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(54) Title: SIGNAL PICKUP WITH A VARIABLE DIRECTIVITY CHARACTERISTIC

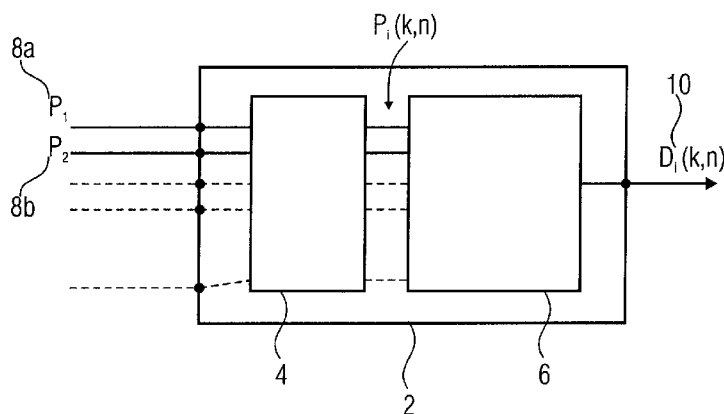


FIGURE 1

(57) Abstract: A signal processor serves to generate a substitution signal (10) having a predetermined spatial directivity characteristic while using a first signal (8a) having a known spatial directivity characteristic and a second signal (8b) having a known spatial directivity characteristic. The first and second signals are converted to a spectral representation. In a signal processor, the spectral representations of the first and second signals are combined in accordance with a combination rule so as to obtain amplitude parameters of a spectral representation of the substitution signal having a predetermined directivity characteristic (10). In accordance with the combination rule, the absolute magnitudes of amplitude parameters of the spectral representations of the first and second signals are combined, so that the predetermined directivity characteristic differs from the directivity characteristics of the first (8a) and second (8b) signals.

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Signal Pickup with a variable directivity characteristicDescription

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The present invention relates to the pickup of signals, such as audio signals, and in particular to how a substitution signal may be derived from signals recorded using a known directivity pattern, or directivity
10 characteristic, which substitution signal has a directivity characteristic which deviates from the former, so as to be able to describe a spatial characteristic of a signal filling up the space.

15 In multi-channel audio reproduction systems, a listener is surrounded by a plurality of loudspeakers. The simplest multi-channel reproduction system is a stereo setup comprising two loudspeakers. Without any additional artificial influences exerted on the sound to be
20 reproduced, a stereo system can reproduce only such sound sources with accurate locatability which are positioned on the line connecting the two loudspeakers of the stereo setup. Sound, or a signal, coming from other spatial directions cannot be reproduced correctly if it concerns
25 spatial orientation. If several loudspeakers are used which are arranged around the listener, several spatial directions may be reproduced correctly, which may result in a more natural spatial sound impression for the listener. The best-known of such multi-channel loudspeaker systems or
30 layouts of this type is the standardized 5.1 reproduction setup (ITU-R 7754-1), which consists of five loudspeakers arranged at angles of 0° , $\pm 30^\circ$ and $\pm 110^\circ$ in relation to the listening position. In addition, a series of further reproduction setups comprising different numbers of
35 loudspeakers at different positions have been proposed or used.

In order to be able to reproduce sound with correct spatial distribution, it is necessary, initially, to pick up the sound such that the spatial direction of the individual sound sources is maintained or may be reproduced. On the one hand, this may be accomplished in that the signals picked up are mixed, as early as after or during the pickup, such that, for each of the five reproduction loudspeakers of the ITU system, an audio channel is created which in the reproduction is associated with the respective loudspeaker.

In the recent past, a different, efficient way of picking up spatial audio signals has been proposed which enables reproducing the spatial impression that prevailed during the pickup with different loudspeaker systems, the relative geometric alignments of which need not be known a priori. This method is referred to as DirAC (Directional Audio Coding, Pulkki, V., "Directional audio coding in spatial sound reproduction and stereo upmixing", in Proceedings of the AES 28th International Conference, pp. 251-258, Pitea, Sweden, June 30 - July 2, 2006). As has already been mentioned, a signal picked up in such a manner may be reproduced using any loudspeaker setups. In this context, the goal is to record the signal picked up by means of DirAC such that it can be reproduced as accurately as possible by means of any multi-channel loudspeaker system, the intention being for the spatial sound impression of the location where the pickup was performed to be reproduced as accurately as possible.

In the DirAC scenario, the audio signal is recorded using an omnidirectional microphone (W) and a set of microphones which enable determining both an intensity vector of the sound field and the diffuseness of same.

In this context, the intensity vector designates the direction from which the sound picked up contributes, with maximum energy, to the signal picked up. The diffuseness

describes, in the form of a parameter, the uniformity of the spatial sound impression. If the sound is perceived with identical intensity from all directions, a case of maximum diffuseness has been achieved. However, minimum
5 diffuseness is present when the sound is perceived or recorded only from one single acoustic source from a precisely defined direction. Following the DirAC analysis, the intensity vector also points to this direction. One possibility of recording a signal in this manner consists
10 in using three microphones (X, Y, Z) having dumbbell-shaped directivity characteristics and being aligned such that their maximum directivities run parallel to the axes of a Cartesian coordinate system (see, for example: Craven G. and Gerzon M. "Coincident microphone simulation covering
15 three dimensional space and yielding various directional outputs". United States Patent 4042779). On account of the dumbbell-shaped directivity characteristic, such a microphone is also frequently referred to as a dipole (see, for example: G.W. Elko: "Superdirectional microphone
20 arrays" in S.G. Gay, J. Benesty (eds.): "Acoustic Signal Processing for Telecommunication", Chapter 10, Kluwer Academic Press, 2000, ISBN: 978-3-540-41953-2; and Merimaa, J., "Applications of a 3-D Microphone Array", in Proceedings of the AES 112th Convention, Munich, Germany,
25 May 2002).

One possibility of obtaining the signals W, X, Y and Z is to use a so-called "sound-field" microphone, which directly generates all of these signals. Alternatively, the signals
30 W, X, Y and Z may also be determined using a plurality of omnidirectional microphones arranged at different locations within the 3-dimensional space, for example at the corners of an octahedron. A virtual microphone having a directivity characteristic whose maximum points to the direction of the
35 connecting line between the two omnidirectional microphones may be formed, from two spatially separated omnidirectional microphones, in that the signals of the two microphones are subtracted from one another.

However, there are a multitude of applications wherein it is not possible to arrange the microphones in three dimensions within the space. For example, it may only be possible to arrange the microphones in a predetermined orientation on the surface of a plane. In addition, it may be required for cost reduction purposes to reduce the number of microphones used for picking up a spatial signal. This may result in that the number of microphones is smaller than would actually be necessary for picking up a signal having the components W, X, Y and Z. If a virtual sound signal which has a predetermined directivity characteristic which is not generated by physically existing microphones is to be generated in such a scenario, the signal must be generated by suitably combining the signals of existing microphones.

If signals having directivity characteristics directed to the respective spatial direction are generated in this manner for all of the relevant missing spatial directions, the DirAC algorithm may continue to be successfully employed, for example, for recording a spatial audio signal. An example of the sound pickup may be a table microphone arrangement which is normally relatively flat. Put more generally, this may be equated with a case wherein the arrangement of microphones is limited to one plane. If omnidirectional microphones are used, in such an arrangement only microphone signals with directivity characteristics in x and y directions may be generated within the plane by measurement. A further microphone may be arranged, for example, at the center of a rectangular arrangement of omnidirectional signals so as to record an omnidirectional signal W. To be able to locate, from the recorded signals, a speaker, for example, who is seated in front of the display in a perpendicular direction or is positioned in front of the plane, or to be able to suitably amplify the signal of the speaker, it is necessary to generate a signal which has a directivity in the z

- direction, i.e. whose maximum signal energy comes from the z direction. In order to generate such a signal, only the signals of the remaining four or five microphones can be used. The subtraction of two microphone signals in a manner analogous with generating the x or y-directional virtual microphone signals is out of the question, since all of the microphones used do not differ from one another with regard to their Z coordinates.
- 10 A signal with a directivity characteristic which has a maximum in the z direction may be generated by means of known "beam-forming" techniques while using a planar microphone array or a microphone matrix arranged within a plane. Beam-forming is also used with WLAN antennas, inter
- 15 alia. In this context use is made, for example, of so-called "filter-and-sum beam-formers" (see, for example: G.W. Elko: "Superdirectional microphone arrays" in S.G. Gay, J. Benesty (eds.): "Acoustic Signal Processing for Telecommunication", Chapter 10, Kluwer Academic Press,
- 20 2000, ISBN: 978-0792378143; and J. Bitzer, K.U. Simmer: "Superdirective microphone array" in M. Brandstein, D. Ward (eds.): "Microphone Arrays - Signal Processing Techniques and Applications", Chapter 2, Springer Berlin, 2001, ISBN: 978-540-41953-2) or differential microphone arrays. Mainly
- 25 signals of individual microphones or antennas are added up in a phase-shifted and weighted manner, so that a constructive interference of the individual antennas or microphones results for a spatial direction associated with the phase shift, i.e. a signal picked up from this spatial
- 30 direction is amplified as a result.

However, using these methods it is not possible to generate a signal having a directivity characteristic of a dipole in the z direction if a microphone array in the x and y planes

35 is available as signal-generating microphones or detectors. These known beam-forming methods are therefore not applicable to methods such as the DirAC algorithm, which

depend on that the input signals have dipole-type directivity characteristics.

In addition, simple beam-forming methods wherein signals of
5 different detectors or microphones which comprise spatial
direction-dependent phase shifts are added up, are not
suitable for generating directivity characteristics for
signals whose bandwidths are so large that the phase
10 difference for different frequencies of the signal to be
picked up or to be detected amounts to more than 360°
solely due to the different wavelengths and the geometric
distances of the employed detectors or microphones within
the bandwidth to be detected. If a method of constant
15 phase-shifted superposition were applied to such a signal,
gains and also extinctions would inevitably result within
the bandwidth to be picked up. In this case, one can no
longer speak of a uniform directivity characteristic for
the detector. With WLAN frequencies of, e.g., 5.15 GHz to
20 5.725 GHz, this problem does not arise since due to the
comparatively small bandwidths of the individual channels
(for WLAN 802.11a for example ≈ 0.03 GHz) with a constant
phase shift, constructive interference may be achieved for
all of the frequencies to be detected. However, if broad-
25 band RF signals or audio signals are picked up, application
of a constant phase shift with a summation of the signals
is no longer practicable. If audio signals of 20 Hz to 20
kHz are to be picked up, the wavelengths will change from
about 15 m to 1.5 cm, so that with common geometrical
30 dimensions (for example with a distance of several cm from
neighboring microphones) one can no longer speak of a
constant phase shift between the individual microphones for
all of the frequencies to be detected.

One may have similar thoughts about the two-dimensional
35 case as may occur, for example, in telecommunication
applications. In this field it may be sufficient to model
or capture the acoustic environment only within a two-
dimensional plane. This may be sufficient if only a planar

loudspeaker configuration is to be used for reproduction. If one assumes that the plane is spanned by the X and Y axes, it will be sufficient to record an omnidirectional signal W, the X and the Y signals, which may be achieved,
5 for example, by means of a microphone array of four microphones arranged at each of the corners of a square.

In practice, however, it is often not even possible to arrange microphones in two dimensions, for example when the
10 microphones are to be arranged on the top frame of a laptop display. In this case, only linear microphone arrays can be used, i.e. all of the microphones are arranged on a straight line. If, without loss of generality, this straight line is defined as the X direction within an XY
15 plane, only the omnidirectional signal W and the dipole signal X may be directly recorded as a consequence. A typical setup could consist of three microphones wherein use is made of two spaced-apart omnidirectional microphones in order to measure the X signal (dipole directivity
20 characteristic), and of an additional omnidirectional microphone arranged at the center of the two X microphones in order to measure the omnidirectional signal W. Even if it should be sufficient to perform the sound analysis merely within the XY plane, it is necessary in such a
25 scenario to determine or to calculate a signal having a directivity characteristic in the Y direction. In the event of a subsequent DirAC analysis, this signal must additionally have the required dipole directivity characteristic. Just like it was discussed in the above 3-
30 dimensional case, a signal having a directivity characteristic in the Y direction may be generated by means of conventional beam-forming techniques. However, such a directivity characteristic having a dipole shape cannot be achieved by means of the three microphones discussed above.

35

Even a recently suggested expansion of "filter-and-sum beam-forming" (H. Kamiyanagida, H. Saruwatari, K. Takeda, F. Itakura: "Direction of arrival estimation based on

nonlinear microphone array" in Proceedings of the IEEE Int. Conf. On Acoustics, Speech and Signal Processing (ICASSP), pp. 3033-3036, Salt Lake City, May 2001) does not result in a directivity characteristic which, for all of the frequency portions of the signal to be detected, has a maximum in a spatial direction. Rather, in the suggested expansion, the direction in which a plurality of audio sources are located with regard to the pickup position is estimated. For this purpose, the amplitude square of a signal generated by means of a beam-forming microphone array is subtracted from the amplitude square of another output signal of a beam-forming microphone array. By evaluating the signal thus generated, the directivity of the microphone array may be constantly adjusted, so that the direction of minimum sensitivity of the array corresponds to the direction of the active sound sources.

By this type of adjustment of the directivity (beam-forming), one does not actually achieve a predetermined directivity with a gain which does not disappear within a large spatial area, but the directivity is changed or tracked only such that the direction of minimum sensitivity of the microphone array corresponds to the direction of the active sound sources. Extending the beam-forming algorithm therefore enables estimating the direction of a plurality of active sound sources, whereas no information may be derived on the origin of the maximum signal energy of the audio signal, i.e. the intensity of the signal or the diffuseness of the signal.

It is therefore necessary to be able to generate signals having predetermined directivity characteristics as substitution signals.

US application 2006/0 115 103 A1 addresses interference suppression, or beam forming, of signals recorded with two microphones. Their amplitude values directly added and

possibly additionally scaled or provided with an additional phase shift.

US application 2007/0014419 addresses a similar method for
5 hearing aids. Here, too, the output signal is formed by linear superposition of the scaled input signals.

US application 2006/0171547 addresses the DirAC
representation of spatial signals and thus describes a
10 possibility of transmitting the directions of origin of signal components in a reproducible manner without requiring a large transmission bandwidth.

This object is achieved by a signal processor as claimed in
15 claim 1, a method as claimed in claim 19, and a computer program as claimed in claim 20.

In some embodiments of the invention, a signal having a predetermined spatial directivity characteristic is
20 generated from a first signal having a known spatial directivity characteristic and from a second signal having a known spatial directivity characteristic in that the first and second signals are initially converted from a temporal to a spectral representation. The spectral
25 representations of the first and second signals are combined, in accordance with a combination rule which depends on the known directivity characteristics of the first and second signals, such that a spectral representation of a signal having the predetermined spatial
30 directivity characteristic is obtained which differs both from the directivity characteristic of the first signal and from that of the second signal. In some embodiments, the directivity characteristic is identical for all of the spectral ranges of the signal generated.

35

In some embodiments, in particular the amplitude magnitudes of the spectral representations of the input signals are

formed before these are combined so as to generate amplitude values for the substitution signal generated.

5 The spectral representation of the signal obtained by the combination may thus have a directivity characteristic which differs from the directivity characteristics of the first and second signals picked up, in particular a directivity characteristic whose maximum gain factor points to a spatial direction which differs from the spatial
10 direction of the maximum sensitivity of the signals having known directivity characteristics. By combining the signals in their spectral representations one may achieve that the directivity characteristic for the entire resulting signal is identical within an identical spatial area. In
15 particular, this means that for different frequency components of the signal to be detected or to be generated, the maximum of the sensitivity, or the maximum gain factor, of the directivity characteristic is positioned in the same spatial direction. This is true even when the bandwidth of
20 the signals picked up, or of the signal to be generated, is so large that a superposition of the signals in the temporal representation with a constant phase shift would result in that the maximum directivity, or the maximum of the directivity characteristic, would change significantly,
25 in terms of space, within the spectral range, or the bandwidth, of the signal to be generated. By combining the spectral combinations, however, one may achieve that a directivity characteristic is generated which is similar or identical for all of the signal portions of a broad-band
30 signal within the same spatial area, or in the same spatial direction.

In some embodiments of the invention, the spectral representation of the signal generated is output, or made
35 available for further processing, directly by a signal processor. In further embodiments of the invention, the spectral representation of the signal generated is converted back to the temporal representation so as to

obtain a temporal representation of the signal having the predetermined directivity characteristic.

In some embodiments of the invention, audio signals, or an
5 audio pickup of pieces of music or of ambient noises or
speakers, which are obtained using a 2-dimensional or 1-
dimensional microphone array, are processed. Consequently,
information on the localization or the position of the
10 audio sources in a direction which is orthogonal with
regard to the arrangement of the microphones is also
obtained. In further embodiments, radio-frequency signals
of large bandwidths, or any other signals, are processed
such that a signal is generated whose maximum contribution
to the signal amplitude comes from a predetermined spatial
15 direction, i.e., in other words, which has a predetermined
spatial directivity characteristic. For these purposes, a
signal having a directivity characteristic is to mean a
signal having directionally weighted signal portions, so
that there is/are one or several spatial directions from
20 which signal portions having maximum gains or maximum
amplitudes are recorded or reconstructed, whereas there are
other spatial directions in which the signal portions are
attenuated or completely suppressed.

25 In some embodiments of the invention, a short-time
frequency transformation, wherein the signals to be
transformed or converted are processed block by block, is
employed for the conversion to the spectral representation.
To this end, for example, a filter bank or a frequency
30 transformation may be used wherein each signal component
which has a predetermined length and may consist of, e.g.,
a sequence of a predefined number of signal samples, has a
plurality of amplitude and phase values associated with it,
as is the case, for example, with short-time Fourier
35 transformation (SFT). A continuous signal in a time
representation is transformed to a sequence of amplitude
and phase factors, or is represented as a sequence of these
factors, each signal component, i.e. each independently

processed time interval, having a plurality of amplitude values $P(k, n)$ associated with it (the index k indicates the analyzed frequency band). In the combination of the spectral representations of the transformed signals, only
5 the magnitudes of the amplitudes are combined to obtain the signal having a predetermined directivity characteristic. In this manner it is ensured that the resulting directivity characteristic may be identical for all of the frequencies of the frequency band to be recorded. This may be the case
10 even when the location where the signals having known directivity characteristics were recorded are not known exactly. The maximum of the sensitivity may thus be obtained for all of the frequencies in the same spatial direction in a simple manner requiring little mathematical
15 effort. In some embodiments of the present invention, the signal thus obtained which has a predetermined spatial directivity characteristic may immediately continue to be used in its spectral representation in order to estimate the spatial direction within the observed frequency band
20 from which the maximum sound energy or signal energy is coming.

In some embodiments, the spectral representation of the generated signal having the predetermined directivity
25 characteristic is used for directly obtaining, by means of a DirAC analyzer or a DirAC algorithm, the DirAC parametrization in two dimensions or three dimensions. The signal generated substitutes for, or replaces, a signal which is not accessible to direct measurement. With this in
30 mind, the signal generated by means of the signal processor may also be referred to as a substitution signal. Therefore, the terms "substitution signal" and signal having a predetermined directivity characteristic shall be used synonymously below. The use of embodiments of signal
35 processors with DirAC analyzers thus results in the great advantage that a complete DirAC analysis and, thus, a parametrization of the spatial sound impression becomes possible without having to perform costly recording of

three independent signals each having directivity characteristics which are, in pairs, orthogonal to one another.

- 5 When the information on the intensity with which the signal from the three spatial directions is recorded is available, a statement may furthermore be made as to the level of diffuseness of the signal picked up. The intensity vector indicates the energy flux density. In a three-dimensional
10 space, the intensity vector has three orthogonal components (e.g. x, y, z), which together reveal the direction of the energy flux.

If the components of the intensity vector are the same or
15 almost zero in all three spatial directions, one can assume that the signal, or the sound, evenly fills the measuring space, since small or near-zero components of the intensity vector are present from all of the spatial directions within the frequency interval examined.

20 In further embodiments of the invention, the spectral representation of the generated signal having a predetermined directivity characteristic is converted to a temporal representation, so that a signal is obtained,
25 approximately, as would have been recorded by a virtual microphone having a predetermined directivity characteristic. The phase factors of the conversion to the frequency range of any of the (input) signals having predetermined directivity characteristics may be used so as
30 to obtain as realistic a phase relation as possible between the individual frequency ranges. This may result in that although only the amplitudes were taken into account in the implementation of the directional dependence, a signal is generated whose audible artefacts are hardly perceivable
35 due to the phase information some which has not been taken into account.

Preferred embodiments of the present invention will be explained below in detail with reference to the appended figures, wherein:

- 5 Fig. 1 shows an embodiment of a signal processor;
- Fig. 2 shows an embodiment of a signal combiner of the
 signal processor;
- 10 Fig. 3 shows an example of an arrangement of microphones
 for recording signals for signal processors;
- Fig. 4 shows an example of directivity characteristics
 of recorded signals;
- 15 Fig. 5 shows an example of a predetermined directivity
 characteristic of a generated signal;
- Fig. 6A and Fig. 6B show embodiments of an apparatus for
20 generating a signal having a predetermined
 directivity characteristic;
- Fig. 7 shows an embodiment of deriving a DirAC
 parametrization; and
- 25 Fig. 8 shows an embodiment of a method of generating a
 signal having a predetermined directivity
 characteristic.
- 30 Fig. 1 shows an embodiment of a signal processor 2
 comprising a signal converter 4 and a signal combiner 6.
 The signal processor 2 shown in Fig. 1 serves to generate a
 signal having a predetermined spatial directivity
 characteristic (a substitution signal) while using a first
35 signal 8a having a known spatial directivity
 characteristic, and a second signal 8b having a known
 spatial directivity characteristic. The signals 8a (P_1) and
 8b (P_2) may be picked by a microphone or be received by an

antenna, for example, and are present in temporal representations. As is indicated in Fig. 1, further embodiments of the invention may use more than two signals as input signals, in particular the number of the signals used as input signals in principle having no upper limit. 5 The signal converter 4 serves to convert the first signal 8a and the second signal 8b to spectral representations of the signals $P_i(k,n)$, respectively, where $i = 1, 2$.

10 The signal combiner 6 combines the spectral representations of the first signal 8a and of the second signal 8b in accordance with a combination rule so as to generate the signal 10 having a predetermined spatial directivity characteristic. The generated signal 10 may have a 15 directivity characteristic which differs from the directivity characteristics of the signals 8a and 8b. In some embodiments of the invention, the combination rule used by the signal combiner 6 for generating the signal 10 having a predetermined directivity characteristic 20 exclusively depends on the known directivity characteristics of the first and second signals 8a and 8b.

In the embodiments using more than two signals having known directivity characteristics as input signals, it is also 25 required that the directivity characteristic be known for each input signal, so as to generate a signal whose directivity characteristic corresponds to the predetermined, or desired, directivity characteristic. The signal processor 2 shown in Fig. 1 thus generates the 30 signal 10, which has a predetermined spatial directivity characteristic, in a spectral representation of same.

The general case of generating the signal having a predetermined directivity characteristic from any number L 35 > 2 of signals having known directivity characteristics will be briefly described below before the concept underlying the invention will be presented once again in more detail with reference to two specific embodiments.

The inventive embodiments will be discussed on the basis of audio signals, even though the concept is also applicable to any other signals, such as radio-frequency signals or radio signals, for example. The starting point for processing the signals by embodiments of inventive signal processors are $L > 2$ signals having known directivity characteristics. These signals may either be (omnidirectional or directional) microphone signals measured directly using the known directivity characteristic, or they may be signals which are tapped by a directivity-pattern output of a microphone array. The manner in which the directivity characteristic was generated is inessential as long as said directivity characteristic is known.

Each of the input signals is subdivided into a sequence of discrete time intervals, or signal components (signal blocks). The signal blocks are converted to a spectral representation, for example to the short-time frequency domain. In the following, $P_i(k,n)$ shall designate the i^{th} input signal within the frequency band. The index n designates the quantity, or the number, of the signal block being contemplated of the series of signal blocks into which the input signal was decomposed. After the frequency conversion, the signal may be represented, within each frequency band of interest, as an amplitude, or magnitude, $|P_i(k,n)|$ and as a phase $\Phi(k,n)$, so that:

$$P_i(k,n) = |P_i(k,n)| e^{j\Phi_i(k,n)},$$

where j designates the imaginary unit. The signal combiner of the signal processor combines the magnitudes $|P_i(k,n)|$ of these input signals while using a combination rule such that a magnitude $|D(k,n)|$ resulting from the combination is associated with a spectral representation of the signal to be generated. This corresponds to a microphone signal having the predetermined spatial directivity

characteristic. It is possible, in particular, to keep the directivity characteristic constant or similar within a broad spatial area and a large frequency range. Generally, the combination of the magnitudes may be described by the following formula:

$$D(k,n) = g(|P_1(k,n)|, |P_2(k,n)|, \dots, |P_L(k,n)|),$$

wherein $D(k,n)$ describes the signal generated (the substitution signal). The function $g(.)$ describes the combination rule in accordance with which the magnitudes of the input signals are combined and which may fundamentally be formed, or composed, of any linear and non-linear functions.

15

In some embodiments of the invention, one of the phases of the input signals is used as the phase information of the generated signal $D(k,n)$. For example, if the first signal is selected by the signal combiner is the signal whose phase is used, the combination rule may be defined as follows:

20

$$D(k,n) = g(|P_1(k,n)|, |P_2(k,n)|, \dots, |P_L(k,n)|) e^{j\phi_1(k,n)}.$$

A signal combiner 6, which converts the above-described concept, is schematically depicted in Fig. 2. In this context, the input signals 8a, 8b and 8c are already present in their spectral representations. As was described above, the phase information may be extracted from the first input signal 8a, it being possible, in further embodiments, to use the phase information of any other input signals. Further embodiments may fully dispense with the phase information. The phase information 12 is extracted from the first input signal 8a, the magnitude 14b of the first input signal 8a being formed by a magnitude former 14a. Equivalently, the magnitude values 16b and 18b of the input signals 8b and 8c are formed by the magnitude formers 16a and 18a, respectively. In a combination block

25

30

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20, the magnitudes 14b, 16b and 18b of the input signals 8a, 8b and 8c are combined using a combination rule (function g) so as to generate the magnitude value 22 of the signal generated. An optional multiplier 24 serves to
5 form the signal $D(k,n)$ by multiplying the magnitude value 22 and the phase factor 12, as was described in the previous paragraphs.

By employing a suitable combination rule g which is
10 dependent on the directivity characteristics of the input signals, a signal having a predetermined spatial directivity characteristic may be generated which, in particular, has a directivity characteristic different from those of the input signals. The signal combiner 6 outlined
15 in Fig. 2 thus generates a signal having a predetermined spatial directivity characteristic in a spectral representation. This signal may directly be used further so as to derive, in connection with the input signals 8a to 8c, parameters which describe the fundamental properties of
20 the signal received, or of the audio signal, in the pickup environment. These parameters may be the DirAC parameters, for example, i.e. the direction of the instantaneous intensity per frequency range, and the diffuseness of the signal in each of the frequency ranges contemplated. In
25 further embodiments, the generated signal which has a predetermined directivity characteristic and is present in the spectral representation may also be transformed back to a temporal representation.

30 In this context it is possible, in particular, to generate a predetermined directivity characteristic within large spatial and frequency ranges without having to make assumptions about the statistics of the input signals used. Such assumptions concerning the signal statistics, for
35 example concerning the stationary behavior, the different spectral properties or the spatial coherence between several signals, are typically made in non-linear filter techniques as are used for reducing the background noise in

speech enhancement systems (also known as spectral subtraction or Wiener filtering).

Embodiments of signal combiners, however, only require the
5 knowledge of the directivity characteristics of the
microphones, or microphone arrays, used for recording the
input signals, and do not a priori make assumptions about
the statistics of the input signals, or about the
statistics of their spectral compositions. As a result, it
10 becomes possible, in an efficient manner which is based on
simple algorithms, to generate a signal having a
predetermined spatial directivity characteristic using
input signals having known directivity characteristics.

15 Fig. 3 schematically shows a potential microphone setup by
means of which the signals W having omnidirectional
directivity characteristics, X having dipole-shaped
directivity characteristics in the X direction, and Y
having dipole-shaped directivity characteristics in the Y
20 direction may be received. Fig. 3 shows a scenario in which
five microphones 30a - 30e are arranged within a plane. In
this context it is assumed that it is impossible, due to
geometric boundary conditions, to arrange further
microphones outside the plane shown in the top view in Fig.
25 3. By means of such an arrangement, signals having
directivity characteristics in the X direction and in the Y
direction may be generated by combinations of the signals
picked up by the microphones 30a - 30d. The central
microphone 30e having omnidirectional directivity may be
30 used, for example, for recording an omnidirectional signal
W and making it available as an input signal having an
omnidirectional directivity characteristic.

Fig. 3 is only one of any number of potential examples
35 which enable recording signals having directivity
characteristics in the X direction, in the Y direction, and
having no specific directivity characteristics, i.e.
recording omnidirectional signals W.

In the description of the following embodiment, wherein a signal is generated which has a predetermined spatial directivity characteristic, and on the basis of which, e.g., a DirAC parametrization of the spatial audio signal may be performed, one assumes that an omnidirectional signal W, a signal having a dipole-type directivity characteristic X in the X direction, and a signal Y having a directivity characteristic in the Y direction, as may be obtained, for example, by means of the 2-dimensional microphone array shown in Fig. 3, are available as the input signal.

As has already been mentioned, it is assumed in this context that the direction of maximum sensitivity of the signal $X(k,n)$ is the X direction, and of the signal $Y(k,n)$ it is the Y direction, of a Cartesian coordinate system. In addition, the signals W, X and Y are to be present in a spectral representation already, i.e. for each frequency range and time block, or signal component, of the signals an amplitude parameter and a phase parameter exist, as was described in the previous paragraphs. To be able to describe the sound field with sufficient accuracy in a 3-dimensional manner, it is required to generate a signal having a directivity characteristic whose maximum comprises a component in the Z direction. If a DirAC parametrization is envisaged, it is necessary, in particular, to generate a signal which has a dipole-type directivity characteristic with a maximum, or with a maximum sensitivity, in the Z direction.

Fig. 4 illustrates the directivity characteristic of a signal formed from the amplitudes $X(k,n)$ and $Y(k,n)$ of the input signals X and Y in accordance with the following combination rule:

$$\sqrt{|X(k,n)|^2 + |Y(k,n)|^2}.$$

Fig 4 shows a 3-dimensional representation of the directivity characteristic of the signal formed in accordance with the above combination or combination rule. What is represented is the gain factor (the directional weighting factor) with which the signal from the respective spatial direction is contained within the combination signal, as against the position of the source in the X direction 40 and in the Y direction 42. A gain factor of 1 signifies that the signal is recorded in an unattenuated manner, i.e. with an amplitude, or intensity, which is not reduced by the combination of the two individual signals X and Y. As simple geometric considerations reveal, the gain factor is constantly zero along the Z axis, since neither the X signal nor the Y signal, or the microphones associated with this signal, are/is sensitive in this direction. Thus, the above-mentioned combination of the X signals and Y signals results in a directivity which corresponds to the torus shown in Fig. 4, whose axis of rotation is the Z axis. If one further takes into account that the directivity characteristic of an omnidirectional signal W has no maximum, i.e. has a spherical shape, it becomes evident that a signal having a dipole-shaped directivity characteristic in the Z direction may be obtained if the signals $W(k,n)$, $X(k,n)$ and $Y(k,n)$ are combined in accordance with the following combination rule:

$$|\hat{Z}(k,n)| = \sqrt{|W(k,n)|^2 - (|X(k,n)|^2 + |Y(k,n)|^2)}.$$

Thus, a signal having a predetermined spatial directivity characteristic (a dipole aligned in the Z direction) is generated by the above relationship, provided that the directivity characteristics of the input signals X, Y and W are known.

Thus, the magnitude of the generated signal within the frequency range concerned directly results from the combination rule.

In an embodiment of the invention, the phase $\phi_w(k,n)$ may be selected in correspondence with the phase of the omnidirectional signal $W(k,n)$, so that the generated signal which is extended by a piece of phase information exhibits
5 the following form:

$$\hat{Z}(k,n) = |\hat{Z}(k,n)| e^{j\phi_w(n,k)}.$$

In a 1-dimensional representation, Fig. 5 illustrates the
10 gain factor in dependence of the angle α (52) between the Z axis and the direction of the signal source. The continuous line 54 describes the directivity characteristic of a conventional dipole. The dotted curve 56 illustrates the directivity characteristic of the signal $Z(k,n)$, which
15 by means of the above combination rule was combined, or derived, from the signals W, X and Y.

As may be seen from Fig. 5, the phase information is no longer complete due to the combination, which makes itself
20 felt in that the directivity characteristic of the so-called homophasic dipole 56 is equivalent to that of the "classic" dipole only for the angles of 0 to 90° and 270 to 360°. The absolute magnitude of the directional gain factors in the range between 90° and 270° is identical to
25 the absolute magnitude of the classic dipole, but no negative signs occur. If, as for example in the DirAC parametrization, one requires only that information which states the spatial direction from which the maximum intensity is recorded, this loss of information is
30 acceptable. Therefore, when the signal Z having the directivity characteristic just described is generated by means of the above combination rule, a subsequent DirAC analysis based on the signal generated may result in a correct determination of the intensity vector within the
35 half-space of the Cartesian coordinate system with a positive Z axis.

Figs. 6A and 6B once again schematically illustrate, respectively, embodiments of an apparatus for generating a signal having a predetermined directivity characteristic, and a system for generating a DirAC parametrization on the basis of a first signal X having a directivity characteristic in the X direction, a second signal Y having a directivity characteristic in the Y direction, and an omnidirectional signal W. Fig. 6A schematically shows an embodiment of a signal processor 2 which has a first signal X, a second signal Y and a third signal W supplied to it. The directivity characteristics of the signals correspond to the embodiment described further up, so that, by means of the signal processor 2, a signal Z having a predetermined spatial directivity characteristic may be generated wherein the maximum directional weighting factor occurs in the Z direction. As was described by means of the previous embodiments, for this purpose the signals X, Y and W, which are initially picked up by microphones in a temporal representation, are converted to a spectral representation, whereupon a signal Z having the predetermined directivity characteristic is formed by the above-described combination of the magnitude values.

As is illustrated in Fig. 6B, the spectral representation of the signals X, Y and W may alternatively be supplied, along with the spectral representation of the specific signal Z, to a DirAC analyzer 60, which, as was described above, generates the parameters characteristic for the DirAC parametrization of an acoustic spatial signal, namely a direction vector 62 and a diffuseness parameter 64. Along with the omnidirectional signal W in its temporal representation, or in its spectral representation, these parameters enable preferring random spatial directions in a subsequent reproduction, or reproducing signals only from the random spatial directions, or faithfully reproducing the sound field as well.

What follows below is an illustration, by means of Fig. 7, of how a DirAC parametrization may be performed using a substitution signal having a predetermined directivity characteristic and being generated by an embodiment of a signal processor, for example how the DirAC parameters (a direction vector or direction information and a diffuseness value ψ may be formed. Fig. 7 shows, in a Cartesian coordinate system comprising the axes X, Y and Z, seven omnidirectional microphones at the corners of an octahedron and at the center thereof, by means of which omnidirectional microphones the complete information required for the DirAC parametrization or analysis, i.e. signals with directivities in X, Y and Z as well as an omnidirectional signal W, may be obtained. In the arrangement which is shown in Fig. 7 and is to be seen as an example only, the signals having directivity characteristics may be obtained from the difference between two omnidirectional microphones spaced apart from each other in the respective spatial direction. In the example shown in Fig. 7, a signal having the directivity characteristic in the X direction (signal $X(k,n)$) may be obtained by subtracting the signals of the microphones 70a and 70b, said signals possibly having been suitably normalized in a frequency-dependent manner. In addition, a signal having a directivity characteristic in the Y direction (signal $Y(k,n)$) may be obtained by subtracting the signals of the microphones 72a and 72b, said signals possibly having been suitably normalized in a frequency-dependent manner. In addition, a signal having a directivity characteristic in the Z direction (signal $Z(k,n)$) may be obtained by subtracting the signals of the microphones 74a and 74b, said signals possibly having been suitably normalized in a frequency-dependent manner. The omnidirectional microphone 76 (W) arranged at the center serves to pick up a signal $W(k,n)$ having an omnidirectional characteristic, as is required for a DirAC analysis. The signal having an omnidirectional characteristic may alternatively also be calculated as a mean value of all of

the microphone signals or as a mean value of the external microphone signals.

For simplicity's sake, it shall be assumed below that a
 5 transformation to the spectral range has already occurred
 for the signals in question, so that indices k indicate the
 contemplated spectral range in each case, whereas the index
 n describes the contemplated signal component, or signal
 block, of the block-wise frequency analysis (e.g. short-
 10 time Fourier transformation SFT) used by way of example
 here.

For the DirAC analysis, an intensity vector is initially
 calculated whose X component results from the following
 15 relation:

$$I_x(k, n) = \operatorname{Re}\{E\{W^*(k, n)X(k, n)\}\} ,$$

where the operator $*$ designates generating the conjugate
 20 complex.

It goes without saying that the equivalent relations apply
 to the Y and Z components of the intensity vector even
 though the above formula only describes the X component.
 25 If, as in the case shown in Fig. 7, all of the information,
 i.e. the signals X, Y, Z and W, was available, the
 coordinates of the direction of maximum loudness 78,
 depicted by way of example in Fig. 7, could be determined
 as follows:

$$\varphi(k, n) = \arctan \frac{I_y(k, n)}{I_x(k, n)} ,$$

$$\vartheta(k, n) = \arctan \frac{I_z(k, n)}{\sqrt{I_x^2(k, n) + I_y^2(k, n)}} .$$

In the above equations, the angle ϕ designates the azimuth 79a, and the angle θ designates the elevation 79b, i.e. those angles which unambiguously describe the direction of the sound source 78 in relation to the origin of the coordinate system. The diffuseness ψ is determined as follows in the DirAC analysis:

$$E(k,n) = \frac{1}{2} \left(E\{X(k,n)|^2\} + E\{Y(k,n)|^2\} + E\{Z(k,n)|^2\} + E\{W(k,n)|^2\} \right)$$

10

$$\psi(k,n) = 1 - \frac{\sqrt{I_x^2(k,n) + I_y^2(k,n) + I_z^2(k,n)}}{E(k,n)}$$

15 If, as was discussed in detail in the preceding embodiment, a microphone signal having a directivity characteristic in the Z direction is not available, the signal $\tilde{Z}(k,n)$ may be generated by means of an embodiment of a signal processor, as was described above. This signal will have the
20 directivity characteristic illustrated with reference to Fig. 5. Therefore, the signal thus generated may be used, in the frequency range, directly for deriving the DirAC parameters. In accordance with the above considerations, the intensity vector for the DirAC parametrization in the Z
25 direction consequently results as:

$$\tilde{I}_z(k,n) = \text{Re}\{E\{W^*(k,n)\tilde{Z}(k,n)\}\}$$

An elevation angle 79b indicating the position of the sound
30 source 78 in the Z direction may consequently be derived even in the case where there is no signal which has been picked up generically and has a directivity characteristic in the Z direction. This is possible in that the intensity $\tilde{I}_z(k,n)$ is used for calculating the angle in accordance with
35 the following formula:

$$\vartheta(k,n) = \arctan \frac{\tilde{I}_z(k,n)}{\sqrt{I_x^2(k,n) + I_y^2(k,n)}} .$$

As was already mentioned above, the directivity
 5 characteristic of the signal generated may correspond to
 that of a homophasic dipole due to the fact that the phase
 information is not taken into account in the combination
 rule. Therefore, unambiguous association of the position of
 the sound source 78 is possible only within the half-plane
 10 with positive Z values. In the case of a 2-dimensional
 microphone array generating X, Y and W signals, by analogy
 with the case of complete DirAC microphone signals, the
 diffuseness parameter results in that the value I_z is
 replaced by the intensity $\tilde{I}_z$ derived above.

15 A DirAC analysis is also possible if one is only interested
 in the 2-dimensional analysis of the sound field. In this
 case, the omnidirectional microphone 76, for example, may
 be dispensed with, since, as will be shown below, a
 20 substitution signal having a torus-shaped directivity
 characteristic may be generated from the X and Y signals,
 which substitution signal enables determining the 2-
 dimensional DirAC parameters of interest, namely azimuth
 79a and the diffuseness parameter ψ_{2D} .

25 Initially, a signal $W_T(k,n)$ is formed as a substitution
 signal having a predetermined directivity characteristic,
 the signal $W_T(k,n)$ having a torus-shaped directivity
 characteristic, as was already described above. The signal
 30 $W_T(k,n)$ may be represented from the signals X and Y as
 follows:

$$W_T(k,n) = \sqrt{|X(k,n)|^2 + |Y(k,n)|^2} e^{j\phi_w(k,n)} .$$

The phase parameter is optional. The "2-dimensional intensity" may be formed as follows:

$$I_{2D,x}(k,n) = \text{Re}\{E\{W_T^*(k,n)X(k,n)\}\},$$

5

where, for the "2-dimensional intensity" in the Y direction, the above formula for the Y signal applies equivalently.

- 10 The azimuth angle 79a may be directly determined, in accordance with the following formula, from the X and Y intensities:

$$\varphi_{2,D}(k,n) = \arctan \frac{I_{2D,y}(k,n)}{I_{2D,x}(k,n)}.$$

15

- The "2-dimensional" overall energy value may be calculated, in accordance with the following formula, by means of the generated signal having a predetermined directivity characteristic (signal W_T having a torus-shaped directivity characteristic):
- 20

$$E_{2D}(k,n) = \frac{1}{2} \left(E\{X(k,n)^2\} + E\{Y(k,n)^2\} + E\{W_T(k,n)^2\} \right).$$

- The diffuseness parameter ψ may thus be calculated, in accordance with the following formula, also in the 2-dimensional case by using the substitution signal W_T , which was generated by means of an embodiment of a signal processor:
- 25

$$\psi_{2D}(k,n) = 1 - \frac{\sqrt{I_{2D,x}^2(k,n) + I_{2D,y}^2(k,n)}}{E_{2D}(k,n)}.$$

As the above embodiments have revealed, it is possible, by using an embodiment of an inventive signal processor, to save microphones, or to compensate for geometric boundary conditions which do not enable ideal placement of microphones, and to nevertheless perform a complete DirAC analysis of a spatial signal.

In the further embodiment of the present invention which will be discussed below, it shall be assumed that only a linear microphone array is available as signal suppliers, i.e. as signal sources for the input signals of the signal processor. This directly results in that a signal having a directivity characteristic only in that direction which is defined by the connecting line between the microphones is available. Without loss of generality, this direction shall be referred to below as the X direction. Consequently, only an omnidirectional signal $W(k,n)$ and a signal $X(k,n)$ may be made available as an input signal for the signal processor. Nevertheless, an acoustic analysis may occur at least within the X-Y plane, for example with the DirAC parametrization, since, as will be described below, a signal $Y(k,n)$ having a predetermined, dipole-type directivity characteristic may be generated from the signals W and X . By analogy with the above considerations, the following combination rule may be modified in order to determine, in accordance with the following relation, the magnitude values of the signal to be generated:

$$|\hat{Y}(k,n)| = \sqrt{|W(k,n)|^2 - |X(k,n)|^2}.$$

In accordance with the above considerations, the directivity characteristic of the signal thus generated corresponds to a torus whose axis of rotation is the X axis. In particular with a 2-dimensional view within the X-Y plane, this means that the directivity characteristic of the signal $\hat{Y}$ within the X-Y plane corresponds to a dipole whose maximum weighting factor lies on the Y axis, and whose directivity characteristic is symmetric to the Y

axis. The directivity characteristic thus generated therefore meets the preconditions for a subsequent DirAC analysis.

- 5 An optional phase factor may correspond to the phase factor of the omnidirectional signal $W(k,n)$, so that, if phase information is desired, the combination rule may be extended in order to form the generated signal such that it may be described in accordance with the following formula:

10

$$\hat{Y}(k,n) = \left| \hat{Y}(k,n) \right| e^{j\phi_w(n,k)}.$$

The signal $\hat{Y}(k,n)$ thus generated has a directivity characteristic which corresponds to a homophasic dipole within the X-Y plane. Thus, a subsequent DirAC analysis will lead to a correct determination of the intensity vector within the half-plane of the X-Y plane, which includes the positive Y axis. Consequently, even in the case of a 1-dimensional microphone array, a DirAC analysis may be performed which enables highly flexible further processing of the signal analyzed if embodiments of inventive signal processors can be used for generating an additional signal having a predetermined spatial directivity characteristic.

25

The following considerations reveal the manner in which a subsequent DirAC analysis (2-dimensional case) may be performed, even in the case of a linear microphone array, by means of the signal $\hat{Y}(k,n)$ generated above. The intensity vector for the DirAC analysis is calculated for the X component as it is done in the normal case, since this component is directly available along with an omnidirectional signal.

- 35 For determining the Y component, the generated substitution signal having the predetermined directivity characteristic $\hat{Y}(k,n)$ is used, so that:

$$\tilde{I}_{2D,y}(k,n) = \text{Re}\{E\{\tilde{W}^*(k,n)\tilde{Y}(k,n)\}\}$$

For determining the azimuth angle, the intensity $\tilde{I}_{2D,y}$ thus generated and the measured intensity $I_{2D,x}$ may be used. Even
 5 in the case of a linear microphone array, a diffuseness parameter ψ_{2D} may be derived from the following relations, even if only such microphones can be used which are arranged linearly one behind the other in a spatial
 10 direction:

$$\tilde{E}_{2D}(k,n) = \frac{1}{2} \left(E\{|X(k,n)|^2\} + E\{|\tilde{Y}(k,n)|^2\} + E\{|W(k,n)|^2\} \right)$$

$$\tilde{\psi}_{2D}(k,n) = 1 - \frac{\sqrt{I_{2D,x}^2(k,n) + \tilde{I}_{2D,y}^2(k,n)}}{\tilde{E}_{2D}(k,n)}$$

15 As was described above, information on the spatial composition of the signal may therefore also be given even if this may be recorded only with a 1-dimensional receiver array.

20 Some embodiments of the present invention therefore perform the following steps:

- Providing a plurality of signals having known directivity characteristics (either measured directly
 25 with this directivity characteristic or calculated by means of beam-forming using a microphone array).
- Transforming each input signal to the short-time frequency domain.
- For each frequency range to be taken into account:
 30 determining the magnitude and a phase information of each input signal.
- For each frequency range of interest: determining the magnitude of the signal to be generated by combining the

magnitudes of the input signals in accordance with a combination rule.

- If phase information is desired, combining the magnitude of the signal generated with one of the phase factors of the spectral representations of one of the input signals.

As was already mentioned above, the application of inventive signal processors enables the more flexible use of microphones, or microphone arrays, with regard to determining parameters for the spatial reconstruction of an ambience of sound which is to be picked up, or of a signal picked up.

- It becomes possible to generate signals having predetermined spatial directivity characteristics (such as a dipole characteristic, for example).

In addition, it is possible to improve, when using inventive signal processors, the spatial selectivity of microphone arrays in that, for example, the spatial area wherein the directional weighting factor is large may be restricted. This may be achieved, for example, in that the output signals of the microphone arrays having known directivity characteristics are processed by means of embodiments of inventive signal processors.

Fig. 8 shows an embodiment of a method of generating a signal having a predetermined spatial directivity characteristic. In a signal generation step 100, a first signal having a known spatial directivity characteristic 8a, and a second signal having a known spatial directivity characteristic 8b are provided. In addition, the temporal representations of the first and second signals are converted, in a transformation step, to a spectral representation of the first signal 8a and to a spectral representation of the second signal 8b. In a combination step, the spectral representation of the first signal 8a

and the spectral representation of the second signal 8b are combined, in accordance with a combination rule, such that a spectral representation, resulting from the combination, of the signal to be generated has the predetermined spatial
5 directivity characteristic, the predetermined directivity characteristic differing from the directivity characteristics of the first and second signals.

In an optional re-transformation step 104, the spectral
10 representation of the generated signal is converted to a temporal representation in order to obtain a signal having a predetermined spatial directivity characteristic which may be reproduced, for example, by means of a loudspeaker.

15 In an alternative analysis step 106, a parameterization, or spectral representation, of the spatial properties of the audio signal picked up is derived from the generated signal having a predetermined spatial directivity characteristic
10 and from the input signals 8a and 8b.

20 Depending on the circumstances, the inventive method of generating a signal having a predetermined spatial directivity characteristic may be implemented in hardware or in software. The implementation may be effected on a
25 digital storage medium, in particular a disk or a CD having electronically readable control signals, which may cooperate with a programmable computer system such that the inventive method of generating a signal having a
30 predetermined spatial directivity characteristic is performed. Generally, the invention thus also consists in a computer program product having a program code, stored on a machine-readable carrier, for performing the inventive
method, when the computer program product runs on a computer. In other words, the invention may therefore also
35 be realized as a computer program having a program code for performing the method, when the computer program runs on a computer.

Claims

1. A signal processor (2) for generating a substitution
signal (10) having a predetermined directivity
characteristic while using a first signal (8a) having
a known directivity characteristic, and a second
signal (8b) having a known directivity characteristic,
comprising:
 - a signal converter (4) for converting the first signal
(8a) and the second signal (8b) to a spectral
representation comprising several frequency bands, for
which at least one amplitude parameter is determined,
respectively; and
 - a signal combiner (6) for combining the spectral
representation of the first signal (8a) and of the
second signal (8b) in accordance with a combination
rule in order to obtain a spectral representation of
the substitution signal (10) having the predetermined
directivity characteristic which differs from the
directivity characteristics of the first (8a) and
second (8b) signals, an amplitude parameter of a
spectral representation of the substitution signal
(10) being formed from a combination of the absolute
magnitudes of the amplitude parameters of the first
signal and of the second signal.
2. The signal processor (2) as claimed in claim 1,
further comprising a second signal converter so as to
convert the spectral representation of the
substitution signal (10) to a temporal representation.
3. The signal processor as claimed in one of claims 1 or
2, wherein the signal combiner (6) is configured to
use the same combination rule for different frequency
bands of the spectral representations of the first
(8a) and second (b) signals.

4. The signal processor (2) as claimed in any of the previous claims, wherein the signal combiner (6) is configured to use a combination rule which depends only on the known directivity characteristics of the first (8a) and second (8b) signals.
5. The signal processor (2) as claimed in any of the previous claims, wherein the signal converter (4) comprises a block processor for subdividing the first (8a) and second (8b) signals into discrete signal components of constant lengths, the signal converter (4) being configured to convert each of the signal components to a spectral representation comprising several frequency bands.
6. The signal processor (2) as claimed in any of the previous claims, wherein the signal combiner is configured (4) to use such a combination rule that in a first signal (8a) having an omnidirectional directivity characteristic, and in a second signal (8b) having a dipole-shaped directivity characteristic in a dipole direction, a substitution signal is generated which has a dipole-shaped directivity characteristic in a predetermined direction perpendicular to the dipole direction.
7. The signal processor (2) as claimed in any of the previous claims, wherein the signal combiner (6) is configured to use a combination rule in accordance with which the magnitude of an amplitude parameter $D(k,n)$ of the substitution signal may be described by the following combination of the amplitude parameters $P_1(k,n)$ of the first signal (8a) and $P_2(k,n)$ of the second signal (8b):

$$|D(k,n)| = \sqrt{|P_1(k,n)|^2 - |P_2(k,n)|^2} .$$

8. The signal processor (2) as claimed in any of the previous claims, wherein the signal converter (4) is configured to additionally determine, for each frequency band, a phase parameter as part of the spectral representation of the signals.
9. The signal processor as claimed in claim 8, wherein the signal combiner (6) is configured to use a combination rule in accordance with which a phase factor ϕ associated with the spectral representation of the substitution signal (10) is obtained using a phase factor $\phi_1(k,n)$ of the first (8a) and/or a phase factor $\phi_2(k,n)$ of the second (8b) signal.
10. The signal processor as claimed in claim 9, wherein the signal combiner (6) is configured to use either the phase factor $\phi_1(k,n)$ of the first (8a) or the phase factor $\phi_2(k,n)$ of the second (8b) signal as the phase factor ϕ .
11. The signal processor as claimed in any of the previous claims, configured to generate the substitution signal (10) while using a third signal (8c) which has a predetermined directivity characteristic, wherein
- the signal converter (4) is configured to convert the third signal (8c) to the spectral representation comprising several frequency bands; and
- the signal combiner (6) is configured to obtain a spectral representation of the substitution signal (10) having the predetermined directivity characteristic, which differs from the directivity characteristics of the first (8a), second (8b) and third (8c) signals, the absolute magnitude of an amplitude parameter of the third signal (8c) being

combined in accordance with the combination rule so as to obtain the amplitude parameter of the spectral representation of the substitution signal (10).

- 5 12. The signal processor (2) as claimed in claim 11, wherein the signal combiner is configured (4) to use such a combination rule that in a first signal (8a) having an omnidirectional directivity characteristic, in a second signal (8b) having a dipole-shaped
10 directivity characteristic in a dipole direction, and in a third signal (8c) having a dipole-shaped directivity characteristic in a second dipole direction perpendicular to the dipole direction, a substitution signal is generated which has a dipole-
15 shaped directivity characteristic in a predetermined direction perpendicular to the dipole direction and to the second dipole direction.
- 20 13. The signal processor (2) as claimed in claims 11 or 12, wherein the signal combiner (6) is configured to use a combination rule in accordance with which the magnitude of an amplitude parameter $D(k,n)$ of the substitution signal may be described by the following combination of the amplitude parameters $P_1(k,n)$ of the
25 first signal (8a), $P_2(k,n)$ of the second signal (8b), and $P_3(k,n)$ of the third signal (8c):

$$|D(k,n)| = \sqrt{|P_1(k,n)|^2 - (|P_2(k,n)|^2 + |P_3(k,n)|^2)}.$$

- 30 14. The signal processor (2) as claimed in any of claims 11 to 13, wherein the signal converter (4) is configured to additionally determine, for each frequency band, a phase parameter as part of the spectral representation of the signals.
- 35 15. The signal processor as claimed in claim 14, wherein the signal combiner (6) is configured to use a

combination rule in accordance with which a phase factor ϕ associated with the spectral representation of the substitution signal (10) is obtained using a phase factor $\phi_1(k,n)$ of the first (8a) and/or a phase factor $\phi_2(k,n)$ of the second (8b), and/or a phase factor $\phi_3(k,n)$ of the third (8c) signals.

16. The signal processor (2) as claimed in claim 15, wherein the signal combiner (6) is configured to use, as the phase factor ϕ , either the phase factor $\phi_1(k,n)$ of the first (8a), the phase factor $\phi_2(k,n)$ of the second (8b), or the phase factor $\phi_3(k,n)$ of the third (8c) signals.

17. The signal processor (2) as claimed in claims 9 or 15, wherein the signal combiner (6) is configured to use a combination rule in accordance with which the substitution signal (10) is combined as follows from the magnitude of the amplitude parameter $|D(k,n)|$ and the phase factor ϕ :

$$D(k,n) = |D(k,n)|e^{j\phi}.$$

18. The signal processor as claimed in any of the previous claims, configured to use audio signals and to generate an audio signal.

19. A method of generating a signal having a predetermined spatial directivity characteristic while using a first signal having a known spatial directivity characteristic and a second signal having a known spatial directivity characteristic, comprising:

converting the first signal (8a) and the second signal (8b) to a spectral representation comprising several frequency bands, for which at least one amplitude parameter is determined, respectively; and

combining the spectral representations of the first signal (8a) and of the second signal (8b) in order to obtain a spectral representation of the substitution signal (10) having the predetermined directivity characteristic which differs from the directivity characteristics of the first (8a) and second (8b) signals, an amplitude parameter of a spectral representation of the substitution signal (10) being formed from a combination of the absolute magnitudes of the amplitude parameters of the first signal and of the second signal (8b).

20. A computer program comprising a program code for performing the method as claimed in claim 19, when the program runs on a computer.

21. A signal analysis system for generating a parameterization describing a spatial characteristic of a signal, comprising:

a signal processor as claimed in any of claims 1 to 18; and

a spatial-signal analyzer so as to generate the parameters of the parameterization using the spectral representations of the first signal (8a), of the second signal (8b) and of the substitution signal (10).

22. The signal analysis system as claimed in claim 21, wherein the spatial-signal analyzer is configured to generate a DirAC parameterization.

23. The signal analysis system as claimed in one of claims 21 or 22, wherein

the signal combiner (6) is configured to generate a substitution signal (10) having a toroidal directivity

characteristic from a first signal (8a) having a dipole-shaped directivity characteristic in an X direction and from a second signal (8b) having a dipole-shaped directivity characteristic in a Y direction orthogonal thereto; and

the sound-field analyzer is configured to determine a direction parameter which indicates a direction of incidence of the sound field within the X-Y plane, and a diffuseness parameter which indicates a diffuseness of the signal energy within the X-Y plane.

24. The signal analysis system as claimed in one of claims 21 or 22, wherein

the signal combiner (6) is configured to generate a substitution signal (10) having a toroidal directivity characteristic from a first signal (8a) having an omnidirectional directivity characteristic and from a second signal (8b) having a dipole-shaped directivity characteristic in an X direction; and

the sound-field analyzer is configured to determine a direction parameter which indicates a direction of incidence of the sound field within an X-Y plane, and a diffuseness parameter which indicates a diffuseness of the signal energy within the X-Y plane.

25. The signal analysis system as claimed in one of claims 21 or 22, wherein

the signal combiner (6) is configured to generate a substitution signal (10) having a dipole-shaped directivity characteristic in a Z direction, which is perpendicular to the X and Y directions, from a first signal (8a) having an omnidirectional directivity characteristic, from a second signal (8b) having a dipole-shaped directivity characteristic in an X

direction, and from a third signal (8c) having a dipole-shaped directivity characteristic in a Y direction orthogonal to the X direction; and

- 5 the sound-field analyzer is configured to determine first and second direction parameters which indicate a direction of incidence of the sound field in the 3-dimensional space, and a diffuseness parameter which indicates a diffuseness of the signal energy in the 3-
- 10 dimensional space.

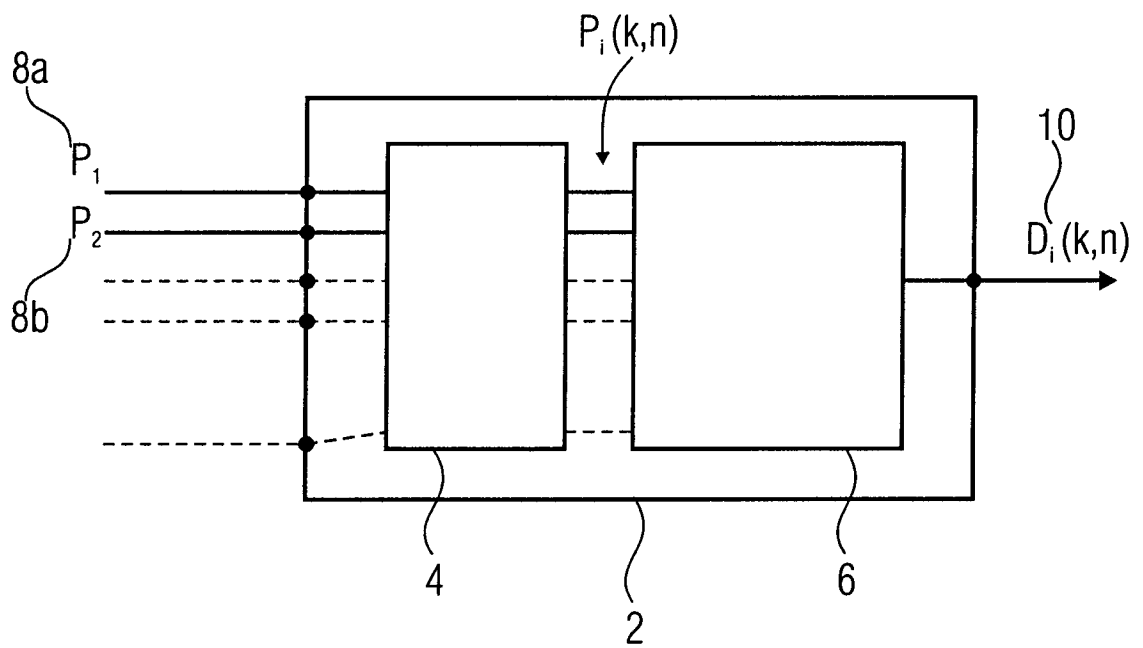


FIGURE 1

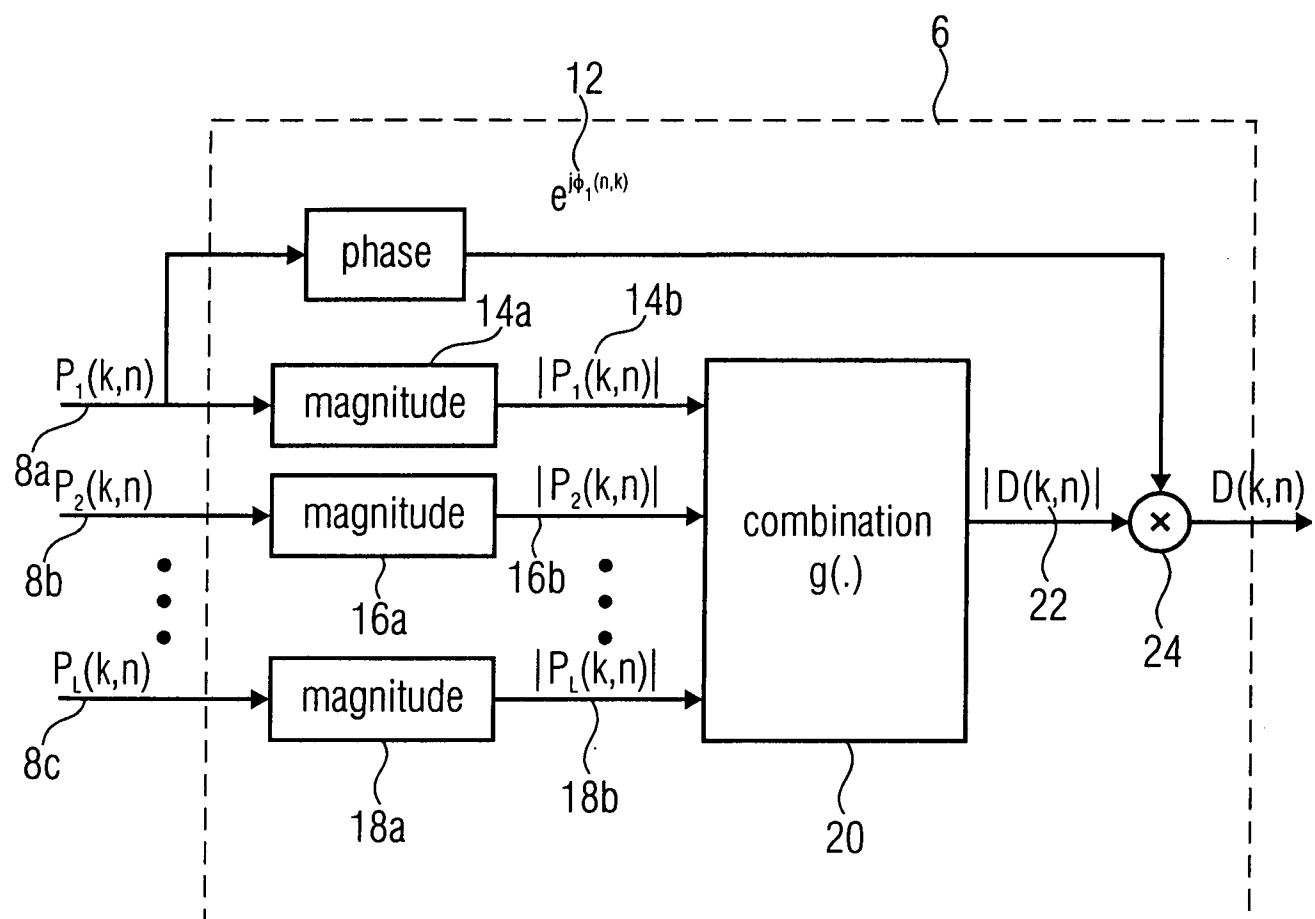


FIGURE 2

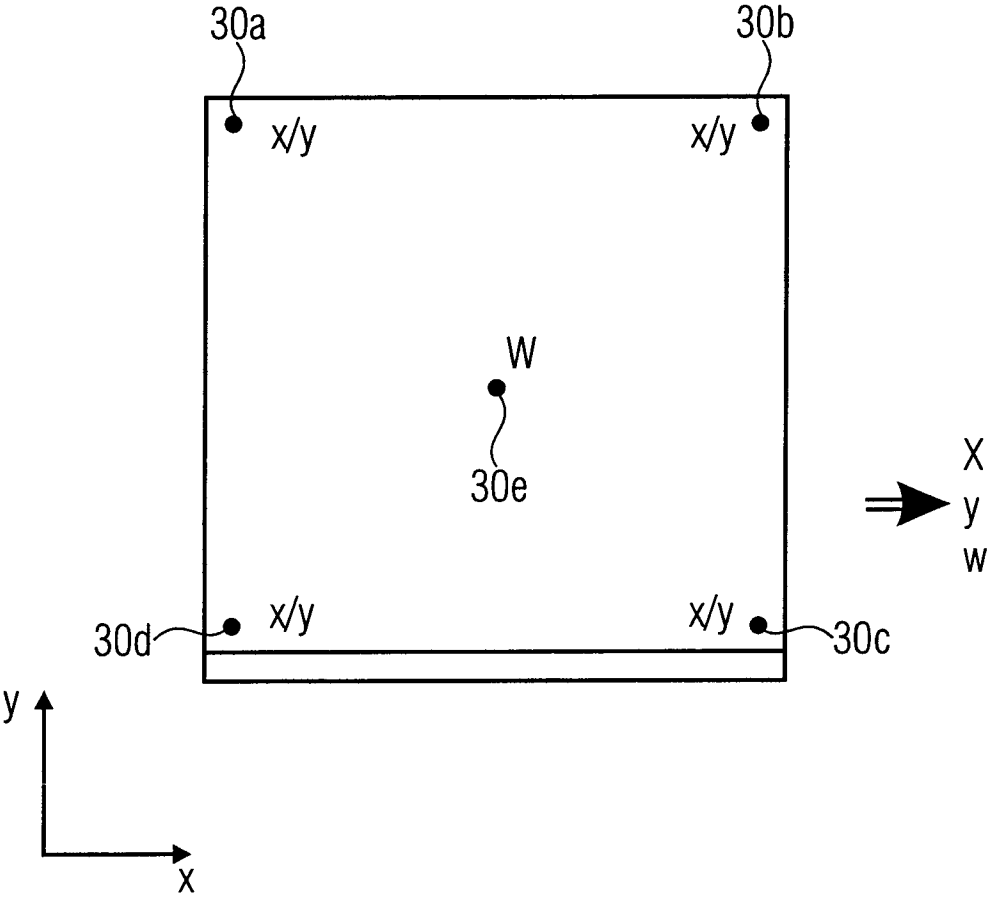


FIGURE 3

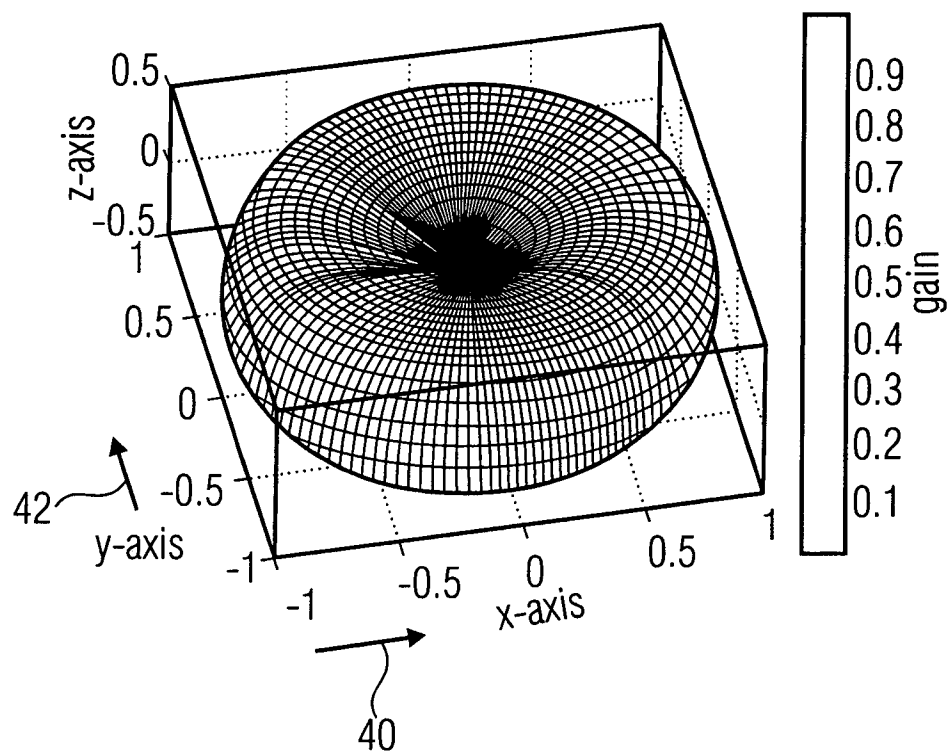


FIGURE 4

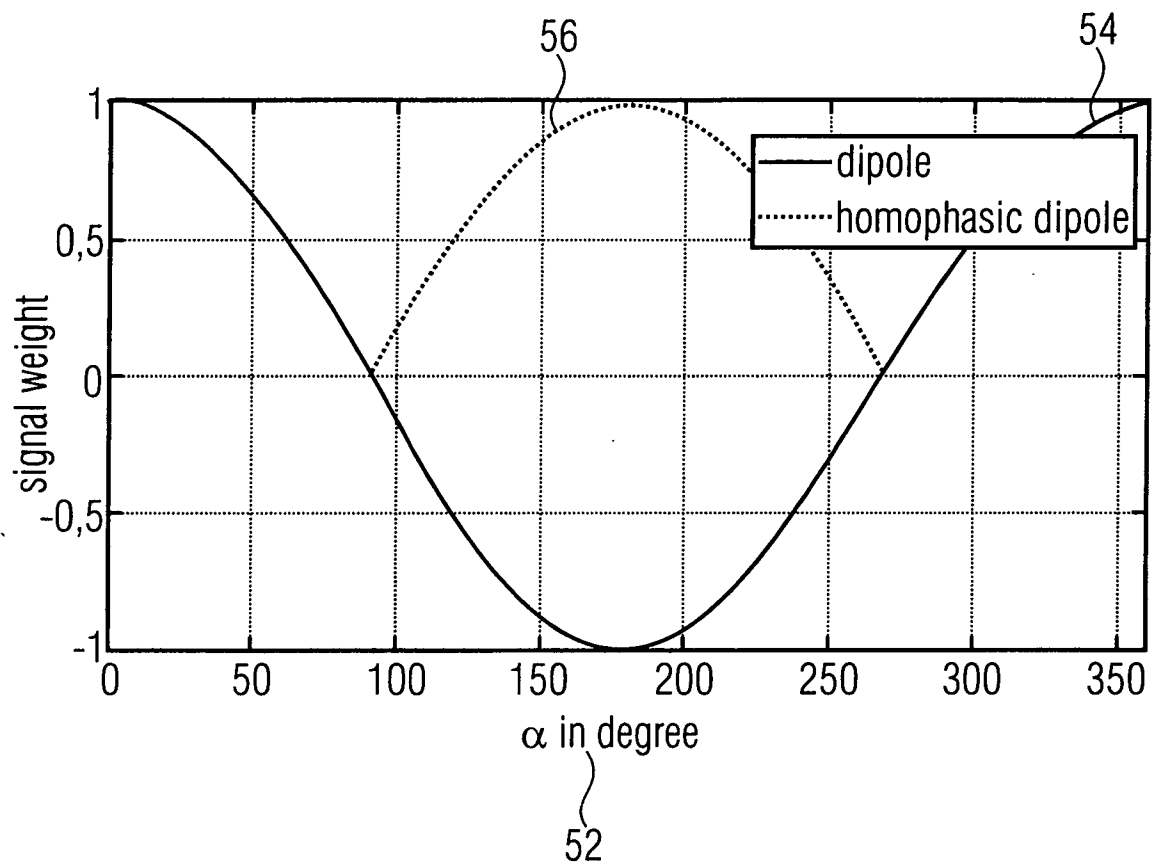


FIGURE 5

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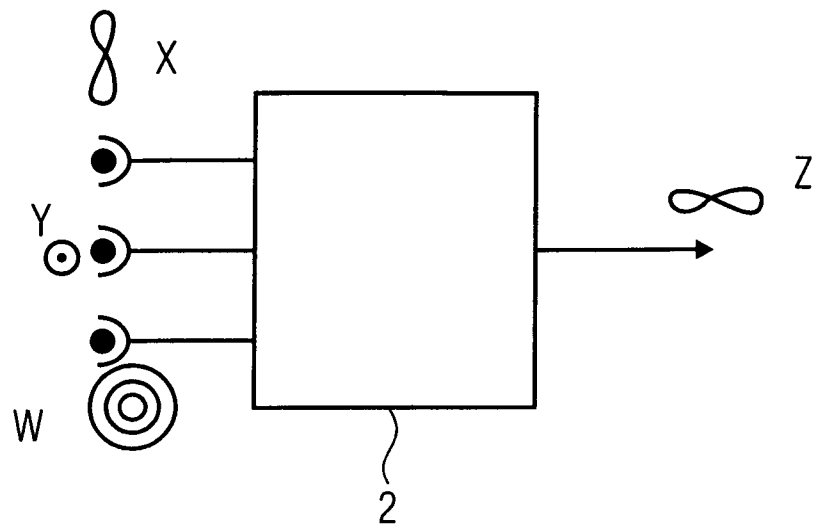


FIGURE 6A

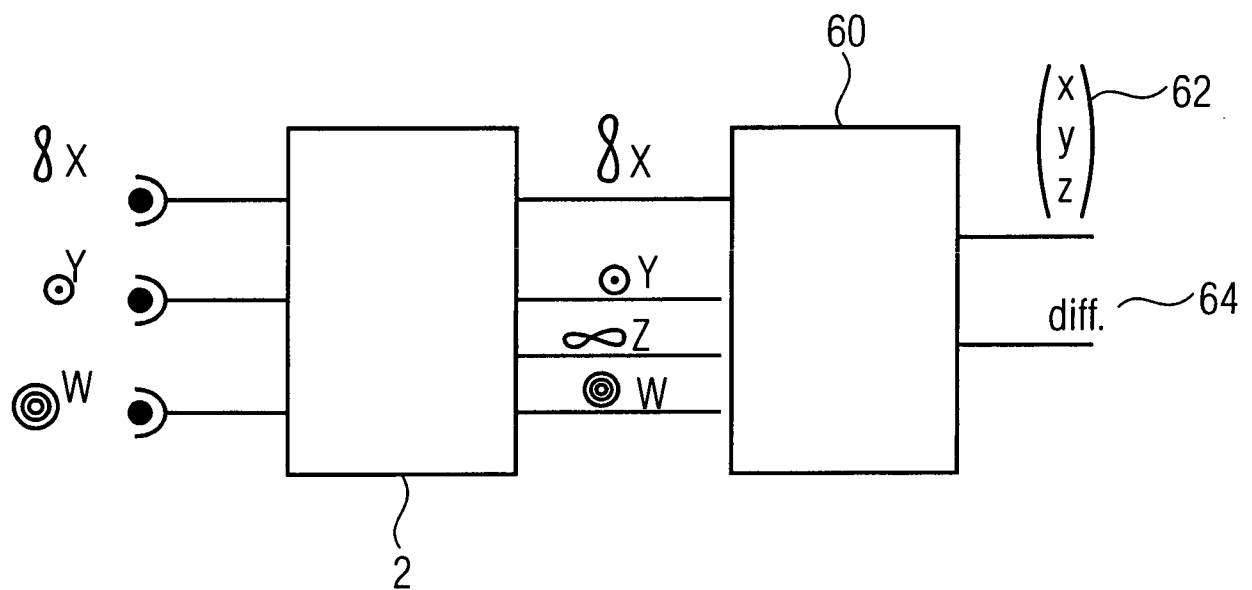


FIGURE 6B

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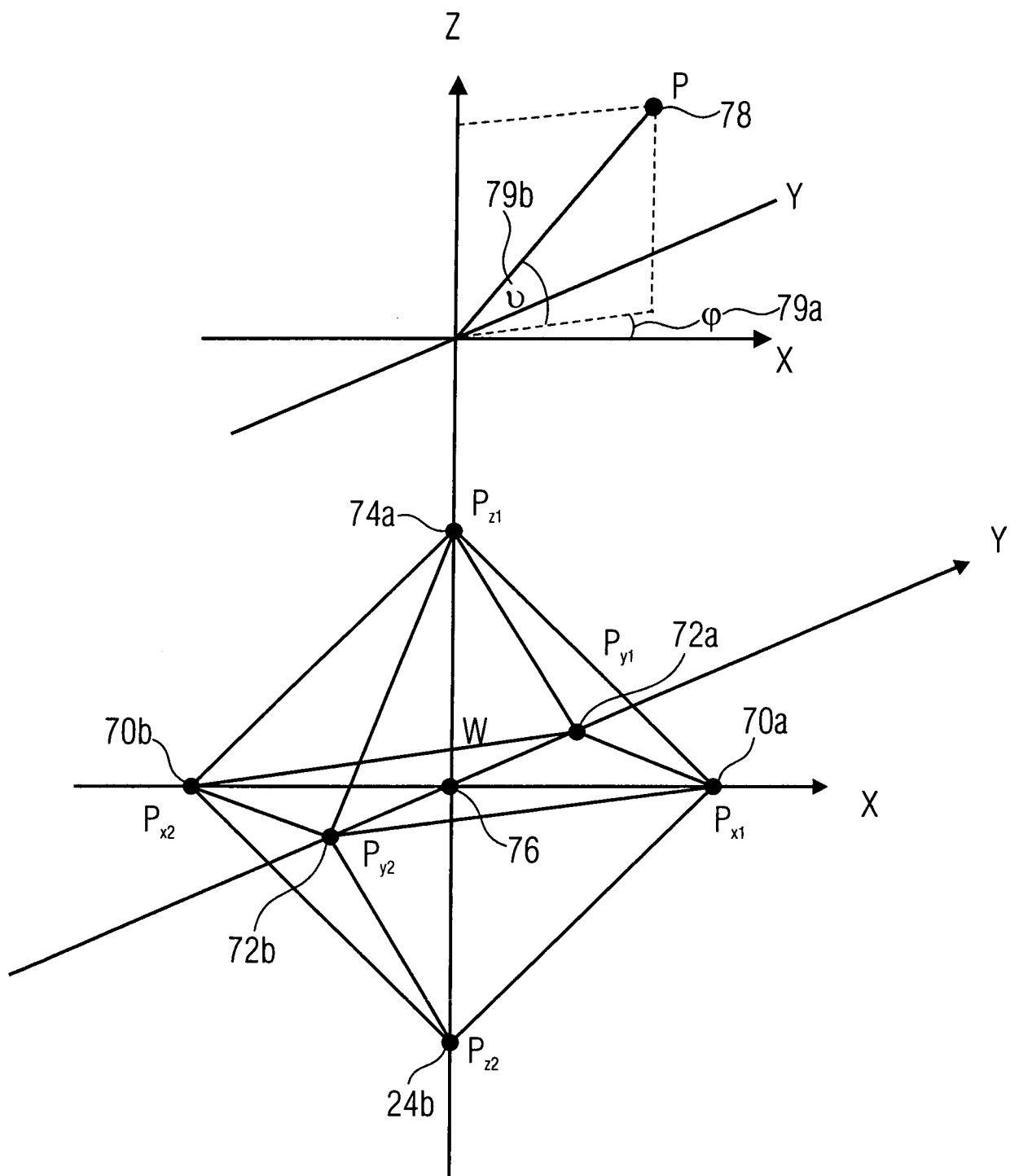


FIGURE 7

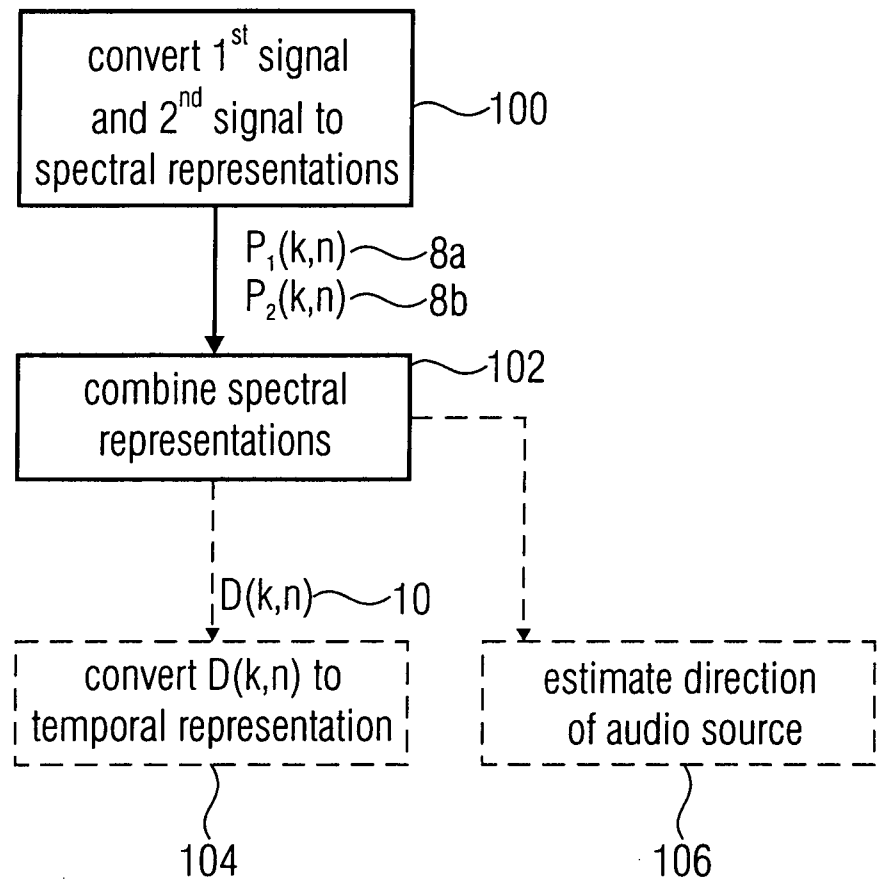


FIGURE 8

INTERNATIONAL SEARCH REPORT

International application No

PCT/EP2008/010658

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04R1/32 H04R1/40

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EP0-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

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X	J. AHONEN, V. PULKKI, T. LOKKI: "Teleconference application and B-format microphone array for Directional Audio Coding" 30TH INTERNATIONAL CONFERENCE: INTELLIGENT AUDIO ENVIRONMENT, [Online] 15 March 2007 (2007-03-15), - 17 March 2007 (2007-03-17) XP002522412 Finland Retrieved from the Internet: URL: http://www.aes.org/tmpFiles/elib/20090424/13927.pdf [retrieved on 2009-04-01] page 5	1,19
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☒ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

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Date of the actual completion of the international search

7 April 2009

Date of mailing of the international search report

12/05/2009

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Heiner, Christoph

INTERNATIONAL SEARCH REPORT

International application No

PCT/EP2008/010658

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

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International application No

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