

**Device and method for optimizing stereophonic or
pseudo-stereophonic audio signals**

The invention relates to audio signals and apparatuses
5 or methods for the generation, transmission, conversion
and reproduction thereof.

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

Such decorrelated signals can firstly be generated by
15 differently positioned sound transducer systems, the
signals from which are optionally postprocessed, or can
be generated by means of what are known as
pseudo-stereophonic techniques, which produce such
suitable decorrelation - on the basis of a mono signal.

20 EP2124486 and EP1850639 describe, by way of example, 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
25 of the microphone and the directional axis 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
30 system). The content of EP2124486 and of EP1850639 is
hereby introduced as a reference.

EP0825800 (Thomson Brandt GmbH) proposes the formation
of different kinds of signals from a mono input signal
35 by means of filtering, which signals are used - for
example by using a method proposed by Lauridsen based
on amplitude and time difference corrections, depending
on the recording situation - to generate virtual

single-band stereo signals separately, these subsequently being combined to form two output signals.

US5173944 (Begault Durand) applies HRTFs (Head Related
5 Transfer Functions) which correlate with 90, 120, 240
and 270 degrees azimuth, respectively, to the varyingly
delayed but uniformly amplified monophonic input
signal, the signals formed in turn finally being
superimposed on the original mono signal. In this case,
10 the amplitude correction and the time difference
corrections are chosen independently of the recording
situation.

The earlier patent application CH01159/09 on behalf of
15 the patent applicant (filed on 07-22-2009) proposes the
ostensibly not purposeful downstream connection of one
or more panoramic potentiometers or equivalent means in
an apparatus according to EP2124486 or EP1850639 after
MS matrixing has taken place (after an MS matrix, for
20 which the relation

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

and

25

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

applies, has been passed through), which do not - as in
the case of intensity stereophonic signals, that is to
30 say for stereo signals which differ exclusively in
terms of their levels but not in terms of time or phase
differences or different frequency spectra - result in
the intended narrowing of the mapping width or the
intended shifting of the mapping direction of the
35 obtained stereo signals, but instead rather result in
the degree of correlation being increased or lowered.

In the case of the configuration according to
EP2124486, according to EP1850639 and/or according to

CH01159/09, different parameters may be chosen in the stereo converter which are used to generate pseudo-stereophonic signals. Though often plural parameters or plural sets of parameters may be used to
5 in order to obtain pseudo-stereophonic audio signals, the 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 not trivial, however.

10 Furthermore, the adjustment of the parameters also frequently has an impact on the degree of correlation between the left channel and the right channel. As part of the invention, however, it has been found that it
15 would make sense to stipulate a uniform degree of correlation for the evaluation of different parameterizations for ϕ or f (or, respectively, the simplifying parameter n), α , β .

20 It therefore is an aim of the present invention to provide a new method and a new apparatus for obtaining pseudo-stereophonic signals.

It is, in particular, an aim to provide a new method
25 and a new apparatus for automatically and optimally choosing such parameters which form the basis for the generation of stereophonic or pseudo-stereophonic signals.

30 It is also an aim to provide a method and an apparatus for optimally and automatically determining particularly the parameters (ϕ , λ , ρ or f (or, respectively, n), α , β) while generating said stereophonic or pseudo-stereophonic signals.

35 Said method and said apparatus are intended to be used to select, from a plurality of decorrelated, in particular pseudo-stereophonic, signal variants, those

whose decorrelation is found to be particularly beneficial.

5 In particular, the selection criteria themselves are intended to be able to be influenced in an as efficient and compact a form 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.

10

According to one aspect, an apparatus and a method for obtaining pseudo-stereophonic output signals $x(t)$ and $y(t)$ by using an MS matrix are therefore proposed, wherein $x(t)$ is the function value of the resulting
15 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.

20

If there are dropouts or similar defects, however, there may be an insignificant quantity of single points outside of the definition range. In this case, the obtainment is iteratively optimized until a portion of
25 $\langle x(t), y(t) \rangle$ is within the predetermined definition range. Since this portion usually differs from the whole only insignificantly on account of dropouts or similar defects, this apparatus must also be covered as equivalent by the scope of protection of the patent
30 claims.

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
35 range can be usefully stipulated by the inequalities

$$|\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\}|$$

and

$$|\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}| \leq |\sin \arg\{f^*[x(t)] + g^*[y(t)]\}|,$$

5 for example, with the relations

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

and

10

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

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

15

A person skilled in the art would, by way of example,
also advantageously stipulate such a parameter a or,
respectively, such a definition range by means of the
inequality

20

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

where $f^*[x(t)]$ and $g^*[y(t)]$ are again the above complex
transfer functions of the output signal $x(t)$, $y(t)$, and

25 $0 \leq a \leq 1$ is true.

In both cases, the user can arbitrarily stipulate such
a definition range, on the basis of the unit circle of
the complex number plane or of the imaginary axis (if
30 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, explained by using two examples, also
35 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).

5

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

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

25

In one embodiment, a method for obtaining pseudo-stereophonic output signals $x(t)$ and $y(t)$ by using a converter is therefore proposed,
wherein $x(t)$ is the function value of the resulting
30 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:

35

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

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

in which the obtainment is iteratively optimized until the following criteria are satisfied:

$$5 \quad |\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\}|,$$

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

$$10 \quad |\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}| \leq |\sin \arg\{f^*[x(t)] + g^*[y(t)]\}|.$$

In another embodiment, a person skilled in the art would advantageously replace these criteria with the criterion

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

A remarkable aspect of the methods for obtaining pseudo-stereophonic signals according to EP2124486 or
20 according to EP1850639 is the fact that they always provide a perfect middle signal. For this reason, the short-time cross correlation

$$25 \quad (1) \quad r = (1/2T) * \int_{-T}^T x(t)y(t) dt \\ * (1/x(t)_{\text{eff}}y(t)_{\text{eff}})$$

is introduced here for the time interval $[-T, T]$ and
30 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
35 different parameterizations for ϕ or f (or, respectively, n), α , β . For this purpose, the invention therefore involves the degree of correlation between the output signals ($x(t)$ and $y(t)$) being normalized. This normalization can preferably be stipulated by

means of the specific variation of λ (left attenuation) or ρ (right attenuation).

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

It also makes sense if a uniform level for the maximum
10 of the resulting left channel and of the resulting right channel is being attained for extremely different parameterizations for ϕ or f (or, respectively, n), α , β . For this purpose, the invention therefore involves the level of the maximum of the resulting left channel
15 and of the resulting right channel being normalized, as a result of which this level is not influenced by the optimization of the parameters.

By way of example, it makes sense for, initially, the
20 modulation for the maximum of the left signal L and of the right signal R to be uniformly confined to 0 dB, for example, by means of a first logic element.

It also makes sense if - for extremely different
25 parameterizations for ϕ or f (or, respectively, n), α , β - criteria which are dependent on $\langle x(t), y(t) \rangle$ or on $|\langle x(t), y(t) \rangle|$ are used for stipulation. For this purpose, the invention therefore involves a respective corresponding range of values which is normalized, so
30 as to be 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 $f^*[x(t)] + g^*[y(t)]$
35 can now be analyzed in more detail in order to draw conclusions concerning the quality of the respective output signal from an apparatus according to EP2124486 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 converter is therefore optimized according to the cited criteria for $|\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}|$ and for $|\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}|$, for example.

This method is found to be particularly beneficial, since a single parameter, namely a , takes optimum account of, in particular, the different nature of the output signals from an apparatus or a method according to EP2124486 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 a preferably needs to be restricted significantly due to disturbing artifacts such as high-frequency sidetone during the articulation.

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

If the signals $x(t)$, $y(t)$ do not satisfy the aforementioned constraints, the invention involves optimization being carried out by redetermining the parameters ϕ or f (or, respectively, n) or α or β - according to an iterative procedure that is matched with the function values $x[t(\phi, f, \alpha, \beta)]$ and $y[t(\phi, f, \alpha, \beta)]$ or, respectively, $x[t(\phi, n, \alpha, \beta)]$ and $y[t(\phi, n, \alpha, \beta)]$ - whilst executing steps presented hitherto until $x(t)$ and $y(t)$ meet the aforementioned constraints.

In a further step, by way of example, the relief of the function $f^*[x(t)] + g^*[y(t)]$ 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

$$\begin{aligned} & T \\ 5 \quad (6) \quad & \int_{-T}^T |f^*[x(t)] + g^*[y(t)]| dt; \\ & -T \end{aligned}$$

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

$$\begin{aligned} 10 \quad & T \\ (7) \quad & \int_{-T}^T |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\} \\ & -T \\ & + i * \sin \arg\{f^*[x(t)] + g^*[y(t)]\}| dt \end{aligned}$$

15 By way of example, a person skilled in the art would also advantageously replace (7) with

$$\begin{aligned} & T \\ 20 \quad (7a) \quad & \int_{-T}^T a * \{1 / \sqrt{1 - (1 - a^2) * \sin^2 \arg\{f^*[x(t)] \\ & -T \\ & + g^*[y(t)]\}}\} dt. \end{aligned}$$

25 In this case too, the user is provided with a tool in so far as he has a free choice of the limit value R^* (or the deviation Δ defined by the inequality (8), see below) for this maximization within the context of (8). Overall, the following constraint must be met for the

30 total number of possible signal variants $x_j(t)$, $y_j(t)$:

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

$$\begin{aligned} 35 \quad & T \\ & \leq \max \int_{-T}^T |f^*[x_j(t)]| \\ & \{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi \quad -T \end{aligned}$$

$$+ g^*[y_j(t)]| dt$$

$$\leq R^* + \Delta$$

5

$$\begin{aligned} & \leq \int_{-T}^T |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\} \\ & + i * \sin \arg\{f^*[x(t)] + g^*[y(t)]\}| dt. \end{aligned}$$

10

A person skilled in the art would in turn advantageously replace (8) with

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

$$\begin{aligned} & \leq \max_{\{f^*[x_j(t)], g^*[y_j(t)]\} \in \Phi} \int_{-T}^T |f^*[x_j(t)] \\ & + g^*[y_j(t)]| dt \end{aligned}$$

20

$$+ g^*[y_j(t)]| dt$$

$$\leq R^* + \Delta$$

25

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

30

R^* and Δ are directly related to the loudness of the output signal that is to be attained (that is to say to those parameters which the listener also takes as a

basis for assessing the validity of a stereophonic map).

If the neighborhood of the limit value R^* , defined by
5 Δ , or the maximum of all possible integrated reliefs is
not reached, optimization, with a view to the limit
value R^* and the deviation Δ or to the aforementioned
maximum - in accordance with an iterative procedure
that is matched with the function values $x[t(\phi, f, \alpha, \beta)]$ and $y[t(\phi, f, \alpha, \beta)]$ or, respectively, $x[t(\phi, n, \alpha, \beta)]$ and $y[t(\phi, n, \alpha, \beta)]$ -, involves new parameters ϕ
10 or f or α or β being determined, and whilst executing
all steps illustrated hitherto until signals $x(t)$, $y(t)$
or parameters ϕ or λ or ρ or f (or, respectively, n) or
15 α or β result, which correspond to optimum
stereophonization.

With an appropriate choice of degree of correlation r ,
of parameter a - stipulating the desired respective
20 definition range - and of 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.

25 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
30 also possible to normalize the axis length in order to
reduce the computing time accordingly.

According to one aspect, it is recommended practice to
use (inherently known) compression algorithms or data
35 reduction methods or to analyze characteristic features
such as the minima or maxima for the
pseudo-stereophonic signals obtained according to
EP2124486 or EP1850639, this being the case in order to

speed up the evaluation thereof according to the invention.

5 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 computing time is significantly reduced as a result.

10 The invention can incidentally be applied to apparatuses or methods which generate stereophonic signals which are reproduced by more than two loudspeakers (for example surround sound systems belonging to the prior art).

15 According to one aspect, the invention involves the cascaded downstream connection of a plurality of means (for example logic elements), some of the parameters of which can be aligned, with an MS matrix (for example according to EP2124486 or EP1850639), wherein feedback
20 for said apparatuses or methods involves the parameters ϕ or λ or ρ or f (or, respectively, n) or α or β being changed in an optimized way until all constraints of the logic elements are met.

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

Brief Description of the Figures

30

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

35 FIGURE 1 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 an MS matrix (for example an MS matrix according to EP2124486 or EP1850639), whereas 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 FIGURE 7, which is optionally also connected downstream to figure 6b.

5

FIGURE 2 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 or ascertains the argument of the sum thereof
10 $f^*[x(t)] + g^*[y(t)]$.

FIGURE 3 shows a first example of a circuit for selecting the definition range by using the parameter a .

15

FIGURE 3a shows a second example - which is advantageous to a person skilled in the art - of a circuit for selecting a fresh definition range by using the parameter a .

20

FIGURE 4 shows a first example of a circuit for a third logic element which checks the signals, which are generated in FIGURE 1 and which are mapped on the complex number plane as shown in FIGURE 2, for the
25 admissible definition range, defined by the parameter a , according to the constraints
 $|\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a| * \cos \arg\{f^*[x(t)] + g^*[y(t)]\}$ and
 $|\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a| * \sin \arg\{f^*[x(t)] + g^*[y(t)]\}$
30 $||$.

FIGURE 4a shows a second example - which is advantageous to a person skilled in the art - of a circuit for a third logic element which checks the
35 signals, which are generated in FIGURE 1 and which are mapped on the complex number plane as shown in FIGURE 2, for the admissible definition range, freshly defined by the parameter a as shown in FIGURE 3a, according to the constraint

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

FIGURE 5 shows an example of a circuit for a fourth
 5 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, whereas the
 user has a free choice of limit value R^* defined by the
 inequality (8) (or of deviation Δ , likewise defined by
 10 the inequality (8)) for this maximization.

FIGURE 5a shows a second example - which is
 advantageous for a person skilled in the art - of a
 circuit for a fourth logic element which finally
 15 analyzes the relief of the function $f^*[x(t)] + g^*[y(t)]$
 for the purpose of maximizing the function values
 thereof, whereas the user has a free choice of limit
 value R^* defined by the inequality (8a) (or of
 deviation Δ , likewise defined by the inequality (8a))
 20 for this maximization.

FIGURE 6a shows an input circuit for an already
 existing stereo signal prior to transfer to a circuit
 as shown in FIGURE 6b for determining the localization
 25 of the signal.

FIGURE 6b shows a circuit for determining the
 localization of the signal, the inputs of which circuit
 are connected to the outputs in FIGURE 5 or,
 30 respectively, FIGURE 5a or, respectively, to the
 outputs in FIGURE 6a.

FIGURE 7 shows a further example of a circuit for
 normalizing stereophonic or pseudo-stereophonic signals
 35 which, when connected downstream to FIGURE 6b, 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 FIGURE 1 when the parameter z is transferred.

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

FIGURE 9 shows an example of a circuit for adjusting
the mapping width of an audio signal.

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Detailed Description

For a stereo converter, for example in an apparatus
according to EP2124486 or EP1850639 - for the case of
15 identical inversely proportional attenuations λ and ρ -
optimized parameters φ , λ , f (or, respectively, the
simplifying parameter n), α , β are to be determined in
order to convert a mono signal into corresponding
pseudo-stereophonic signals which have optimum
20 decorrelation and loudness (the two criteria according
to which the listener assesses the quality of a stereo
signal). Such determination is intended to be achieved
with as few technical means as possible.

25 FIGURE 1 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 converter with an MS
matrix 110 (for example a stereo converter according to
30 EP2124486 or EP1850639), whereas the input signal M and
 S can (prior to passing through an amplifier connected
upstream to the MS matrix) optionally be fed to a
circuit as shown in FIGURE 7, which is optionally and
ideally connected downstream to FIGURE 6b, and is
35 activated as soon as the parameter z resulting from
FIGURE 6b 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 modulation, showing the maximum of 0 dB, of the left channel L and the right channel R.

- 5 The signals L and R resulting from the arrangement 110 (for example an MS matrix according to EP2124486 or EP1850639) are amplified uniformly by the factor ρ^* (amplifiers 118, 119) such that the maximum of both signals has a level of exactly 0 dB (normalization on
10 the unit circle of the complex number plane). This is achieved, by way of example, by the downstream connection of a logic element 120 which uses the feedbacks 121 and 122 and variation or correction of the gain factor ρ^* of the amplifiers 118 and 119 to
15 cause a modulation of the maximum value of L and R to reach 0 dB.

The resulting stereo signals $x(t)$ (123) and $y(t)$ (124), the amplitudes of which are directly proportional to L
20 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

$$\begin{aligned} & T \\ 25 \quad (1) \quad r &= (1/2T) * \int_{-T}^T x(t)y(t) dt * (1/x(t)_{eff}y(t)_{eff}). \\ & -T \end{aligned}$$

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

30

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.

- 35 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 modulation 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.

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

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

FIGURE 2 clarifies the circuit principle which maps the input signals $x(t)$, $y(t)$ on the complex number plane or determines the argument of the sum thereof
15 $f^*[x(t)] + g^*[y(t)]$. Within this circuit the resulting signals $x(t)$ and $y(t)$ from the output of FIGURE 1 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
20 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 . Therefore, the transfer functions

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

and

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

are obtained

30

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 $f^*[x(t)] + g^*[y(t)]$.

35

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

FIGURE 3 clarifies the circuit principle for selecting the definition range, whereas continuous regulation is made possible by means of the parameter $0 \leq a \leq 1$, on the basis of the unit circle of the complex number plane or of the imaginary axis. The user can therefore determine the definition range a on the complex number plane. For this, the cosine (333) and sine (334) of the argument which has just been determined for $f^*[x(t)] + g^*[y(t)]$ are calculated. The signal resulting from 333 is then fed to an amplifier 335 and is amplified by the gain factor $0 \leq a \leq 1$, such gain factor being freely selectable by the user.

FIGURE 4 shows the circuit principle for the third logic element, which checks the signals, which are generated in FIGURE 1 and which are mapped on the complex number plane as shown in FIGURE 2, according to the constraints

$$(4) \quad |\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\}|$$

and

$$(5) \quad |\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}| \leq |\sin \arg\{f^*[x(t)] + g^*[y(t)]\}|.$$

The real part and the imaginary part of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ and the signals resulting from 334 and 335 are in this case fed to a further logic element 436, which checks whether the criteria (4) and (5) are satisfied, hence whether the values of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ are within the range of values defined by the user by means of a .

If this is not the case, a feedback 437 is used to determine new optimized values φ or f (or,

respectively, n) or α or β , and the entire system described hitherto is passed through again until the values of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ are within the range of values defined by the user by means of a . The output signals for the logic element 436 are now transferred to the last logic element 538 (FIGURE 5).

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

15

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

20

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

25

$$\leq R^* + \Delta$$

30

$$\begin{aligned} & \leq \int_{-T}^T |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\} \\ & + i * \sin \arg\{f^*[x(t)] + g^*[y(t)]\}| dt \end{aligned}$$

must be met. If this is not the case, a feedback 539 is used to iteratively determine new optimized values ϕ or f (or, respectively, n) or α or β , and the entire system described hitherto 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 account of the limit value R^* or the deviation Δ , both defined by the user.

5 An alternative circuit principle which is advantageous to a person skilled in the art is clarified by FIGURES 3a, 4a and 5a, which replace the corresponding FIGURES 3, 4 and 5 in a preferred variant:

10 FIGURE 3a in turn allows the selection of a new definition range by means of the parameter a , $0 \leq a \leq 1$, wherein a is used to allow continuous regulation, on the basis of the unit circle of the complex number plane or of the imaginary axis. The user
15 can therefore freely stipulate the definition range determined by 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 resulting from 333a is then
20 fed to an amplifier 335a and is amplified by the gain factor $1/a^2$, which is freely selectable by the user. In addition, the squared sine of the argument of the sum of the transfer functions $f^*[x(t)] + g^*[y(t)]$ is calculated.

25

FIGURE 4a, which is intended to be connected downstream to the output of FIGURE 3a, shows a circuit principle - which is advantageous to a person skilled in the art - for a new third logic element, which checks the
30 signals, which are generated in FIGURE 1 and which are mapped on the complex number plane as shown in FIGURE 2, according to the simplified constraint

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

35

The squared real part and the 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 or f (or, respectively, n) or α or β , and the entire system described hitherto 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 (FIGURE 5a).

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

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

-T

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

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

15 Hence, the original pseudo-stereo converter, for example according to one of the embodiments in EP2124486 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 or f (or, respectively, n) or α or β until $x(t)$ and $y(t)$ meet the aforementioned constraints (4), (5) and (8) or (4a) and (8a).

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

Stipulation of the Mapping Direction

35 Occasionally, it is also important to mirror the stereophonic mapping obtained about the main axis of the directional pattern on which the stereophonic processing 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 mapping direction can also be ascertained automatically on behalf of the phantom sources generated by means of the illustrated pseudo-stereophonic methodology, by way of example, as is shown in FIGURE 6b (which is directly connected downstream to FIGURE 5 or FIGURE 5a, whereas FIGURE 6a may likewise be added to FIGURE 6b 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 $f^*(x(t_i)) + g^*(y(t_i))$ or, respectively, $f^*(l(t_i)) + g^*(r(t_i))$ 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 FIGURE 2 is compared with the transfer function $f^*(l(t_i)) + g^*(r(t_i))$ of the left signal $l(t)$ and the right signal $r(t)$ of the original stereo signal L° , R° (which transfer function is ascertained by using the circuit shown in FIGURE 6a, the design of which corresponds to the first part of the circuit for the input signals $x(t)$, $y(t)$ in FIGURE 2). 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 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 ascertained) stipulatable number b , which should be less than or equal to the number of correlating function values of the transfer functions $f^*(x(t_i)) + g^*(y(t_i))$ and $f^*(l(t_i)) + g^*(r(t_i))$ 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 FIGURES 1-5 or FIGURES 1, 2, 3a to 5a are swapped.

If an originally stereophonic signal is to be encoded into a mono signal plus the function f describing the directional pattern (or, respectively, 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 and the resulting right channel need to be swapped (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 FIGURE 7).

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

25

Narrowing or expanding of the mapping width

For this application too, the additional use of compression algorithms or data reduction methods which are part of the prior art or the consideration of characteristic features, such as the minima or maxima for the pseudo-stereophonic signals obtained, is recommended in order to speed up evaluation thereof in accordance with the invention.

35

Of particular interest (for example for reproducing stereophonic signals in automobiles) is the subsequent narrowing or expanding of the mapping width of the stereo signal obtained by using the specific variation

of the degree of correlation r of the resulting stereo signal or, respectively, the attenuations λ or else ρ (for forming the resulting stereo signal). The previously determined parameters f (or, respectively, 5 n) which describe the directional pattern of the signal that is to be stereophonized, the angle α - 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 10 β can be retained in this case, and it makes sense that now only final amplitude correction is necessary, for example as per the logic element 120 in FIGURE 1, provided that this narrowing or expanding of the mapping width is performed manually.

15

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

$$20 \quad (9) \quad 0 \leq S^* - \varepsilon \leq \max | \operatorname{Re} \{ f^*[x(t)] + g^*[y(t)] \} | \leq S^* + \varepsilon \leq 1$$

and also on the criterion

25

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

30 (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 now necessary to determine only suitable function values $x(t)$, $y(t)$ which are dependent on the degree of 35 correlation r of the resulting stereo signal or, respectively, on the attenuations λ or else ρ (for the formation of the resulting stereo signal) or, where required, on a logic element which is identical to the logic element 120 in FIGURE 1, in accordance with an

iterative operating principle which is based on feedback.

5 The arrangement according to the invention in FIGURES 1 to 5, 6a and 6b or FIGURES 1, 2, 3a to 5a, 6a, 6b can accordingly be enhanced within the context of an arrangement, for instance, of the form shown in FIGURES 7, 8 and/or 9. FIGURE 7 thereby shows a further example of a circuit for normalizing stereophonic or
10 pseudo-stereophonic signals which, when connected downstream to FIGURE 6b, 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
15 FIGURE 1 when the parameter z is transferred, and the input signals in FIGURE 1 are transferred directly as input signals to FIGURE 7 at the time of this transfer.

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

In the present arrangement, the left channel and the right channel are swapped in the MS matrix 110 by using
25 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 equal to 1, otherwise such a swap does not take place.

30 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). By way of example,
35 this is achieved by the downstream connection of a logic element 120 which uses the feedbacks 121 and 122 and variation or correction of the gain factor ρ^* of the amplifiers 118 and 119 to cause a modulation 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 FIGURE 8 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 $f^*[x(t)]$ and $g^*[y(t)]$ already mentioned in connection with FIGURE 2 are therefore 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 FIGURE 9, now needs to be connected downstream, which arrangement checks, for a limit value S^* - suitably chosen by the user with respect to the mapping width of the stereo signal that is to be achieved - or a suitably chosen deviation ε - both defined by the inequality (9) - whether the constraint

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

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 or, respectively, for the attenuations λ or else ρ (for the formation of the resulting stereo signal), and the previous steps just described, as illustrated in FIGURES 7 to 9, are performed until the above constraint (9) is met.

35

The output signals for the logic element 640 are now transferred to an arrangement, for example based on the logic element 642 in FIGURE 9. 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 mapping 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 (10), with respect to the mapping width of the stereo signal that is to be achieved. Overall, the constraint

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

must be met. If this is not the case, a feedback 643 is used to determine a new optimized value for the degree of correlation r or, respectively, for the attenuations λ or else ρ (for the formation of the resulting stereo signal), and the previous steps just described, as illustrated in FIGURES 7 to 9, 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 mapping width taking account of the limit value U^* and the deviation κ , both suitably chosen by the user.

In terms of the mapping width - determined by the degree of correlation r or, respectively, the attenuations λ or else ρ (for the formation 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

Areas of Application for the Invention

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 (or, respectively, the simplifying parameter n), α , β , λ or ρ .

35

An already existing stereo signal can be evaluated in respect of r or a or R^* or Δ or the mapping direction (or parameters S^* or ε or U^* or κ described below) and can then likewise be anew encoded as a mono signal by
5 using the parameters ϕ , f (or, respectively, n), α , β , λ or ρ in view of an apparatus or a method according to EP2124486 or EP1850639.

Similarly, the arrangement just described, to which the
10 elements below may possibly be added, can be used as a decoder for mono signals. If ϕ , f (or, respectively, n), α , β , λ or ρ or the mapping direction (for example expressed by the parameter z , which can assume the value 0 or 1) are known, such a decoder is reduced to
15 an arrangement according to EP2124486 or EP1850639.

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

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

The invention is also of particular importance in
30 connection with the obtaining of stable FM stereo 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
35 the left channel and the right channel of the original stereo signal. The complete or incomplete subchannel 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 or in order to determine or optimize the parameters f (or, respectively, n), which describe the directional pattern of the signal that is to be stereophonized, 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 formation of the resulting stereo signal or, resulting therefrom, the gain factor ρ^* in FIGURE 1 for normalizing the left channel and the 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

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

and

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

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

$$(4) \quad |\operatorname{Re}\{f^*[x(t)] + g^*[y(t)]\}| \leq |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\}|$$

and

$$(5) \quad |\operatorname{Im}\{f^*[x(t)] + g^*[y(t)]\}| \leq |\sin \arg\{f^*[x(t)] + g^*[y(t)]\}|$$

(a person skilled in the art would advantageously replace constraints (4) and (5), given the same parameter a , $0 \leq a \leq 1$, with the new constraint

$$5 \quad (4a) \operatorname{Re}^2\{f^*[x(t)] + g^*[y(t)]\} * 1/a^2 + \operatorname{Im}^2\{f^*[x(t)] + g^*[y(t)]\} \leq 1)$$

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

15

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

20

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

25

$$\leq R^* + \Delta$$

30

$$\begin{aligned} & \leq \int_{-T}^T |a * \cos \arg\{f^*[x(t)] + g^*[y(t)]\} \\ & + i * \sin \arg\{f^*[x(t)] + g^*[y(t)]\}| dt) \end{aligned}$$

(a person skilled in the art would advantageously replace constraint (8) with

35

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

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

or the mapping direction of the reproduced sound sources, for example by determining the corresponding quadrants for the function values of the aforementioned transfer functions (2) and (3) for the original stereo signal (which can be optimized by virtue of subsequent swapping of the resulting left channel and the resulting right channel, for example, see above), or the limit value S^* or the deviation ε (for which, by way of example, it must be true that

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

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

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

-T

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

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

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

List of symbols used

	φ (Phi)	Angle of incidence
	α (alpha)	Left fictitious opening angle
5	β (beta)	Right fictitious opening angle
	λ	Attenuation for the left input signal
	ρ	Attenuation for the right input signal
10	The attenuations λ and ρ can be used to adjust the degree of correlation of the stereo signal.	
	Ψ	Polar angle
	f	Radial coordinate, which describes the directional pattern of the M signal
15	$P\alpha, P\beta$	Gain factor for α and β
	$L\alpha, L\beta$	Time difference for α and β
	$S\alpha$	Simulated left signal component of the S signal
	$S\beta$	Simulated right signal component of the S signal
20	$x(t)$	Left output signal
	$y(t)$	Right output signal
	$f^*[x(t)]$	Complex transfer function
	$g^*[y(t)]$	Complex transfer function
25	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
30	R^*	Limit value for the loudness of the resulting output signals $x(t), y(t)$
	Δ	Deviation
	S^*	1st limit value for the mapping width of the resulting output signals $x(t), y(t)$
35	ε	Deviation
	U^*	2nd limit value for the mapping width of the resulting output signals $x(t), y(t)$
	κ	Deviation

Patent Claims

1. A method for obtaining pseudo-stereophonic output signals $x(t)$ and $y(t)$ by using an MS matrix (110),
5 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 , characterized in
that the obtainment is iteratively optimized until
10 $\langle x(t), y(t) \rangle$ is within a predetermined definition
range.
2. The method as claimed in claim 1, in which the
parameters ϕ or f (or, respectively, n) or α or β
15 are iteratively adjusted so that $\langle x(t), y(t) \rangle$ is
within the predetermined definition range.
3. The method as claimed in claim 1 or 2, in which
the level of the maximum of the resulting left
20 channel and the resulting right channel is
normalized or, equivalently, the axis length of
the reference system for $\langle x(t), y(t) \rangle$ is
normalized.
4. The method as claimed in one of claims 1 to 3, in
25 which the degree of correlation of the output
signals $x(t)$ and $y(t)$ is normalized.
5. The method as claimed in one of claims 1 to 4, in
30 which the obtainment is iteratively optimized by
using criteria which are dependent on $|\langle x(t),$
 $y(t) \rangle|$.
6. The method as claimed in one of claims 1 to 5, in
35 which the cited definition range is determined by
the user.
7. The method as claimed in one of claims 1 to 6, in
which the cited definition range is automatically

determined with greater constraint for speech than for music.

- 5 8. The method as claimed in one of claims 1 to 7, in
which 0 is less than or equal to the selectable
limit value R^* minus the selectable deviation Δ ,
less than or equal to the absolute value,
integrated over the time interval $[-T, T]$, of the
sum of the complex transfer functions for $x(t)$ and
10 $y(t)$, wherein the complex transfer function for
 $x(t)$ is equal to $x(t)$ divided by the square root
of 2, multiplied by the sum of -1 and the
imaginary unit, and the complex transfer function
for $y(t)$ is equal to $y(t)$ divided by the square
15 root of 2, multiplied by the sum of 1 and the
imaginary unit, less than or equal to the maximum
of all reliefs, integrated over the time interval
 $[-T, T]$, of the sums of the above transfer
functions for available paired functions $x_j(t)$ and
20 $y_j(t)$ with the total number p , less than or equal
to the limit value R^* plus the deviation Δ , less
than or equal to the parameter a , $0 \leq a \leq 1$,
integrated over the time interval $[-T, T]$, said
parameter divided by the square root of the term 1
25 minus the product of the squared sine of the
argument of the sum of the above complex transfer
functions for $x(t)$ and $y(t)$ and the term 1 minus
the square of a .
- 30 9. The method as claimed in one of claims 1 to 8,
characterized by the additional ascertainment or
stipulation of the mapping direction of an
existing stereo signal.
- 35 10. The method as claimed in one of claims 1 to 9,
characterized by the additional stipulation of the
mapping width of the resulting pseudo-stereophonic
signal by using the possible variation of the

degree of correlation (r) of such signal or of an attenuation (λ or ρ).

11. The method as claimed in one of claims 1 to 10,
5 characterized by the additional evaluation of an existing stereo signal which can be reproduced by a plurality of loudspeakers.

12. The method as claimed in one of claims 1 to 11,
10 characterized by the additional application of compression methods or data reduction methods or other selective evaluation methods, to audio signals.

13. The method as claimed in one of claims 1 to 12,
15 characterized by the additional conversion of the obtained stereophonic output signals into stereo signals which are reproduced by more than two loudspeakers.

20 14. The method as claimed in one of claims 1 to 13, characterized by the application thereof to FM stereo signals.

25 15. An apparatus for obtaining pseudo-stereophonic output signals $x(t)$ and $y(t)$ by using a converter, 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
30 output channel at the time t , characterized in that the cited converter is an iterative converter, wherein the obtainment is iteratively optimized until $\langle x(t), y(t) \rangle$ is within a predetermined definition range.

35 16. The apparatus as claimed in claim 15, in which the converter is adjusted in order to iteratively adjust the parameters ϕ or f (or, respectively, n)

or α or β until $\langle x(t), y(t) \rangle$ is within the predetermined definition range.

5 17. The apparatus as claimed in one of claims 15 and 16, having normalization means for normalizing the level of the maximum of the resulting left channel and the resulting right channel or, equivalently, for normalizing the axis length of the reference system for $\langle x(t), y(t) \rangle$.

10

18. The apparatus as claimed in one of claims 15 to 17, having normalization means for normalizing the degree of correlation of the output signals $x(t)$ and $y(t)$.

15

19. The apparatus as claimed in one of claims 15 to 18, in which the obtainment is iteratively optimized by using criteria which are dependent on $|\langle x(t), y(t) \rangle|$.

20

20. The apparatus as claimed in one of claims 15 to 19, in which the cited definition range is determined by the user.

25 21. The apparatus as claimed in one of claims 15 to 20, having means for determining the cited definition range with greater constraint for speech than for music.

30 22. The apparatus as claimed in one of claims 15 to 21, in which 0 is less than or equal to the selectable limit value R^* minus the selectable deviation Δ , less than or equal to the absolute value, integrated over the time interval $[-T, T]$,
35 of the sum of the above transfer functions for $x(t)$ and $y(t)$, less than or equal to the maximum of all reliefs, integrated over the time interval $[-T, T]$, of the sums of the above transfer functions for available paired functions $x_j(t)$ and

$y_j(t)$ with the total number p , less than or equal to the limit value R^* plus the deviation Δ , less than or equal to the parameter a , $0 \leq a \leq 1$, integrated over the time interval $[-T, T]$, said parameter divided by the square root of the term 1 minus the product of the squared sine of the argument of the sum of the above complex transfer functions for $x(t)$ and $y(t)$ and the term 1 minus the square of a .

23. The apparatus as claimed in one of claims 15 to 22 having means for additionally ascertaining or stipulating the mapping direction of a stereo signal.

24. The apparatus for obtaining pseudo-stereophonic signals as claimed in one of claims 15 to 23, characterized by means for additionally stipulating the mapping width of the resulting pseudo-stereophonic signal by using the possible variation of the degree of correlation r of such signal or of an attenuation λ or of an attenuation ρ .

25. The apparatus as claimed in one of claims 15 to 24 having means for additionally evaluating an existing stereo signal which can be reproduced by a plurality of loudspeakers.

26. The apparatus as claimed in one of claims 15 to 25, having means for compression or data reduction or other selective evaluation of audio signals.

27. The apparatus as claimed in one of claims 15 to 26, characterized by one or more converters for converting the obtained stereophonic output signals into stereo signals which are designed by more than two loudspeakers.

28. The use of the apparatus, as claimed in one of claims 15 to 27, for processing FM stereo signals.