi) - (f(\alpha) * f(\varphi) * \sin\varphi) / \sin\alpha] / [f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta],$$

where the expression $[f^2(\beta)/(4\sin^2\beta) + f^2(\varphi) + (f(\beta) * f(\varphi) * \sin\varphi) / \sin\beta]$ may not be equal to zero nor an element of an environment of zero. The principles given above by way of example for the simplification of the circuit schemata according to FIG. 15 resp. FIG. 16 resp. FIG. 17 of W02009/138205 or also according to EP1850639 for the case $L_{\alpha}' = L_{\beta}'$ or $P_{\alpha} = P_{\beta}$ or also for the case of identical discriminants of L_{α}' and L_{β}' , if appropriate in combination with the inversely proportional attenuations λ or ρ or with the amplification factors $1/\lambda$ or $1/\tau$ or λ' , can be applied

in the same way on the circuit schemata obtained according to FIG. 11F. In particular, FIG. 11F shows – see for example the cases of constant values for the angle φ – that the angle (φ) to be ascertained manually or by metrology, enclosed by the main axis and the sound source, can also be chosen or stipulated arbitrarily.

N.B. If φ lies to the left of the main axis of the mono signal that is to be rendered stereophonic, φ is positive; if on the contrary φ lies to the right of the main axis of the mono signal that is to be rendered stereophonic, φ is negative.

N.B. In particular, instead of the respective calculation of the time differences L_α or L_β or L_A or L_B or L'_α or L'_β or L'_A or L'_B resp. of the amplification factors P_α or P_β or P'_α or P'_β or P_M or P'_M or P''_M or P_A or P_B , if appropriate taking into account the inversely proportional attenuations λ or ρ or the amplification factors $1/\lambda$ or $1/\tau$ or λ' , it is also possible to use a dictionary with corresponding values for the inventive delaying or amplification of the input signal that is to be rendered stereophonic. For example, α resp. β resp. φ can be varied in 5° steps and a corresponding dictionary can be created on the basis of the values obtained for the time differences L_α or L_β or L_A or L_B or L'_α or L'_β or L'_A or L'_B resp. for the amplification factors P_α or P_β or P'_α or P'_β or P_M or P'_M or P''_M or P_A or P_B , if appropriate taking into account the inversely proportional attenuations λ or ρ or the amplification factors $1/\lambda$ or $1/\tau$ or λ' .

FIG. 4F illustrates the simplest form of application of all-pass filters to the object of the invention. It provides that for inventive arrangements according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp.

PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also
 FIG. E5 bzw. FIG. E6 or also FIG. E7 or also FIG. E8
 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also
 FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D
 5 resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also
 just described arrangements of the form of FIG. 17D
 resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D
 resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D
 resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or
 10 also FIG. E10 or simplified FIG. E11 resp. FIG. E12
 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or
 also FIG. E16 or simplified FIG. E17 resp. FIG. 18D
 resp. FIG. 22D resp. FIG. 26D, the characteristic good
 emphasis of the center sound sources is eliminated in
 15 favor of a dispersed sound with good source
 localization. For example, an all-pass filter of any
 order (see for example FIG. 1F resp. 2F) is connected
 downstream to the left and/or right channel of a
 stereophonic or pseudostereophonic output signal of an
 20 inventive arrangement according to EP1850639 or
 W02009/138205 or W02011/009649 or W02011/009650 or
 CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or
 also FIG. E4 or also FIG. E5 bzw. FIG. E6 or also FIG.
 E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp.
 25 FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A
 resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also
 FIG. E2 or also just described arrangements of the form
 of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or
 also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or
 30 also FIG. 25D resp. FIG. 27D or simplified FIG. 28D
 resp. FIG. E9 or also FIG. E10 or simplified FIG. E11
 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14
 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17
 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D. FIG. 4F,
 35 in connection with the object of the invention, thus
 provides for a convincing dispersation of the phantom
 sources of the resulting signal.

Similarly, a phase shifter (see for example FIG. 3F) connected downstream to the left and/or right channel of a stereophonic or pseudostereophonic output signal of an inventive arrangement according to

5 EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 bzw. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp.

10 FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or

15 simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D, provides an additional dispersation of the

20 sound, see for example FIG. 5F. In particular, a combination of all-pass filters and phase shifters for the subsequently influencing of the sound image in a targeted manner is also possible.

All described arrangements can be used in

25 particular on systems for reproducing sound with more than two loudspeakers. An approximation of multi-channel signals is especially made possible, as shown in FIG. 6F resp. FIG. 7F resp. FIG. 8F.

For example, according to FIG. 6F, a sideways-

30 balanced five-channel signal according to Rec. ITU-R BS.774-1 (see FIG. 9F) can be derived from a mono signal. For this purpose, a mono signal M according to FIG. 6F is converted into a pseudostereophonic signal L, R by means of an inventive arrangement according to

35 EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322

resp. FIG. E3 or also FIG. E4 or also FIG. E5 bzw. FIG.
 E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or
 also FIG. 10D resp. FIG. 11D or also FIG. 12D resp.
 FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D
 5 resp. FIG. E1 or also FIG. E2 or also just described
 arrangements of the form of FIG. 17D resp. FIG. 19D or
 simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or
 simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or
 simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or
 10 simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or
 simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or
 simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp.
 FIG. 26D. Similarly, the signal L is converted into a
 pseudostereophonic signal LS , C_1 and similarly the
 15 signal R is converted into a pseudostereophonic signal
 C_2 , RS. If during these conversions care is given to
 ensure that C_1 is approximately equal to C_2 , the
 decoded five-channel signal according to Rec. ITU-R
 BS.775-1 (see FIG. 9F) with respect to table 2 of Rec.
 20 ITU-R BS.775-1 (see FIG. 10F) after being downmixed to
 the stereo 2/0 format is approximately identical to the
 (amplified) signal L, R, and after being downmixed to
 the mono 1/0 format is approximately identical to the
 (amplified) signal M. For example, an existing five-
 25 channel signal according to Rec. ITU-R BS.775-1 can
 thus be converted according to the disclosure of the
 invention on the basis of a monophonic signal with the
 addition of the respective filter parameters f (resp.
 n) resp. φ resp. α resp. β resp. λ resp. ρ or $1/\lambda$ or
 30 $1/\tau$ resp. λ' resp. s for L, R resp. LS , C_1 resp. C_2 , RS,
 and this therefore results in bandwidth savings of
 nearly 80%!

A first variant embodiment, given by way of
 example, for decoding a five-channel signal according
 35 to Rec. ITU-R BS.775-1 with a good center sound
 emphasis is illustrated in FIG. 7F. In this case, the
 mono signal M is used directly as center channel C and

from the latter, by means of an inventive arrangement according to EP1850639 or W02009/138205 or W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also
 5 FIG. E5 bzw. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D
 10 resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also FIG. E10 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or
 15 also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D, is converted into a pseudostereophonic signal L, R. Similarly, the signal M resp. C is converted into a pseudostereophonic signal LS, RS. The decoded five-channel signal according to
 20 Rec. ITU-R BS.775-1 (see FIG. 9F) with respect to table 2 of the Rec. ITU-R BS.775-1 (see FIG. 10F) after being downmixed to the stereo-2/0 format does not correspond to the (amplified) signal L, R, yet after being downmixed to the mono-1/0 format corresponds exactly to
 25 the (amplified) signal M. For example, an existing five-channel signal according to Rec. ITU-R BS.775-1 can thus be converted according to the disclosure of the invention on the basis of a monophonic signal with the addition of the respective filter parameters f
 30 (resp. n) resp. φ resp. α resp. β resp. λ resp. ρ or $1/\lambda$ or $1/\tau$ resp. λ' resp. s for L, R resp. LS, RS, and this therefore results in bandwidth savings of nearly 80%!

If less value is set on bandwidth savings, the
 35 following variant embodiment according to FIG. 8F for deriving a five-channel signal according to ITU-R BS.775-1 with a good sideways balance is for example

also possible. The signal L is directly used as signal M_1 , and from the latter, converted into a pseudo-stereophonic signal LS , C_1 by means of an inventive arrangement according to EP1850639 or W02009/138205 or
5 W02011/009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 bzw. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D
10 resp. FIG. 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or
15 also FIG. E10 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D. Similarly, the signal R is directly used as signal M_2 , and from the latter
20 converted in the same way into a pseudostereophonic signal C_2 , RS . If during these conversions care is given to ensure that C_1 is approximately equal to C_2 , the decoded five-channel signal according to Rec. ITU-R BS.775-1 (see FIG. 9F) with respect to table 2 of Rec.
25 ITU-R BS.775-1 (see FIG. 10F) after being downmixed to the stereo 2/0 format is approximately identical to the (amplified) signal L , R , yet after being downmixed to the mono 1/0 format is approximately identical to the (amplified) signal $M_1 + M_2$. For example, an existing
30 five-channel signal according to Rec. ITU-R BS.775-1 can thus be converted according to the disclosure of the invention on the basis of a monophonic signal with the addition of the respective filter parameters f (resp. n) resp. φ resp. α resp. β resp. λ resp. ρ or
35 $1/\lambda$ or $1/\tau$ resp. λ' resp. s for. LS , C_1 resp. C_2 , RS , and this therefore results in bandwidth savings of nearly 60%!

By analogy, for n-channel systems for reproducing sounds, related arrangements can be achieved that altogether enable, by means of an inventive arrangement according to EP1850639 or W02009/138205 or W02011/

5 009649 or W02011/009650 or CH01264/10 resp. PCT/EP2011/063322 resp. FIG. E3 or also FIG. E4 or also FIG. E5 bzw. FIG. E6 or also FIG. E7 or also FIG. E8 resp. FIG. 9D or also FIG. 10D resp. FIG. 11D or also FIG. 12D resp. FIG. 13A resp. FIG. 14A resp. FIG. 13D resp. FIG.

10 14D resp. FIG. E1 or also FIG. E2 or also just described arrangements of the form of FIG. 17D resp. FIG. 19D or simplified FIG. 20D or also FIG. 21D resp. FIG. 23D or simplified FIG. 24D or also FIG. 25D resp. FIG. 27D or simplified FIG. 28D resp. FIG. E9 or also

15 FIG. E10 or simplified FIG. E11 resp. FIG. E12 resp. FIG. E13 or simplified FIG. E14 resp. FIG. E15 or also FIG. E16 or simplified FIG. E17 resp. FIG. 18D resp. FIG. 22D resp. FIG. 26D, a coding on the basis of one or several signals with the addition of the respective

20 filter parameters f (resp. n) resp. ϕ resp. α resp. β resp. λ resp. ρ or $1/\lambda$ or $1/\tau$ resp. λ' resp. s .

N.B. A series of elements of the described circuits can easily be exchanged, arranged differently, can be grouped into a single element or separated into

25 several elements, based on individual parameters of the original element, including a series of trivial variations such as the cascading of several amplifiers etc. These variant embodiments, despite having not been explicitly mentioned, are part of the object of the

30 invention.

N.B. Represented inventive arrangements or methods can additionally use prior art compression algorithms, data reduction methods etc. resp. the analysis of characteristic features, such as the minima

35 or maxima for the faster evaluation in accordance with

the invention of existing or generated signals or signal portions.

Bibliographical references

1. Tony Hirvonen, Athanasios Mouchtaris: On
5 the Multichannel Sinusoidal Model for Coding Audio
Object Signals. – AES Convention Paper 8418.
2. Rec. ITU-R BS. 775-1

Patent claims

1. Device for stereophonizing a mono signal resp. for obtaining pseudostereophonic signals, characterized by:

5 means for calculating time differences between channels;

 means for calculating new time differences by multiplying said time differences with a time parameter which is greater than zero;

10 means for the stereophonization of a mono signal to be rendered stereophonic on the basis of the new time differences.

2. Device according to claim 1, characterized in that said time differences are calculated on the basis
15 of an angle ascertained or stipulated manually or by metrology and enclosed by a sound source and a main axis of the mono signal to be rendered stereophonic.

3. Device according to one of the claims 1 to 2, characterized in that said time parameter is between 29
20 milliseconds and 146 milliseconds, preferably 100 milliseconds.

4. Device according to one of the claims 1 to 3, characterized in that said time parameter can be chosen freely by a user.

25 5. Device according to one of the claims 1 to 4, characterized in that a parameter describing a directional pattern of a signal that is to be rendered stereophonic, or an angle ascertained or stipulated manually or by metrology and enclosed by the main axis
30 and the sound source, or a fictitious left opening angle or a fictitious right opening angle or an attenuation or an attenuation or amplification factors

or a degree of correlation or a parameter for defining
 an allowable value range or a limit value for
 determining resp. maximizing an absolute value or a
 deviation or a parameter defining a localisation
 5 direction or a limit value for selecting an image width
 or a deviation or a limit value for optimizing the
 function values with respect to an image width or a
 deviation for forming a resulting stereo signal or said
 time parameter can be optimized automatically or
 10 interactively.

6. Device according to one of the claims 1 to 5,
 characterized in that either:

- one stereo decoder for the pseudostereo
 conversion and two panorama potentiometers connected
 15 downstream thereto are passed through, wherein each
 panorama potentiometer forms two collective buses; or:

- one stereo decoder for the pseudostereo
 conversion and one amplifier, connected upstream of the
 stereo decoder, for amplifying an input signal from the
 20 stereo decoder are passed through; or:

- one stereo decoder for the pseudostereo
 conversion and respectively one amplifier connected
 upstream of the stereo decoder for each input signal of
 the stereo decoder are passed through; or:

25 - a modified stereo decoder for the pseudostereo
 conversion, which contains an adder and a subtractor in
 order to add resp. to subtract input signals (M, S)
 respectively amplified by predetermined factors, is
 passed through in order to generate signals that are
 30 identical with said collective bus signals.

7. Device according to one of the claims 1 to 6, characterized by means for data compression or data reduction.

8. Device according to one of the claims 1 to 7,
5 characterized by:

- means for obtaining a first pseudostereophonic signal L, R from a mono signal M,

- means for obtaining a second pseudostereophonic signal LS, C₁ from the obtained signal L,

10 - means for obtaining a third pseudostereophonic signal C₂, RS from the obtained signal R.

9. Device according to one of the claims 1 to 8, characterized by outputs for signals for systems for reproducing sounds with n loudspeakers, wherein $n \geq 2$.

15 10. Device according to one of the claims 1 to 9, characterized by a dictionary with values for the delaying of the input signal to be rendered stereophonic, which is respectively used for the calculation of the time differences.

20 11. Device according to claim 10, characterized by means that contain the dictionary in the interval $[0, \pi/2]$ on the basis of values varied respectively by $\pi/36$.

25 12. Method for stereophonizing a mono signal resp. for obtaining pseudostereophonic signals, characterized by:

means for calculating time differences between channels;

30 means for calculating new time differences by multiplying said time differences with a time parameter

which is greater than zero;

means for the stereophonization of a mono signal to be rendered stereophonic on the basis of the new time differences

5 13. Method according to claim 12, characterized in that said time differences are calculated on the basis of an angle ascertained or stipulated manually or by metrology and enclosed by a sound source and a main axis of the mono signal to be rendered stereophonic.

10 14. Method according to one of the claims 12 to 13, characterized in that said time parameter is between 29 milliseconds and 146 milliseconds, preferably 100 milliseconds.

15 15. Method according to one of the claims 12 to 14, characterized in that said time parameter can be chosen freely by a user.

20 16. Method according to one of the claims 12 to 15, characterized in that a parameter describing a directional pattern of a signal that is to be rendered stereophonic, or the angle ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or a fictitious left opening angle or a fictitious right opening angle or an attenuation or an amplification factors
25 or a degree of correlation or a parameter for defining an allowable value range or a limit value for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation direction or a limit value for selecting an image width
30 or a deviation or a limit value for optimizing the function values with respect to an image width or a deviation for forming a resulting stereo signal or said time parameter can be optimized automatically or interactively.

17. Method according to one of the claims 12 to 16, characterized in that a parameter describing a directional pattern of a signal that is to be rendered stereophonic, or the angle ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or a fictitious left opening angle or a fictitious right opening angle or an attenuation or an amplification factors or a degree of correlation or a parameter for defining an allowable value range or a limit value for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation direction or a limit value for selecting an image width or a deviation or a limit value for optimizing the function values with respect to an image width or a deviation for forming a resulting stereo signal or said time parameter are optimized on the basis of algebraic invariants.

18. Method according to one of the claims 12 to 17, characterized in that

- a delay time is calculated by the multiplication of constants $[\sqrt{5} - 1]/2$ with said time parameter or a preset value for this time parameter, multiplied with said just mentioned constant, is used as delay time,

- an amplification factor is equal to the constant $4/5$,

- the mono signal to be rendered stereophonic is multiplied by an amplification factor $4/5$, in order to obtain a mid signal,

- the mono signal to be rendered stereophonic is delayed by a delay time in order to obtain a side signal,

- the stereo decoding of the mid and side signal into a stereo signal takes place; or alternatively: the stereo decoding of the mid and side signal into a stereo signal on the basis of a modified stereo decoder takes place, which contains an adder and a subtractor in order to add resp. to subtract input signals respectively amplified by predetermined factors.

19. Method according to one of the claims 12 to 18, characterized in that

10 - a delay time is calculated by the multiplication of constants $[\sqrt{5} - 1]/2$ with said time parameter or a preset value for this time parameter, multiplied with said just mentioned constant, is used as delay time,

15 - an amplification factor is equal to the constant $5/4$,

- the mono signal to be rendered stereophonic is used directly as mid signal,

- the mono signal to be rendered stereophonic is delayed by a delay time and subsequently amplified by an amplification factor $5/2$ in order to obtain a side signal; or alternatively: the mono signal to be rendered stereophonic is amplified by an amplification factor $5/2$ and subsequently delayed by a delay time in order to obtain a side signal; or alternatively: a main signal to be rendered stereophonic is amplified by an amplification factor P and subsequently delayed by a delay time and subsequently amplified by an amplification factor Q in order to obtain a side signal, wherein for the amplification factors P and Q the relation $P * Q = 5/2$ applies,

- the stereo decoding of the mid and side signal into a stereo signal takes place; or alternatively: the stereo decoding of the mid and side signal into a stereo signal on the basis of a modified stereo decoder takes place, which contains an adder and a subtractor in order to add resp. to subtract input signals (M, S) respectively amplified by predetermined factors.

20. Method according to one of the claims 12 to 19, characterized in that the side signal prior to passing through the stereo decoder additionally appears amplified by an attenuation or an amplification factor.

21. Method according to one of the claims 12 to 20, characterized in that the mid signal prior to passing through the stereo decoder additionally appears amplified by a reciprocal attenuation $1/\lambda$ or an amplification factor $1/\tau$, wherein $1 \geq \tau + \lambda' \geq 0$ and $\lambda = \tau + \lambda'$.

22. Method according to one of the claims 12 to 21, characterized in that either:

- one stereo decoder for the pseudostereo conversion and two panorama potentiometers connected downstream thereto are passed through, wherein each panorama potentiometer forms two collective buses; or:

- one stereo decoder for the pseudostereo conversion and one amplifier, connected upstream of the stereo decoder, for amplifying an input signal from the stereo decoder are passed through; or:

- one stereo decoder for the pseudostereo conversion and respectively one amplifier connected upstream of the stereo decoder for each input signal of the stereo decoder are passed through; or:

- a modified stereo decoder for the pseudostereo conversion, which contains an adder and a subtractor in order to add resp. to subtract input signals (M, S) respectively amplified by predetermined factors, is
5 passed through in order to generate signals that are identical with said collective bus signals.

23. Method according to one of the claims 12 to 22, characterized in that an automatic or interactive optimization of a parameter describing a directional
10 pattern of a signal that is to be rendered stereophonic, or of said angle ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or of a fictitious left opening angle or of a fictitious right opening angle or
15 of an attenuation or of amplification factors or of a degree of correlation takes place on the basis of one or several weighting functions.

24. Method according to one of the claims 12 to 23, characterized in that the optimization of a parameter
20 describing the directional pattern of a signal that is to be rendered stereophonic, or of said angle ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or of a fictitious left opening angle or of a fictitious right
25 opening angle or of an attenuation or of amplification factors or of a degree of correlation or of a parameter for defining an allowable value range or a limit value for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation
30 direction or a limit value for selecting an image width or a deviation or a limit value for optimizing the function values with respect to an image width or a deviation for forming a resulting stereo signal or said time parameter takes place on the basis of the first
35 main reflection of an existing stereophonic or pseudostereophonic signal.

25. Method according to one of the claims 12 to 24, characterized in that the optimization of a parameter describing the directional pattern of a signal that is to be rendered stereophonic, or of said angle

5 ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or of a fictitious left opening angle or of a fictitious right opening angle or of an attenuation or of amplification factors or of a degree of correlation or of a parameter

10 for defining an allowable value range or a limit value for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation direction or a limit value for selecting an image width or a deviation or a limit value for optimizing the

15 function values with respect to an image width or a deviation for forming a resulting stereo signal or said time parameter can be stipulated or can be influenced by the user with respect to the characteristics of the achieved reverberation or of the achieved first main

20 reflection.

26. Method according to one of the claims 12 to 25, characterized in that an optimization of a parameter describing the directional pattern of a signal that is to be rendered stereophonic, or of said angle

25 ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or of a fictitious left opening angle or of a fictitious right opening angle or of an attenuation or of amplification factors or of a degree of correlation or of a parameter

30 for defining an allowable value range or a limit value for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation direction or a limit value for selecting an image width or a deviation or a limit value for optimizing the

35 function values with respect to an image width or a deviation for forming a resulting stereo signal or said time parameter takes place on the basis of an operator

containing the specific transfer functions for the formation of the first main reflection from the stereophonic or pseudostereophonic signal delayed by the delay time.

5 27. Method according to one of the claims 12 to 26, characterized in that an optimization of a parameter describing the directional pattern of a signal that is to be rendered stereophonic, or of said angle
10 ascertained or stipulated manually or by metrology and enclosed by the main axis and the sound source, or of a fictitious left opening angle or of a fictitious right opening angle or of an attenuation or of amplification factors or of a degree of correlation or of a parameter for defining an allowable value range or a limit value
15 for determining resp. maximizing an absolute value or a deviation or a parameter defining a localisation direction or a limit value for selecting an image width or a deviation or a limit value for optimizing the function values with respect to an image width or a
20 deviation for forming a resulting stereo signal or of said time parameter takes place on the basis of a dictionary or of a dictionary of operators.

28. Method according to one of the claims 12 to 27, characterized by the additional use of compression
25 algorithms or data reduction methods.

29. Method according to one of the claims 12 to 28, characterized in that at least one all-pass filter of first or second or n^{th} order is used.

30 30. Method according to one of the claims 12 to 29, characterized in that at least one phase shifter is used.

31. Method according to one of the claims 12 to 30, characterized in that

- a first pseudostereophonic signal is obtained from a mono signal,

- a second pseudostereophonic signal is obtained from an obtained left signal,

5 - a third pseudostereophonic signal is obtained from an obtained right signal; wherein optionally:

- the obtained signals are matched to one another.

32. Method according to one of the claims 12 to 31, characterized in that

10 - a mono signal is used directly as signal,

- from the mono signal a first pseudostereophonic signal is obtained,

- from the mono signal a second pseudostereophonic signal is obtained.

15 33. Method according to one of the claims 12 to 32, characterized in that

- from the left channel of a stereo signal a first pseudostereophonic signal is obtained,

20 - from the right channel of a stereo signal a second pseudostereophonic signal is obtained; wherein optionally:

- the obtained signals are matched to one another.

25 34. Method according to one of the claims 12 to 33, characterized in that signals for systems for reproducing sound with n loudspeakers, $n \geq 2$, are created.

35. Method according to one of the claims 12 to 34,
characterized in that instead of the respective
calculation of the time differences or of the
amplification factors, if necessary taking into account
5 the inversely proportional attenuations or the
amplification factors, a dictionary with corresponding
values for the delaying or amplifying of the input
signal to be rendered stereophonic is used.

36. Method according to one of the claims 12 to 35,
10 characterized in that the dictionary is created in the
interval $[0, \pi/2]$ on the basis of values varied
respectively by $\pi/36$ from a fictitious opening angle or
from said angle ascertained or stipulated manually or
by metrology.