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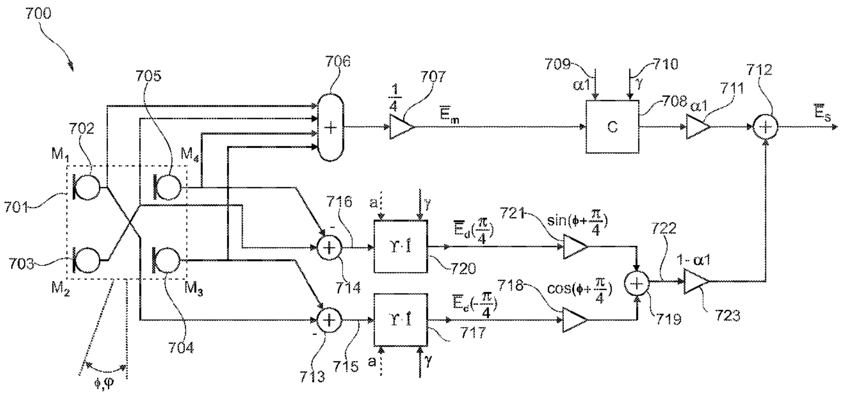

Fig. 7

(57) Abstract: A microphone system is provided, wherein the microphone system comprises a microphone array comprising a plurality of microphone units each adapted to generate a primary signal indicative of an acoustic wave received from the respective microphone unit, a compensation unit, and a combining unit, wherein the microphone system is adapted to generate at least one dipole response and a monopole response from the primary signals, wherein the compensation unit is adapted to generate a compensated monopole signal from the monopole response, and wherein the combining unit is adapted to combine the compensated monopole signal and the at least one dipole response to an output signal.

MICROPHONE SYSTEM AND METHOD OF OPERATING THE SAME

FIELD OF THE INVENTION

The invention relates a microphone system, in particular to a steerable superdirectional microphone system.

Beyond this, the invention relates to a method operating a microphone system.

Moreover, the invention relates to a computer readable medium.

Furthermore, the invention relates to a program element.

BACKGROUND OF THE INVENTION

First-order superdirectional microphones can be constructed out of a linear combination of an omni-directional response and a dipole-response. For a proper combination, an integrator is applied to the dipole response. As a result of this integrator, the dipole response has a large noise-gain for the lower-frequencies. To limit this noise-gain, a leakage factor can be applied to the integrator, but this may lead to a non-flat response for the lower-frequencies.

OBJECT AND SUMMARY OF THE INVENTION

Thus, there may be a need to provide an alternative microphone system and a method of operating the same, a computer readable element, and a program element which may exhibit an improved performance, in particular for lower-frequencies.

In order to meet the need defined above, a microphone system, a method of operating a microphone system, a computer readable medium and a program element according to the independent claims are provided. Further improvements are disclosed in the dependent claims.

According to one aspect of the invention a microphone system is provided, wherein the microphone system comprises a microphone array comprising a plurality of microphone units each adapted to generate a primary signal indicative of an acoustic wave received from the respective microphone unit, a compensation unit, and a combining unit, wherein the microphone system is adapted to generate at least one dipole response and a monopole response from the primary signals, wherein the compensation unit is adapted to generate a compensated monopole signal from the monopole response, and wherein the combining unit is adapted to combine the compensated monopole signal and the at least one dipole response to an output signal.

In particular, the microphone array may comprise at least two microphone units, e.g. two, three, four or eight microphone units. The combining unit may be an adding unit which adds the compensated monopole signal and the dipole response. In particular, the adding may be a weighted adding, i.e. the compensated monopole signal and/or the dipole response may be multiplied by a weighting factor before adding. Furthermore, the compensated monopole signal and/or the monopole response and/or the dipole response may be amplified before the respective signals are combined. Therefore, one or several amplifiers may be included into the microphone system. By providing an array having at least three microphone units uniformly or non-uniformly arranged on a circle it may be possible to provide a steerable microphone system, e.g. a steerable superdirectional microphone system, where the maximum/main-lobe of the superdirectional response can be pointed in any azimuthal direction on the 2D plane.

According to a further aspect of the invention a method of operating a microphone system comprising a microphone array is provided, wherein the method comprises generating at least one dipole response from primary signals of the microphone array, generating a monopole response from the primary signals of the microphone array, generating a compensated monopole signal from the monopole response, and combining the compensated monopole signal and the at least one dipole response.

In particular, the method may further comprise integrating the at least one dipole response before combining it with the compensated monopole signal.

According to another aspect of the invention a program element is provided, which, when being executed by a processor, is adapted to control or carry out a method according to an aspect of the invention.

According to another aspect of the invention a computer-readable medium is provided, in which a computer program is stored which, when being executed by a processor, is adapted to control or carry out a method according to an aspect of the invention.

By providing a microphone system which is adapted in such a way that a monopole response is used to provide a compensated monopole signal, which is then used to compensate a dipole response, it may be possible to compensate a declining in the dipole response by an inclining of the monopole response. In particular, it may be possible to provide a method of operating a microphone system, which enables to flatten the response for a lower frequency range.

The term "compensation unit" may particularly denote any kind of unit which is suitable for enabling compensation in any kind. For example, the term compensation unit may also include a compensation filter which is adapted to filter the monopole response in a specific way, e.g. in a way which may ensure that the compensated monopole signal, which is an output signal of the compensation unit, may enabling a compensation for some deficiencies of the monopole response and/or of another signal or response. In that sense the compensated monopole signal may include the original monopole response in an unchanged or changed form, e.g. amplified, and may further include an additional signal portion derived from or independent from the monopole response. That is, the term "compensated monopole signal" is to understand in a broad sense as any signal which is suitable to compensate for some deficiencies.

The term "microphone array" may particularly denote any kind of spatial arrangement of a plurality of microphone units wherein each of the plurality of microphone units generate a primary signal. The minimum number of microphone units may be two, while every higher number may be suitable. The microphone units may be arranged in a regular pattern on a 2D plane, e.g. uniformly on a circular array or may be arranged in an irregular pattern, e.g. non-uniformly on a circular array. In case of four microphone units the microphone units may be arranged in a rectangular or square pattern.

Next, embodiments of the method of locating information blocks on storage media are described. However, these embodiments also apply to the apparatus for locating information blocks on storage media, the system for locating information blocks on storage media, the computer-readable medium, and the program element.

According to further embodiments the microphone system further comprises an integrator which is adapted to integrate the at least one dipole response.

The application of an integrator may be a suitable measure to increase the signal level of the dipole response. This may be advantageous in case that primary signals, which are used to generate the dipole signal, e.g. by subtracting two primary signals from each other, may be quite similar in terms of phase-differences, e.g. in case the microphone units are arranged close together, so that the primary signals have potentially a very small phase difference.

According to another embodiment of the microphone system the integrator is a leaky integrator exhibiting a leakage value.

In particular, the leakage value may be an input value for the integrator. Additionally, a weighting factor may be used as an input value for the integrator which weighting factor corresponds to the weight of the monopole response in the combination. That is, a parameter that is used for controlling the integrator may be an integrator leakage parameter while an alternative or additional parameter may be a weighting factor.

According to another embodiment of the microphone system the compensation unit is adapted to generate the compensated monopole signal in such a way that at low frequencies a flat output signal is achievable for the angle where the superdirectional response has its maximum/main-lobe.

In particular, the compensation unit may be defined in such a way that for lower frequencies, e.g. between 10 Hz and 1000 Hz or between 100 Hz and 1000 Hz, a unit-response is obtained.

According to another embodiment of the microphone system the microphone array is a small microphone array.

The term "small" may particular denote the case in which the distance between adjacent microphone units is smaller than the typical wavelengths of the acoustic waves or sound waves which are measured by the microphone units.

According to another embodiment of the microphone system the compensation unit is a compensation filter.

According to another embodiment of the microphone system the compensation filter is a recursive filter.

In particular, the recursive filter may be formed by:

$$C_N(\alpha_1,\gamma) = \begin{cases} \dfrac{1-\gamma_2\cdot e^{-j\theta}}{1-\gamma\cdot e^{-j\theta}}, & \text{for} \quad N=0 \\[2ex] \dfrac{1-\gamma_2\cdot e^{-j\theta N}}{1-[N(\gamma-1)+1]\cdot e^{-j\theta N}}, & \text{for} \quad N\geq 1, \end{cases}$$

wherein j denotes the imaginary unit, $C_N(\alpha_1,\gamma)$ represents the filter, α_1 represents the weighting factor of the monopole response, θ is given by $\theta = 2\pi f/f_s$ wherein f_s is the sampling frequency, γ is the leakage factor of a recursive N'th order leaky integrator and γ_2 is given by:

$$\gamma_2 = \begin{cases} \frac{\alpha_1 + (\gamma - 1)}{\alpha_1} & \text{for} \quad N = 0 \\ \frac{\alpha_1 + N \cdot (\gamma - 1)}{\alpha_1} & \text{for} \quad N \geq 1. \end{cases}$$

According to another embodiment of the microphone system a leakage value of an integrator and/or a weighting factor is used as input values for the compensation filter.

According to another embodiment the microphone system further comprises an integrator which is adapted to integrate the at least one dipole signal, wherein the compensation filter is a linear combination of at least two compensation filters,

In particular, the two compensation filters may be a so called Turin integrator and a so called Simpson integrator and/or the integrator may be a so called Al-Alaoui integrator.

Summarizing, a gist of an aspect of the invention may be seen in the provision of a microphone system that may exhibit an improved performance in particular in the lower-frequencies range. The microphone system may comprise a small microphone array including at least two microphone units, but preferably more than two microphone units to enable an azimuth steerable microphone system, each generating a primary signal. From the primary signals a monopole response and at least one dipole response may be generated. The dipole response or the dipole responses, e.g. two dipole responses, may be integrated by using a leaky integrator and the integrated dipole response(s) may be added to a compensated monopole signal which is generated from the monopole response, e.g. by filtering the same. The compensated monopole signal may be generated in such a way that a decreasing of the integrated dipole responses at lower frequencies is compensated by an increasing of the compensated monopole signal at lower frequencies so that a flat response may be enabled for the whole range of frequencies of interest, e.g. the range of human hearing. A microphone system according to an aspect of the invention may be applied in car-radio chips of Car Entertainment Systems, for example and may be also beneficial for MEMS microphone technology.

The aspects and embodiments defined above and further aspects of the invention are apparent from the examples of embodiments to be described hereinafter and are explained with reference to these examples of embodiments. It should be noted that features described in connection with a specific embodiment or aspect may be combined with another embodiment or another aspect.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be described in more detail hereinafter with reference to examples of embodiments but to which the invention is not limited.

Fig. 1 schematically illustrates the geometry of a four microphone array.

Fig. 2 schematically illustrates a comparison of different discrete-integrators.

Fig. 3 schematically illustrates directional responses for f = 1/4 f_s.

Fig. 4 schematically illustrates a comparison of different discrete-integrators with leakage factor of 0.95.

Fig. 5 schematically illustrates target response for combined monopole/dipole for a leakage factor of 0.95 and a weighting factor of 0.5.

Fig. 6 schematically illustrates target response for combined monopole/dipole with compensation filter for a leakage factor of 0.95 and a weighting factor of 0.5.

Fig. 7 schematically illustrates a microphone system according to an embodiment.

DESCRIPTION OF EMBODIMENTS

The illustration in the drawing is schematic. In different drawings, similar or identical elements are provided with similar or identical reference signs. In connection with Fig. 1 to Fig. 5 some basic principles of superdirectional microphones are described which may be helpful for understanding of the invention.

Fig. 1 schematically illustrates the geometry of a four microphone array 100. In particular, a (steerable) first-order superdirectional microphone can be implemented via a combined monopole and dipole. For this, four omnidirectional microphone units or microphones in a planar configuration may be used, which are depicted in Fig. 1 as 101, 102, 103 and 104. As can be seen, the spacing between two diagonal microphones (e.g. distance between microphone 102 and microphone 104) is exactly $\sqrt{2}$ times the spacing between two non-diagonal microphones (e.g. distance between microphone 101 and 102).

The normalized superdirectional microphone-response (with a maximum response/main-lobe on ϕ radians) may be formulated as:

$$\overline{E}_s(\phi,\varphi,\alpha_1) = \alpha_1 \cdot \overline{E}_m(\varphi) + (1-\alpha_1)\cdot \overline{E}_d(\phi,\varphi), \qquad (1)$$

where the first-order characteristic is determined by α_1. Furthermore, $\overline{E}_d(\phi,\varphi)$ is the normalized dipole-response oriented with its maximum to ϕ and $\overline{E}_m(\varphi)$ is the normalized monopole response.

The normalized (frequency-independent) dipole-response may be computed as:

$$\overline{E}_d(\phi,\varphi) = I_{ideal}\cdot \Upsilon \cdot E_d(\phi,\varphi), \qquad (2)$$

where:

$$E_d(\phi,\varphi) = \cos\left(\phi+\frac{\pi}{4}\right)\cdot E_d(-\pi/4,\varphi) + \sin\left(\phi+\frac{\pi}{4}\right)\cdot E_d(\pi/4,\varphi), \qquad (3)$$

and where (for small values of Ω, where the distance d is smaller than the wavelength of the sound):

$$\begin{aligned} E_d(\pi/4,\varphi) &= E_2 - E_4 \\ &= S\cdot\left(e^{j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi-\frac{\pi}{4})} - e^{-j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi-\frac{\pi}{4})}\right), \\ &= j\cdot 2\cdot S\cdot \sin\left(\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi-\frac{\pi}{4}\right)\right) \qquad (4) \\ E_d(\pi/4,\varphi) &\approx j\cdot S\cdot 2\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi-\frac{\pi}{4}\right) \qquad (5) \end{aligned}$$

$$\begin{aligned} E_d(-\pi/4,\varphi) &= E_3 - E_1 \\ &= S\cdot\left(e^{j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi+\frac{\pi}{4})} - e^{-j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi+\frac{\pi}{4})}\right), \\ &= j\cdot 2\cdot S\cdot \sin\left(\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi+\frac{\pi}{4}\right)\right), \qquad (6) \\ E_d(-\pi/4,\varphi) &\approx j\cdot S\cdot 2\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi+\frac{\pi}{4}\right). \qquad (7) \end{aligned}$$

with φ the angle of incidence of sound, E_i the signal picked up by each of the microphone units M_i, i.e. a primary signal, S the sensitivity of each of the microphones and Ω given by:

$$\Omega = \frac{\omega\cdot d}{2\cdot c}, \qquad (8)$$

with ω the frequency (in radians), d the distance between the microphones and c the speed of sound.

Furthermore I_{ideal} is an ideal integrator, defined as:

$$I_{ideal} = \frac{1}{j\omega}, \qquad (9)$$

and Υ is an extra compensation term defined as:

$$\Upsilon = \frac{c}{\sqrt{2}\cdot d}, \qquad (10)$$

The normalized monopole-response $\overline{E}_m(\varphi)$ may be computed as:

$$\begin{aligned}\overline{E}_m(\varphi) &= \frac{1}{4}\cdot\sum_{i=1}^{4} E_i \\ &= \frac{1}{4}\cdot S\cdot\Big[e^{j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi-\frac{\pi}{4})} + e^{-j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi-\frac{\pi}{4})} \\ &\quad + e^{j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi+\frac{\pi}{4})} + e^{-j\cdot\sqrt{2}\cdot\Omega\cdot\cos(\varphi+\frac{\pi}{4})}\Big], \\ &= \frac{1}{2}\cdot S\cdot\left[\cos\left(\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi-\frac{\pi}{4}\right)\right) + \right. \\ &\quad \left.\cos\left(\sqrt{2}\cdot\Omega\cdot\cos\left(\varphi+\frac{\pi}{4}\right)\right)\right]. \qquad (11)\end{aligned}$$

The overline indicates a normalized response with a maximum response S (equal to the response of a single sensor or microphone unit).

The integrator is required to remove the jω-dependency in the dipole response.

The method described above may be the simplest way to construct a steerable first-order microphone (via parameter ϕ) with a variable characteristic (via parameter α_1). Although methods like delay-and-subtract, Linear Constrained Minimum Variance (LCMV) and Generalized Sidelobe Canceller (GSC) may also be modified to obtain steerable capabilities, they may require (FIR) filters that need to be recomputed for different values of ϕ and α_1, which is computationally unattractive.

An N'th order approximation of the ideal integrator in discrete-time may be written as:

$$y[nT] = \begin{cases} y[(n-1)T] + b_0\cdot x[(n-r)T], & \text{if} \quad N = 0 \\ y[(n-N)T] + \sum_{k=0}^{N} b_k\cdot x[(n-k)T], & \text{if} \quad N \geq 1, \end{cases} \qquad (12)$$

where N is the order of the integrator, $T = f_s^{-1}$ with f_s the sampling frequency, $r \in \{0, 1\}$ and b_k satisfies the symmetry condition:

$$b_k = b_{N-k}. \qquad (13)$$

In frequency-domain, this yields:

$$I_N = \begin{cases} T\cdot\dfrac{b_0\cdot e^{-j\theta r}}{1-e^{-j\theta}}, & \text{for} \quad N = 0 \\ \dfrac{T\cdot\sum_{k=0}^{N} b_k\cdot e^{-j\theta k}}{1-e^{-j\theta N}}, & \text{for} \quad N \geq 1, \end{cases} \qquad (14)$$

where $\theta = 2\pi f / f_s$ lies in the fundamental interval $-\pi \leq \theta \leq \pi$.

The N'th order optimal discrete integrator coefficients may be chosen in such a way that the integrator response is as close as possible to the ideal (discrete-time) response:

$$\hat{I}_N \approx \frac{T}{j\theta}. \qquad (15)$$

For N = 0, $b_0 = 1$ is obtained when trying to match the (magnitude) integrator response to the ideal response for low frequencies via Taylor approximation. For r = 0, this results in the following frequency-response for the zero-order integrator.

$$\hat{I}_0 = \frac{T}{1 - e^{-j\theta}}. \qquad (16)$$

For r = 0, $\hat{I}_0$ is called the backward Euler (rectangular) integrator. For r = 1, the integrator is of the forward type. Furthermore, the phase-shift of this integrator (for both r = 0 and r = 1) is not fixed to $-\pi/2$ for all frequencies, as is the case for the ideal integrator of Eq. (9).

For N = 1, $b_0 = 1$ is obtained when trying to match the integrator-response to the ideal response for low frequencies in the same way as for the Euler integrator. This results in the following frequency-response:

$$\hat{I}_1 = \frac{1}{j} \cdot \left(\frac{\frac{T}{2} \cdot \cos(\theta/2)}{\sin(\theta/2)} \right). \qquad (17)$$

This response $\hat{I}_1$ is called the Turin (or trapezoid) integrator. It can be seen that for small values of θ, the Turin integrator approximates the ideal integrator. In contrast to the Euler integrator, the phase-shift of this first-order is exactly $-\pi/2$, just as for the ideal integrator.

For N = 2, $b_0 = 1/3$ and $b_1 = 4/3$ is obtained when trying to match the integrator response to the ideal response for low frequencies in the same way as for the Euler and Turin integrator. This results in the following frequency-response:

$$\hat{I}_2 = \frac{1}{j} \cdot \left(\frac{\frac{2T}{3} + \frac{T}{3} \cdot \cos(\theta)}{\sin(\theta)} \right). \qquad (18)$$

This response $\hat{I}_2$ is called the Simpson integrator. Just as for the Turin integrator, the phase-shift is exactly $-\pi/2$.

Higher-order N >2 integrators may not be of great interest, as their phase response is not fixed to $-\pi/2$, as is the case for the Turin and the Simpson integrator. Furthermore, they may also show numerical instabilities due to the fact that there are complex poles on the unit-circle.

Fig. 2 shows a comparison of the magnitude responses of the three above mentioned integrator types (with f_s = 48000 Hz), i.e. line 201 for an Euler integrator (r = 0), line 202 for a Turin integrator, and line 203 for a Simpson integrator, with respect to the ideal integrator response (line 204). As can be seen in Fig. 2, the ideal integrator response lies in between responses of the Turin and the Simpson integrator. Therefore, it may be suitable to construct a combined Turin and Simpson integrator that weights the outputs of each individual integrator:

$$\hat{I}_{AA}(a) = a \cdot \hat{I}_2 + (1 - a) \cdot \hat{I}_1, \qquad (19)$$

where a is the weighting factor. This integrator may be called Al-Alaoui integrator. Just as the Turin and the Simpson integrator, this Al-Alaoui integrator has a phase response of exactly -π/2 radians for every value of a.

If the superdirectional response $\overline{E}_s(\phi, \varphi, \alpha_1)$ is computed with $\alpha_1 = 0.5$ and $\phi = 0$ in combination with the discrete integrator $\hat{I}_N$, with N = 0, 1, 2, as in Eq. (16), Eq. (17) and Eq. (18) and $f = \frac{1}{4} f_s$ the result as shown in Fig. 3 is obtained. In particular, the line 301 indicates the result for the Euler integrator (r = 0), line 302 indicates the result for the Turin integrator, line 303 indicates the result for the Simpson integrator while line 304 indicates the result for the ideal integrator or ideal response.

As can be seen, the response of the Euler integrator poorly matches the ideal response, while this match is improved for the Simpson integrator. For the Euler integrator it holds that although the magnitude response well approximates the ideal integrator, the phase response is not perfect, which results in a bad response in Fig. 3. For frequencies close to 1/2 f_s (i.e. the Nyquist frequency), the response of the Turin integrator yields the omni-directional response but attenuated with -6 dB for $\alpha_1 = 0.5$. This can be easily seen, as the Turin integrator quickly drops to zero for frequencies close to the Nyquist frequency. In contrast, the Simpson integrator rises quickly for frequencies close to the Nyquist frequency. This yields a dominant dipole response (with a large gain-factor) for the Simpson integrator at these frequencies.

A leakage-term can be added to discrete time integrators. Such a leakage factor may alleviate the noise-gain problem that occurs for very low frequencies when compensating for the jω dependency in the dipoles. Without a leakage-factor, the (white) sensor-noise would be amplified enormously which is undesired. Furthermore, adding a leakage term to the integrator may help in improving the numerical-stability for higher order-integrators.

The leakage factor γ, which is a value lower, but close to 1, is applied in the feedback-branch of the discrete-time filter. If the same factor γ would be applied in the feedback-branch for integrators with different order N, these integrators would have a different response for the low-frequency range. For θ = 0, the response is given by:

$$\hat{I}_N(\theta = 0) = \frac{T \cdot N}{1-\gamma}. \qquad (20)$$

Now the response may be normalized for the low-frequencies which results in:

$$y[nT] = \begin{cases} \gamma \cdot y[(n-1)T] + b_0 \cdot x[(n-r)T], & \text{if} \quad N = 0 \\ [N(\gamma-1)+1] \cdot y[(n-N)T] + \sum_{k=0}^{N} b_k \cdot x[(n-k)T], & \text{if} \quad N \geq 1. \end{cases} \qquad (21)$$

In the frequency domain, this yields:

$$\hat{I}_N(\gamma) = \begin{cases} T \cdot \frac{b_0 \cdot e^{-j\theta r}}{1-\gamma \cdot e^{-j\theta}}, & \text{if} \quad N = 0 \\ \frac{T \cdot \sum_{k=0}^{N} b_k \cdot e^{-j\theta k}}{1-[N(\gamma-1)+1] \cdot e^{-j\theta N}}, & \text{if} \quad N \geq 1. \end{cases} \qquad (22)$$

This response of the integrators with leakage term is shown in Fig. 4. Here a leakage factor γ of 0.95 is used and again f_s is set to 48000 Hz. Again Fig. 4 shows the responses in dB over the frequency for the four different integrators with leakage factor γ = 0.95. In particular, line 402 shows the response for the Turin integrator, line 403 shows the response for the Simpson integrator, line 404 shows the response for the ideal integrator, while line 405 shows the response for the Al-Alaoui integrator with a = 0.65. The curves for the leaky-integrators nicely coincide for the low-frequencies, while for the high-frequencies the Al-Alaoui integrator lies in between the Turin and the Simpson integrator.

Fig. 5 schematically illustrates target response for combined monopole/dipole for a leakage factor of 0.95 and a weighting factor of 0.5. In particular, Fig. 5 is showing the target response over frequency for the Turin integrator (line 502), the Simpson integrator (line 503) and the Al-Alaoui integrator for a = 0.65 (line 505). Applying the basic principle of a combined monopole/dipole with an integrator with a leakage term may lead to a non-flat target response for $\alpha_1 \neq 1$, since for the lower frequencies the monopole is dominant, but is always weighted with α_1. For example, for $\alpha_1 = 1/2$, the lower frequency target response approaches -6 dB, while ideally it should be 0 dB.

According to an embodiment of the invention a compensation filter is applied to the monopole to compensate for imperfect target response for lower frequencies. In particular, the compensation filter may be applied as follows:

$$\overline{E}_s(\phi, \varphi, \alpha_1) = \alpha_1 \cdot C_N(\alpha_1, \gamma) \cdot \overline{E}_m(\varphi) + (1-\alpha_1) \cdot \overline{E}_d(\phi, \varphi), \qquad (23)$$

where $C_N(\alpha_1,\gamma)$ is a compensation filter for the monopole component that is defined in such a way that for lower-frequencies, a unit-response is obtained. The compensation filter is a recursive filter, defined as:

$$C_N(\alpha_1,\gamma) = \begin{cases} \frac{1-\gamma_2 \cdot e^{-j\theta}}{1-\gamma \cdot e^{-j\theta}}, & \text{for} \quad N=0 \\ \frac{1-\gamma_2 \cdot e^{-j\theta N}}{1-[N(\gamma-1)+1] \cdot e^{-j\theta N}}, & \text{for} \quad N \geq 1, \end{cases} \tag{24}$$

where it can be shown that for a flat low-frequency response and $\alpha_1 > 0$, γ_2 is computed as:

$$\gamma_2 = \begin{cases} \frac{\alpha_1+(\gamma-1)}{\alpha_1} & \text{for} \quad N=0 \\ \frac{\alpha_1+N \cdot (\gamma-1)}{\alpha_1} & \text{for} \quad N \geq 1. \end{cases} \tag{25}$$

It should be noted that $C_0(\alpha_1, \gamma) = C_1(\alpha_1, \gamma)$.

For the Al-Alaoui integrator, the compensation filter is constructed out of a linear combination of the compensation filter for the Turin and the Simpson integrator as:

$$C_{AA,comp}(\alpha_1,\gamma,a) = a \cdot C_{2,comp}(\alpha_1,\gamma) + (1-a) \cdot C_{1,comp}(\alpha_1,\gamma). \tag{26}$$

The target-responses for $\alpha_1 = 0.5$, $f_s = 48000$ Hz and $\gamma = 0.95$ with the application of the compensation filter on the monopole is shown in Fig. 6. As can be seen in Fig. 6 the compensation filter nicely flattens the low-frequency response for all the integrator types. In particular, line 602 relates to a response of a Turin integrator, line 603 relates to the response of a Simpson integrator, while line 605 relates to the response of an Al-Alaoui integrator.

Fig. 7 schematically illustrates a microphone system 700 according to an embodiment. In particular, Fig. 7 shows a microphone array 701 comprising four microphone units or elements 702, 703, 704, and 705 which are arranged in a square pattern. Each of the microphone units generates a primary signal which can be used to generate dipole and monopole responses. For generating the monopole response the primary signals are connected to an adder schematically depicted as box 706 in Fig. 7. An output signal of the adder 706 is amplified by amplifier 707 to generate the monopole response $\overline{E}_m$. In case of four microphone units the amplifier may have an amplification factor of 1/4. The monopole response $\overline{E}_m$ is then connected to a compensation unit which may be formed by a compensation filter 708 having as steering inputs the weighting factor α_1 and the leakage factor γ, which is indicated by the arrows 709 and 710, to generate a compensated monopole signal which may be amplified by an amplifier 711 and which forms one input of a first combination unit 712, e.g. an adder.

Furthermore, the primary signals are connected to two adding or subtracting units 713 and 714 to generate two dipole responses. In particular, the primary signal of the first microphone unit 702 and of the third microphone unit 704 is connected to the first subtracting unit 713 in which the primary signal of the first microphone unit is subtracted from the primary signal of the third microphone unit to generate a first dipole response indicated by arrow 715. Additionally, the primary signal of the second microphone unit 703 and of the fourth microphone unit 705 is connected to the second subtracting unit 714 in which the primary signal of the fourth microphone unit is subtracted from the primary signal of the second microphone unit to generate a second dipole response indicated by arrow 716. The first dipole response 715 is inputted into a first integrator, e.g. a Turin, a Simpson or an Al-Alaoui integrator 717 which has as steering inputs the leakage factor γ as well, and optionally in case of the Al-Alaoui integrator the weighting factor a, to generate a first dipole response $\overline{E}_d(-\pi/4)$ which may then be amplified by another amplifier 718, having an amplification factor of $\cos(\phi+\frac{\pi}{4})$, wherein the amplified response is inputted into a second combining unit 719, e.g. an adder. The second dipole response 716 is inputted into a second integrator, e.g. a Turin, a Simpson or an Al-Alaoui integrator 720 which has as steering inputs the leakage factor γ as well, and optionally in case of the Al-Alaoui integrator the weighting factor a, to generate a second dipole response $\overline{E}_d(\pi/4)$ which may then be amplified by another amplifier 721, having an amplification factor of $\sin(\phi+\frac{\pi}{4})$, wherein the amplified response is inputted into the second combining unit 719. The second combining unit 719 then combines the two inputs, e.g. adds the same, to generate a total dipole response 722 which may be amplified by another amplifier 723 having as an amplification factor $1-\alpha_1$. An output of the amplifier 723 then forms a second input to the first combining unit 712 which then outputs the output signal $\overline{E}_s$ which forms the output of the azimuth steerable superdirectional microphone system 100. Of course more than four microphone units may be used and possibly three dipoles may be generated by using microphones in a 3D geometry, like in a tetrahedron configuration.

Thus, Fig. 7 schematically shows a complete microphone system with the compensation filter C on the monopole response.

The angle ϕ, where a maximum-response of the superdirectional microphone is obtained, can be varied by recomputing the weights $\sin(\phi+\frac{\pi}{4})$ and $\cos(\phi+\frac{\pi}{4})$.

As can be seen, the compensation filter C may require the parameter α_1, indicative for the first-order characteristic, and constructible via a weighted combination of the monopole response and the dipole response and the integrator leakage parameter γ, while the block with the integrator $\hat{I}$ and the compensation-factor Υ may only require the integrator leakage parameter γ (and also the parameter a, if the Al-Alaoui integrator is used).

Finally, it should be noted that the above-mentioned embodiments illustrate rather than limit the invention, and that those skilled in the art will be capable of designing many alternative embodiments without departing from the scope of the invention as defined by the appended claims. In the claims, any reference signs placed in parentheses shall not be construed as limiting the claims. The word "comprising" and "comprises", and the like, does not exclude the presence of elements or steps other than those listed in any claim or the specification as a whole. The singular reference of an element does not exclude the plural reference of such elements and vice-versa. In a device claim enumerating several means, several of these means may be embodied by one and the same item of software or hardware. The mere fact that certain measures are recited in mutually different dependent claims does not indicate that a combination of these measures cannot be used to advantage.

CLAIMS:

1. A microphone system, the microphone system comprising:

- microphone array comprising a plurality of microphone units each adapted to generate a primary signal indicative of an acoustic wave received from the respective microphone unit,
- compensation unit, and
- combining unit,

wherein the microphone system is adapted to generate at least one dipole response and a monopole response from the primary signals,

wherein the compensation unit is adapted to generate a compensated monopole signal from the monopole response, and

wherein the combining unit is adapted to combine the compensated monopole signal and the at least one dipole response to an output signal.

2. The microphone system according to claim 1, further comprising:

- an integrator which is adapted to integrate the at least one dipole response.

3. The microphone system according to claim 2, wherein the integrator is a leaky integrator having a leakage value.

4. The microphone system according to claim 1, wherein the compensation unit is adapted to generate the compensated monopole signal in such a way that at low frequencies a flat output signal is achievable.

5. The microphone system according to claim 1, wherein the microphone array is a small microphone array.

6. The microphone system according to claim 1, wherein the compensation unit is a compensation filter.

7. The microphone system according to claim 6, wherein the compensation filter is a recursive filter.

8. The microphone system according to claim 6, wherein a leakage value of an integrator and/or a monopole/dipole weighting factor is used as input values for the compensation filter.

9. The microphone system according to claim 6, further comprising:
- an integrator which is adapted to integrate the at least one dipole signal,

wherein the compensation filter is a linear combination of at least two compensation filters,

10. A method of operating a microphone system comprising a microphone array, the method comprising:
- generating at least one dipole response from primary signals of the microphone array,
- generating a monopole response from primary signals of the microphone array,
- generating a compensated monopole signal from the monopole response, and
- combining the compensated monopole signal and the at least one dipole response.

11. The method according to claim 10, further comprising:
- integrating the at least one dipole response before combining with the compensated monopole signal.

12. A program element, which, when being executed by a processor, is adapted to control or carry out a method according to claim 10.

13. A computer-readable medium, in which a computer program is stored which, when being executed by a processor, is adapted to control or carry out a method according to claim 10.

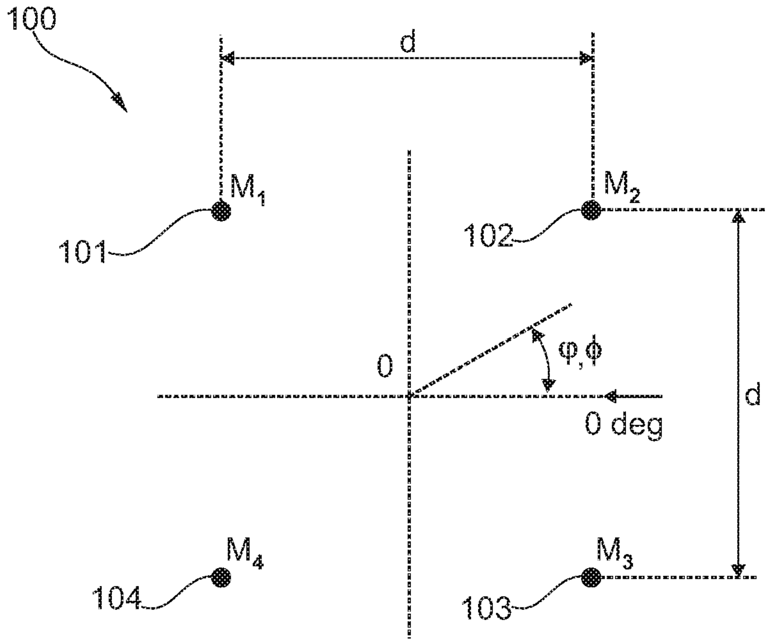
100
d
M1
M2
101
102
0
φ,ϕ
0 deg
d
M4
M3
104
103

Fig. 1

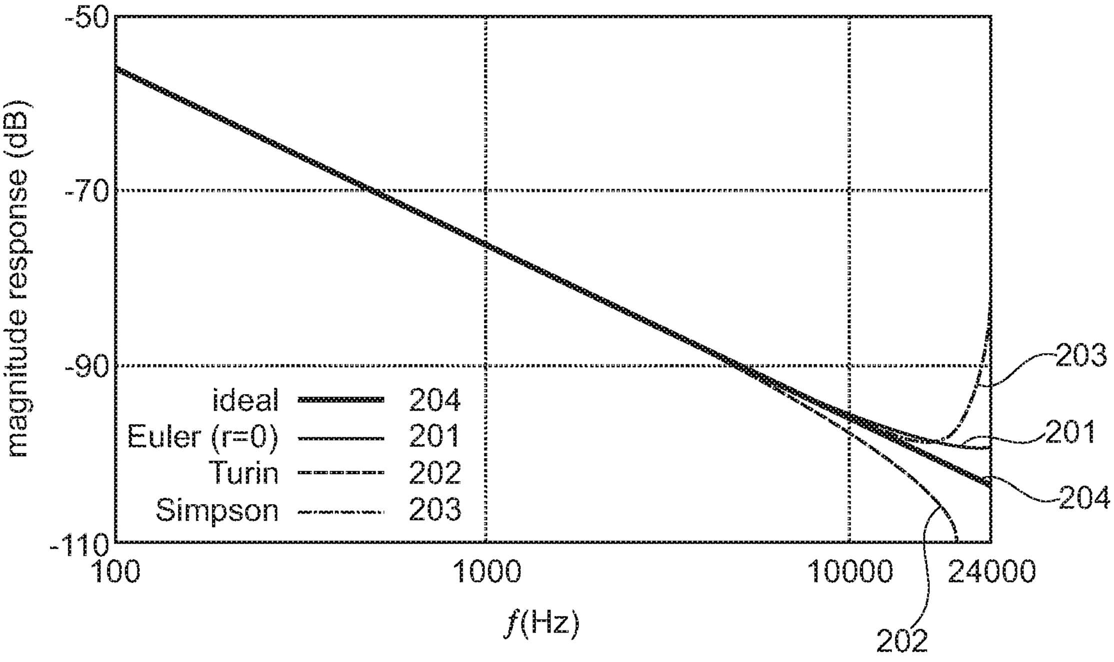
-50
-70
-90
-110
magnitude response (dB)
ideal 204
Euler (r=0) 201
Turin 202
Simpson 203
100
1000
10000
24000
f(Hz)
203
201
204
202

Fig. 2

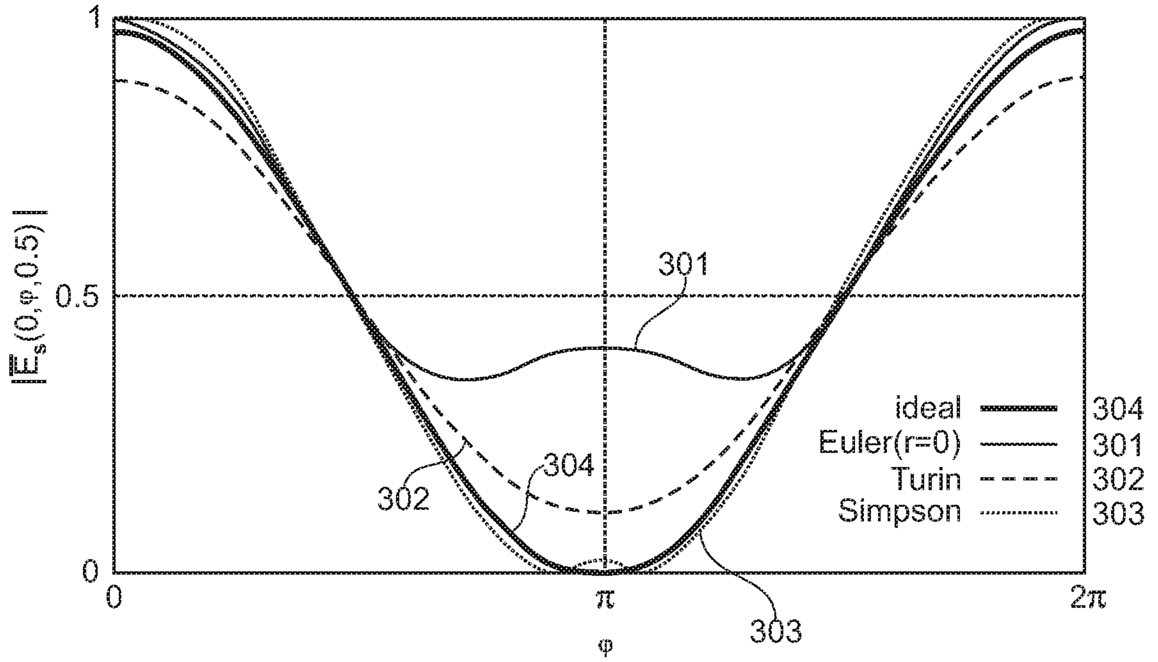
1
0.5
0
0
π
2π
φ
$|\overline{E}_s(0,\varphi,0.5)|$
301
302
304
303
ideal 304
Euler(r=0) 301
Turin 302
Simpson 303

Fig. 3

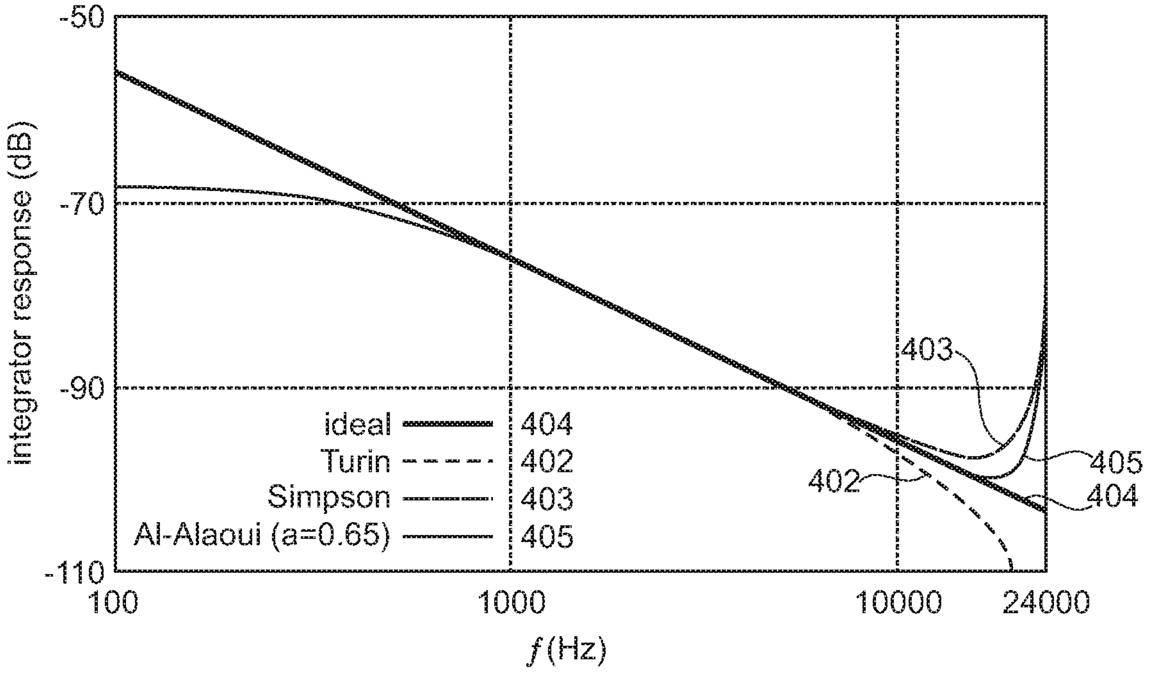
-50
-70
-90
-110
100
1000
10000
24000
f (Hz)
integrator response (dB)
403
405
402
404
ideal 404
Turin 402
Simpson 403
Al-Alaoui (a=0.65) 405

Fig. 4

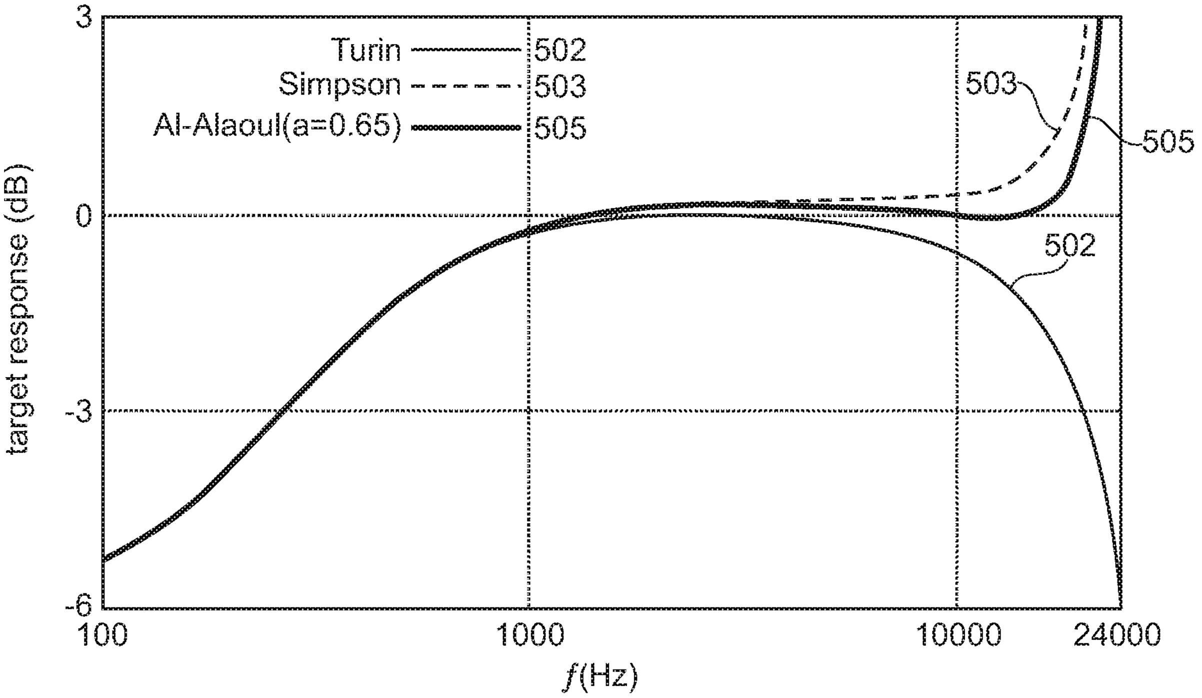
3
0
-3
-6
target response (dB)
Turin 502
Simpson 503
Al-Alaoul(a=0.65) 505
503
505
502
100
1000
10000
24000
f(Hz)

Fig. 5

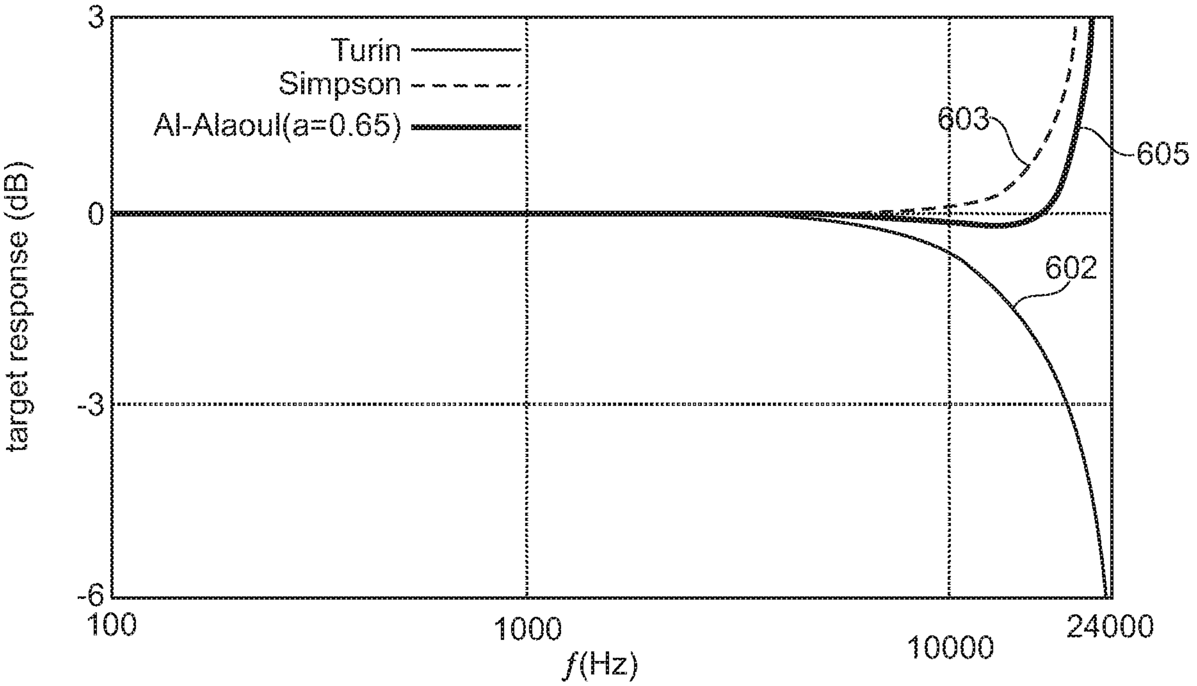
3
0
-3
-6
target response (dB)
Turin
Simpson
Al-Alaoul(a=0.65)
603
605
602
100
1000
10000
24000
f(Hz)

Fig. 6

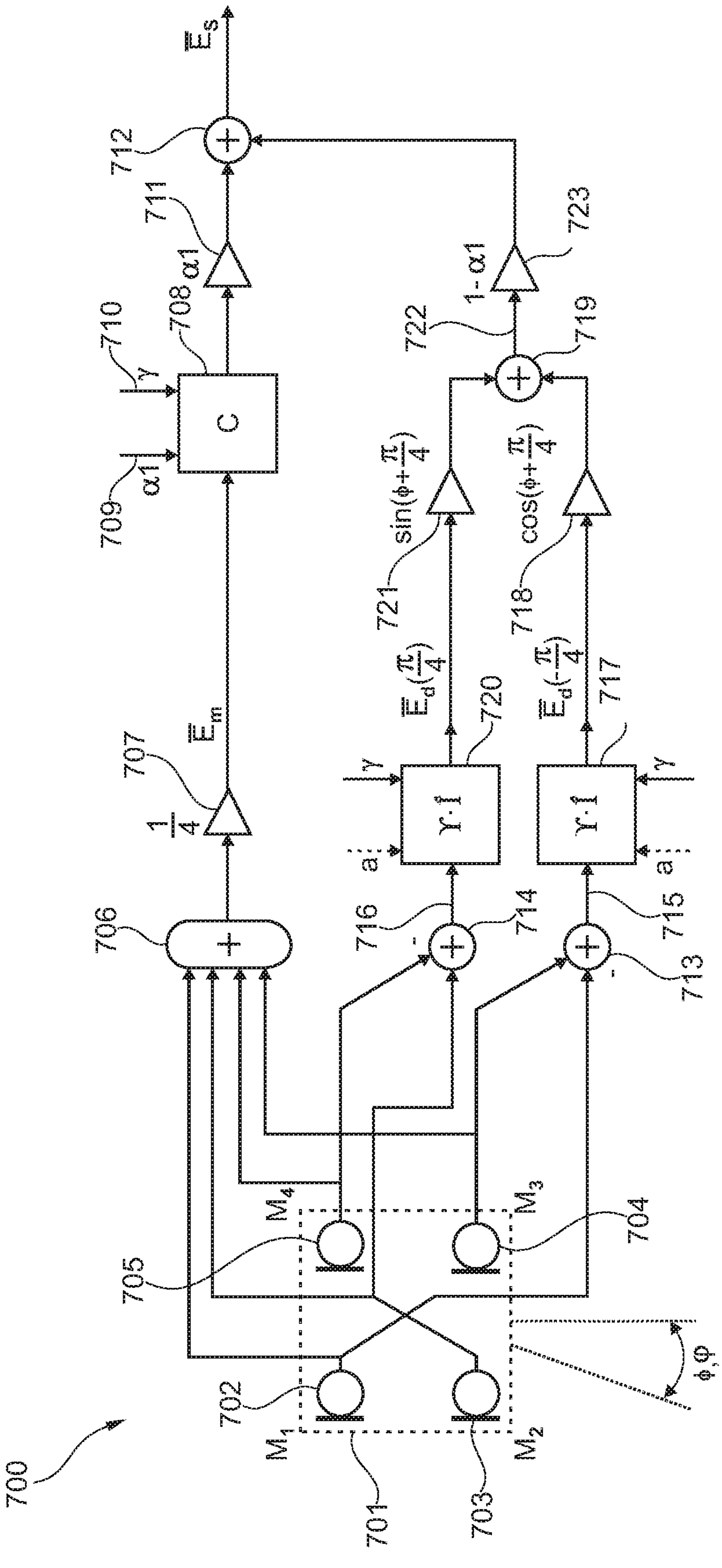
700
701
702
703
704
705
M1
M2
M3
M4
706
+
707
1/4
Ē_m
708
709
α1
710
γ
C
711
α1
712
+
Ē_s
713
714
715
716
-
-
a
a
γ
γ
γ·f
γ·f
717
718
719
720
721
722
723
1- α1
sin(φ+π/4)
cos(φ+π/4)
Ē_d(π/4)
Ē_d(-π/4)
φ,φ

Fig. 7