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(54) **Sound capture system**

Tonaufnahmesystem

Système de capture sonore

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EP 3 001 697 B1

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Description

TECHNICAL FIELD

[0001] The disclosure relates to a sound capture system, in particular to a sound capture system with a spherical microphone array for use in a modal beamforming system.

BACKGROUND

[0002] A microphone array-based modal beamforming system commonly comprises a spherical microphone array of a multiplicity of microphones equally distributed over the surface of a solid or virtual sphere for converting sounds into electrical audio signals and a modal beamformer combining the audio signals generated by the microphones to form an auditory scene representative of at least a portion of an acoustic sound field. This combination enables picking up acoustic signals dependent on their direction of propagation. As such, microphone arrays are also sometimes referred to as spatial filters. Spherical microphone arrays exhibit low- and high-frequency limitations, so that the sound field can only be accurately described over a limited frequency range. Low-frequency limitations essentially result when the directivity of the particular microphones of the array is inadequate in relation to the wave-length and the high amplification, which is necessary in this frequency range. This leads to a high amplification of (self) noise and thus to the need to limit the usable frequency range to a maximum lower frequency. High-frequency issues can be attributed to spatial aliasing effects. Similar to time aliasing, spatial aliasing occurs when a spatial function, e.g., the spherical harmonics, is under-sampled. For example, in order to distinguish 16 harmonics, at least 16 microphones are needed. In addition, the positions and, depending on the type of sphere used, the directivity of the microphones are important. A spatial aliasing frequency characterizes the upper critical frequency of the frequency range in which the spherical microphone array can be employed without generating any significant artifacts.

[0003] Two spherical microphone array configurations are commonly employed. The sphere may exist physically, or may merely be conceptual. In the first configuration, the microphones are arranged around a rigid sphere (e.g., made of wood, hard plastic or the like). In the second configuration, the microphones are arranged in free-field around an "open" sphere, referred to as an open-sphere configuration. Although the rigid-sphere configuration provides a more robust numerical formulation, the open-sphere configuration might be more desirable for use at low frequencies, where large spheres are realized. Such spherical arrays are disclosed, e.g., in US 2014/0270245 2. A1 and EP2747449 A1.

[0004] In open-sphere configurations most practical microphones have a drum-like or disc-like shape. In practice it is desirable to move the capsules closer to the

center of the array in order to maintain the directional performance of the array up to the highest audio frequencies. Thus, for microphones of a given size the gap between adjacent microphones will become smaller as they are pulled in towards the center, perhaps to the point at which adjacent microphones touch each other.

[0005] This situation worsens when directional microphones, i.e., microphones having an axis along which they exhibit maximum sensitivity, are employed, as directional microphones are commonly much bulkier than omnidirectional microphones, i.e., microphones having sensitivity independent of the direction. An exemplary type of directional microphone is called a shotgun microphone, which is also known as a line plus gradient microphone. Shotgun microphones may comprise an acoustic tube that, with its mechanical structure, reduces noises that arrive from directions other than directly in front of the microphone along the axis of the tube. Another exemplary directional microphone is a parabolic dish that concentrates the acoustic signal from one direction by reflecting away other noise sources coming from directions other than the desired direction. A sound capture system that avoids the dimensional problems noted above is desired.

SUMMARY

[0006] The sound capture system of the invention is defined in claim 1. The dependent claims define preferred embodiments of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0007] The system may be better understood with reference to the following drawings and description. The components in the figures are not necessarily to scale, emphasis instead being placed upon illustrating the principles of the invention. Moreover, in the figures like referenced numerals designate corresponding parts throughout the different views.

Figure 1 is a schematic diagram of an exemplary 13-microphone array for use in a sound capture system. Figure 2 is a block diagram of an exemplary evaluation circuit for the array shown in Figure 1. Figure 3 is a schematic diagram of an exemplary 12-microphone array for use in a sound capture system. Figure 4 is a block diagram of an exemplary evaluation circuit for the array shown in Figure 3. Figure 5 is a perspective view of a 6-element microphone array mounted in a rigid spherical sound-diffracting structure (rigid sphere). Figure 6 is a perspective view of a microphone array having a sound-diffracting structure with a hexahedron shape. Figure 7 is a perspective view of a rigid spherical sound-diffracting structure with indentations of the surface formed by conical cavities.

Figure 8 is a schematic of a first exemplary signal coupler applicable in the evaluation circuits shown in Figures 2 and 4.

Figure 9 is a schematic of a second exemplary signal coupler applicable in the evaluation circuits shown in Figures 2 and 4.

Figure 10 is a block diagram of an alternative evaluation circuit for the evaluation circuit shown in Figure 2.

Figure 11 is a schematic representation of an exemplary beamformer arrangement based on the microphone array with rigid inner sphere shown in Figure 3 and the evaluation circuit as shown in Figure 4.

Figure 12 is an amplitude frequency diagram that illustrates radial functions for a rigid sphere with omnidirectional microphones and for an open sphere with first order directional microphones.

DETAILED DESCRIPTION

[0008] A first exemplary sound capture system (e.g., for use in a modal beamformer system) is illustrated in Figures 1 and 2. Figure 1 is a schematic diagram of an array 100 of microphones. Array 100 includes n (e.g., $n = 6$) first omnidirectional microphones 103-108 that provide first output signals with an omnidirectional response pattern. Microphones 103-108 are disposed at different positions at a first equidistance d_1 from a point of symmetry 101. Array 100 further includes m (e.g., $m = 6$) second omnidirectional microphones 109-114 that provide second output signals with an omnidirectional response pattern. Microphones 109-114 are disposed at different positions at a second equidistance d_2 from the point of symmetry 101. First microphones 103-108 are arranged on an open sphere 115 in a basic hexahedron structure, and second microphones 109-114 are arranged on an open sphere 116 also in a basic hexahedron structure. The difference between the first equidistance d_1 and the point of symmetry 101 may be between 0.5cm and 1.5cm, e.g., 0.85cm. The difference between the second equidistance d_2 and the first equidistance d_1 may be between 9cm and 11cm, e.g., 10cm. As can readily be seen, the difference between the first equidistance d_1 and the point of symmetry 101 is smaller than the difference between the second equidistance d_2 and the first equidistance d_1 . A center omnidirectional microphone 102 provides a fourth output signal with an omnidirectional response and is disposed at the point of symmetry 101.

[0009] Referring to Figure 2, an evaluation circuit 200 for array 100 receives the first output signals from the first microphones 103-108 and the second output signals from the second microphones 109-114, and superimposes by way of signal couplers 220-225 the output signals of pairs of the second omnidirectional microphones to produce, in response thereto, six third output signals 207-212 with a directional response pattern. Each of the second omnidirectional microphones 109-114 forms a

pair of second microphones with another second omnidirectional microphone that is disposed in line with it and the point of symmetry. Signals with a directional response pattern are signals provided by a directional (e.g., unidirectional) microphone constellation and signals with an omnidirectional response pattern are signals provided by an omnidirectional microphone constellation. Evaluation circuit 200 may further receive the fourth output signal from the center microphone 102 and superimpose by way of signal couplers 214-219 the output signal of each first microphones 103-108 with the fourth output signal from the center microphone 102 for producing, in response thereto, six fifth output signals 201-206 with a directional response pattern. Alternatively, the center microphone 102 may also be used, in a similar way as done above in connection with the inner ring microphones 103-108, for the outer ring microphones 109-114 for the combination/formation of the desired virtual directional microphones. Also a combination (difference) between the outer ring microphones 109-114 with the inner ring microphones 103-108 is possible, for example, microphone 109 may be combined with microphone 103 and so on. Following this concept leads to a maximum logical combination of six different distances and hence 26 optimal frequency ranges which could further be beneficially combined.

[0010] A second exemplary sound capture system, which does not fall within the scope of the claimed invention, is illustrated in Figures 3 and 4. Figure 3 is a schematic diagram of an array 300 of microphones. Array 300 includes n (e.g., $n = 6$) first omnidirectional microphones 303-308 that provide first output signals with an omnidirectional response pattern. First microphones 303-308 are disposed at different positions at a first equidistance d_3 from a point of symmetry 301. Array 300 further includes m (e.g., $m = 6$) second omnidirectional microphones 309-314 that provide second output signals with an omnidirectional response pattern. Microphones 309-314 are disposed at different positions at a second equidistance d_4 from the point of symmetry 301. In the present example, first microphones 303-308 may be arranged on an open sphere or a rigid sphere 315 in a basic hexahedron structure, and second microphones 309-314 are arranged on an open sphere 316 also in a basic hexahedron structure. The diameter of the inner (rigid) sphere 315 may be 1.5cm or more so that the difference between the first equidistance d_3 and the point of symmetry 301 may be greater than 0.75cm. The difference between the second equidistance d_4 and the point of symmetry 301 may be between 9 and 11cm for example. Again, the difference between the first equidistance d_1 and the point of symmetry 101 is smaller than the difference between the second equidistance d_2 and the first equidistance d_1 . No center omnidirectional microphone is required here in contrast to the sound capture system illustrated above in connection with Figures 1 and 2.

[0011] Referring to Figure 4, an evaluation circuit 400 for array 300 with open sphere receives the first output

signals from first microphones 303-308 and the second output signals from second microphones 309-314, and superimposes by way of signal couplers 414-419 the first output signals from pairs of omnidirectional microphones to produce, in response thereto, six fifth output signals 401-406 with a directional response pattern. Each of the second omnidirectional microphones 309-314 forms a pair of second microphones with another second omnidirectional microphone that is disposed in line with it and the point of symmetry. Evaluation circuit 400 may further superimpose by way of signal couplers 420-425 the second output signals from pairs of second omnidirectional microphones for producing, in response thereto, six third output signals 407-412 with a directional response pattern. Each of the second omnidirectional microphones 309-314 forms a pair of second microphones with another second omnidirectional microphone that is disposed in line with it and the point of symmetry.

[0012] Microphones mounted on a solid sphere do not have to be directional and hence it is not necessary to use directional microphones on the inner (rigid) sphere 315. The way described above of forming virtual directional microphones from omnidirectional microphones disposed on an open sphere does not work with a solid body residing at a central line on opposite sides of a solid body, i.e., a rigid sphere. This means that when using a rigid sphere 315 signal couplers 414-419 in the system shown in Figure 4 can be omitted without substitution and instead microphones 303-308 directly provide signals 401-406. With the rigid sphere 315, an obstacle resides in-between two opposite microphones on the outer sphere, e.g., microphone 309 and 312, which may require combining microphone 309 with microphone 303 instead of microphone 312. Otherwise, due to the fact that the outer sphere is only used to cover the low spectral range of a modal beamformer, the diffraction of a small obstacle in the spectral range with larger wave lengths than the dimensions of the obstacle, i.e., the solid center sphere, may play a minor role in practice.

[0013] Figure 5 is a schematic illustration of a 6-element 3D microphone array 500, which is applicable instead of sphere 315 in the array shown in Figure 3, and which is mounted in a sound-diffracting structure provided by rigid sphere 501. Note that only three of the six microphone elements can be seen in Figure 5 (i.e., microphones 502, 503, and 504), while the remaining three microphone elements are hidden on the back side of the sphere 500. All six microphone elements are mounted on the surface of sphere 500 at points where an included regular octahedron's vertices would contact the spherical surface. Other shapes (structures) such as a hexahedron shape may be used as well. The individual microphones are omnidirectional microphones.

[0014] Figure 6 shows a perspective view of a 3D microphone array 600 having a hexahedron shape. Although not shown in the figures, microphone array 600 of Figure 6, has a plurality of individual omnidirectional microphones, analogous to first microphones 303-308 in

the array shown in Figure 3, distributed around and integrated into different rigid, triangular sections 601 of sphere 600, where the microphone elements are mounted onto the surface of each square section 601. Depending on the particular implementation, the microphones may be distributed uniformly or non-uniformly around the polyhedron, with each square section 601 having the same number of microphone elements or different square sections 601 having different numbers of microphone elements, including some square sections 601 having no microphone elements.

[0015] Figure 7 illustrates a 3D microphone array 700 having a rigid spherical sound-diffracting structure 701 with microphones 702 embedded in cavities whose dimensions and shapes are optimized to tailor to the directivity pattern. Figure 7 shows a circular conical cavity, however alternatively sectional cavities, inverse spherical caps, inverse circular paraboloids or any other appropriate shaped cavity may be used to form an indentation of the spherical surface. The cavity shape can be tailored and optimized to obtain the best compromise between directivity and low-pass filtering, which is achieved in the sound-diffracting structure 701 due to a combination of obstacle size and cavity design. A person of ordinary skill in the art will appreciate that there is a large variety of shapes of indentations that can be implemented.

[0016] Figure 8 illustrates in more detail a microphone pair 801 (such as first and second microphone pairs described above in connection with Figures 1-4) and a related signal coupler 802 (such as couplers 214-225 and 420-425 in Figures 2 and 4) in an exemplary sound capture system 800 such as the sound capture systems shown in Figures 2 and 4. Microphone pair 801 features two omnidirectional microphones 803 and 804 of a pair of microphones. Within the evaluation circuit 802, the outputs of omnidirectional microphones 803 and 804 are subtracted, one from the other, by e.g. a differential amplifier 805. Before this subtraction, the output of, e.g., omnidirectional microphone 804 is passed through a delay element 806 to delay the outputs of the two omnidirectional microphones 803 and 804 relative to each other. This element may be, for example, an allpass, a fractional delay filter or time delay circuit. The output of differential amplifier 805 is optionally passed through a filter 807 to compensate for frequency shifts introduced by delay element 806. For example, microphone 803 may be used as center microphone 102 in the system shown in Figure 1 and microphone 804 as any of the microphones 103-108 and vice versa. Referring to Figure 9, in omnidirectional microphone element 900, signals from two omnidirectional microphones 901 and 902 are delayed by a time delay at delay elements 903 and 904, respectively. The delayed signal from microphone 901 is subtracted from the undelayed signal from microphone 902 at subtractor 905 to form a forward-facing cardioid signal. Similarly, the delayed signal from microphone 902 is subtracted from the undelayed signal from microphone 901 at subtractor 906 to form a signal with a directional re-

sponse pattern (e.g. backward-facing cardioids). In order to reduce the number of delay elements, the evaluation circuit 200 for array 100 may be simplified so that only one delay element is required for evaluating the output signals of the first microphones 103-108 in connection with the center microphone 102 as shown in Figure 10. A delay element 1001 is connected downstream of the center microphone 102 and the signal couplers 214-219 are provided simply by subtractors 1002-1007.

[0017] A modal beamformer circuit 1100 that may receive and process third signals 201-212 and fifth signals 401-412 from the sound capture systems described above in connection with Figures 1, 2 and 3, 4 is shown in Figure 11. Modal beamformer circuit 1100 receives the third and fifth signals 201-212 and 401-412, transforms these signals 201-212 or 401-412 into the spherical harmonics, and steers the spherical harmonics.

[0018] An exemplary beamformer arrangement 1100 based on microphone array 300 with omnidirectional microphones 309-314 disposed on an open outer sphere 316 as shown in Figure 3 and based on evaluation circuit 400 for array 300 with rigid inner sphere 315 as shown in Figure 4 is illustrated in Figure 11. The $Q = 6$ signals output by microphones 309-314 are fed into evaluation circuit 400, i.e., into signal couplers 420-425 which output $Q = 6$ directional microphone output signals 407-412. Output signals 407-412 are fed into a matrixing module 1101 which supplies N spherical harmonics to a rotational module 1102. Rotational module 1102 generates M rotated spherical harmonics (modes) from the N spherical harmonics which are weighted (multiplied with frequency dependent weighting coefficients $C_1 \dots C_M$) in a modal weighting module 1103 and then summed up in a summing module 1104 to an outer sphere output signal. A signal processing chain similar or identical to the one described above (i.e., the chain including matrixing module 1101, rotational module 1102, modal weighting module 1103, and summing module 1104) includes a matrixing module 1105, rotational module 1106, modal weighting module 1107, and summing module 1108. An adder 1111 receives the output of summing module 1104 via a lowpass filter 1109 and the output of summing module 1108 via a highpass filter 1110, and outputs a specific directional signal 1112 of microphone array 300. As can be seen, the inner rigid sphere and the outer open sphere are used for different spectral ranges, which in combination allows for a broader spectral range of directional signal 1112.

[0019] Figure 12 illustrates radial functions $W_m(ka)$ up to the $M = 4$ th order based on a sphere radius $a = 0.09$ m for a rigid sphere with omnidirectional microphones (solid lines) and for an open sphere with first order directional microphones (dashed lines). It can be seen that the open sphere microphone array performs better at lower frequencies and the closed (rigid) sphere microphone array is better at higher frequencies.

[0020] While various embodiments of the invention have been described, it will be apparent to those of or-

dinary skill in the art that many more embodiments and implementations are possible within the scope of the invention. Accordingly, the invention is not to be restricted except in light of the attached claims.

Claims

1. A sound capture system comprising:

a first number of omnidirectional first microphones (103-108) that provide first output signals with an omnidirectional response pattern and that are disposed on a first open sphere (115) at different positions at a first equidistance (d_1) from a point of symmetry (101);
 a second number of omnidirectional second microphones (109-114) that provide second output signals with an omnidirectional response pattern and that are disposed on a second open sphere (116) at different positions at a second equidistance (d_2) from the point of symmetry (101);
 a center omnidirectional microphone (102) that provides a fourth output signal with an omnidirectional response and that is disposed at the point of symmetry (101);
 an evaluation circuit (200) that is configured to receive the first output signals and the second output signals, and to produce from the first output signals fifth output signals (201-206) with a directional response pattern and to produce from the second output signals third output signals (207-212) with a directional response pattern, wherein the evaluation circuit (200) is further configured to receive the fourth output signal from the center microphone (102) and to superimpose the output signal of each of the first microphones (103-108) with the fourth output signal from the center microphone (102) to produce, in response thereto, the fifth output signals (201-206) with a directional response pattern; and
 a modal beamformer arrangement (1101-1108) comprising a first signal chain (1101-1104), a second signal chain (1105-1108), a low-pass filter (1109), a high-pass filter (1110) and an adder (1111), the adder (1111) being coupled with the first signal chain (1101-1104) via the low-pass filter (1109) and with the second signal chain (1105-1108) via the high-pass filter (1110), the first signal chain (1101-1104) being supplied with the third output signals (207-212) and the second signal chain (1105-1108) being supplied with the fifth output signals (201-206); where
 the second number is a multiple of two;
 the first equidistance (d_1) is smaller than the second equidistance (d_2);
 each of the second microphones (109-114)

- forms with another of the second microphones (109-114) a pair of second microphones (109-114), the microphones of a pair of second microphones (109-114) being disposed in line with each other and the point of symmetry (101); and
the first signal chain (1101-1104) and the second signal chain (1105-1108) each comprise a matrixing modul (1101, 1105), a rotating module (1102, 1106), a modal weighting module (1103, 1107) and a summing module (1104, 1108).
2. The sound capture system of claim 1 where the difference between the first equidistance (d1) and the point of symmetry (101) is smaller than the difference between the second equidistance (d2) and the first equidistance (d1).
 3. The sound capture system of claim 2 where the difference between the first equidistance (d1) and the point of symmetry (101) is between 0.5cm and 1.5cm and the difference between the second equidistance (d2) and the first equidistance (d1) is between 9 cm and 11 cm.
 4. The sound capture system of claim 1 where the first number is a multiple of two; and each of the first microphones (103-108) forms with another of the first microphones a pair of first microphones (103-108), the microphones of a pair of first microphones (103-108) being disposed in line with each other and the point of symmetry (101).
 5. The sound capture system of claim 1 where the number of first microphones (103-108) and the number of second microphones (109-114) are identical.
 6. The sound capture system of claim 5 where the number of first microphones (103-108) and the number of second microphones (109-114) is six and the six first microphones (103-108) and the six second microphones (109-114) are disposed in a hexahedron structure.
 7. The sound capture system of claim 1 where the evaluation circuit (200) comprises:

at least one first delay path (1001) configured to receive the fourth output signal and to delay the fourth output signal to generate a delayed fourth output signal; and
first subtraction nodes (1002-1007) configured to receive the first output signals from the first microphones (103-108) and the delayed fourth output signal, and configured to generate the fifth output signals (201-206) based on differences between the first output signals and the

delayed fourth output signal.

8. The sound capture system of claim 1 where the evaluation circuit (200) further comprises:

second delay paths (806) configured to receive the first output signals and configured to delay the first output signals to generate delayed first output signals; and
second subtraction nodes (805) configured to receive the fourth output signal and the delayed first output signals and, and configured to generate the fifth output signals (201-206) based on differences between the fourth output signal and the delayed first output signals.

9. The sound capture system of claim 1 where the evaluation circuit (200) comprises:

third delay paths (903; 904) configured to receive the second output signals and to delay the second output signals to generate a delayed second output signals; and
third subtraction nodes (905; 906) configured each to receive the second output signal of one microphone of a pair of second microphones (109-114) and the delayed second output signal of the other microphone of the respective pair of second microphones (109-114), and configured to generate the third output signals (201-206) based on the difference between the second output signal of the one microphone of a pair of second microphones (109-114) and the delayed second output signal of the other microphone of the respective pair of second microphones (109-114).

Patentansprüche

1. Tonaufnahmesystem, umfassend:

eine erste Anzahl ungerichteter erster Mikrofone (103-108), die erste Ausgangssignale mit einem ungerichteten Ansprechmuster bereitstellen und die auf einer ersten offenen Kugel (115) an verschiedenen Stellen in einer ersten Äquidistanz (d1) von einem Symmetriepunkt (101) angeordnet sind;
eine zweite Anzahl ungerichteter zweiter Mikrofone (109-114), die zweite Ausgangssignale mit einem ungerichteten Ansprechmuster bereitstellen und die auf einer zweiten offenen Kugel (116) an verschiedenen Stellen in einer zweiten Äquidistanz (d2) von dem Symmetriepunkt (101) angeordnet sind;
ein mittiges ungerichtetes Mikrofon (102), das ein viertes Ausgangssignal mit einem ungerich-

- teten Ansprechen bereitstellt und das an dem Symmetriepunkt (101) angeordnet ist; einen Auswertestromkreis (200), der dazu konfiguriert ist, die ersten Ausgangssignale und die zweiten Ausgangssignale zu empfangen und aus den ersten Ausgangssignalen fünfte Ausgangssignale (201-206) mit einem gerichteten Ansprechmuster zu erzeugen und aus den zweiten Ausgangssignalen dritte Ausgangssignale (207-212) mit einem gerichteten Ansprechmuster zu erzeugen, wobei der Auswertestromkreis (200) ferner dazu konfiguriert ist, das vierte Ausgangssignal von dem mittigen Mikrofon (102) zu empfangen und das Ausgangssignal von jedem der ersten Mikrofone (103-108) mit dem vierten Ausgangssignal von dem mittigen Mikrofon (102) zu überlagern, um als Antwort darauf die fünften Ausgangssignale (201-206) mit einem gerichteten Ansprechmuster zu erzeugen; und eine modale Beamformer-Anordnung (1101-1108), die eine erste Signalkette (1101-1104), eine zweite Signalkette (1105-1108), einen Tiefpassfilter (1109), einen Hochpassfilter (1110) und einen Addierer (1111) umfasst, wobei der Addierer (1111) mit der ersten Signalkette (1101-1104) über den Tiefpassfilter (1109) und mit der zweiten Signalkette (1105-1108) über den Hochpassfilter (1110) gekoppelt ist, wobei die erste Signalkette (1101-1104) mit den dritten Ausgangssignalen (207-212) beliefert wird und die zweite Signalkette (1105-1108) mit den fünften Ausgangssignalen (201-206) beliefert wird; wobei die zweite Anzahl ein Vielfaches von zwei ist; die erste Äquidistanz (d_1) kleiner als die zweite Äquidistanz (d_2) ist; jedes der zweiten Mikrofone (109-114) mit einem anderen der zweiten Mikrofone (109-114) ein Paar zweiter Mikrofone (109-114) bildet, wobei die Mikrofone eines Paares zweiter Mikrofone (109-114) in einer Linie miteinander und mit dem Symmetriepunkt (101) angeordnet sind; und die erste Signalkette (1101-1104) und die zweite Signalkette (1105-1108) jeweils ein Matriziermodul (1101, 1105), ein rotierendes Modul (1102, 1106), ein modales Wichtungsmodule (1103, 1107) und ein Summiermodul (1104, 1108) umfassen.
2. Tonaufnahmesystem nach Anspruch 1, wobei die Differenz zwischen der ersten Äquidistanz (d_1) und dem Symmetriepunkt (101) kleiner ist als die Differenz zwischen der zweiten Äquidistanz (d_2) und der ersten Äquidistanz (d_1).
3. Tonaufnahmesystem nach Anspruch 2, wobei die Differenz zwischen der ersten Äquidistanz (d_1) und dem Symmetriepunkt (101) zwischen 0,5 cm und 1,5 cm beträgt und die Differenz zwischen der zweiten Äquidistanz (d_2) und der ersten Äquidistanz (d_1) zwischen 9 cm und 11 cm beträgt.
4. Tonaufnahmesystem nach Anspruch 1, wobei die erste Anzahl ein Vielfaches von zwei ist; und jedes der ersten Mikrofone (103-108) mit einem anderen der ersten Mikrofone ein Paar erster Mikrofone (103-108) bildet, wobei die Mikrofone eines Paares erster Mikrofone (103-108) in einer Linie miteinander und mit dem Symmetriepunkt (101) angeordnet sind.
5. Tonaufnahmesystem nach Anspruch 1, wobei die Anzahl erster Mikrofone (103-108) und die Anzahl zweiter Mikrofone (109-114) identisch sind.
6. Tonaufnahmesystem nach Anspruch 5, wobei die Anzahl erster Mikrofone (103-108) und die Anzahl zweiter Mikrofone (109-114) sechs ist und die sechs ersten Mikrofone (103-108) und die sechs zweiten Mikrofone (109-114) in einer Hexaederstruktur angeordnet sind.
7. Tonaufnahmesystem nach Anspruch 1, wobei der Auswertestromkreis (200) Folgendes umfasst:
- mindestens einen ersten Verzögerungsweg (1001), der dazu konfiguriert ist, das vierte Ausgangssignal zu empfangen und das vierte Ausgangssignal zu verzögern, um ein verzögertes viertes Ausgangssignal zu erzeugen; und erste Subtraktionsknoten (1002-1007), die dazu konfiguriert sind, die ersten Ausgangssignale von den ersten Mikrofonen (103-108) und das verzögerte vierte Ausgangssignal zu empfangen, und die dazu konfiguriert sind, die fünften Ausgangssignale (201-206) basierend auf Differenzen zwischen den ersten Ausgangssignalen und dem verzögerten vierten Ausgangssignal zu erzeugen.
8. Tonaufnahmesystem nach Anspruch 1, wobei der Auswertestromkreis (200) ferner Folgendes umfasst:
- zweite Verzögerungswege (806), die dazu konfiguriert sind, die ersten Ausgangssignale zu empfangen, und die dazu konfiguriert sind, die ersten Ausgangssignale zu verzögern, um verzögerte erste Ausgangssignale zu erzeugen; und zweite Subtraktionsknoten (805), die dazu konfiguriert sind, das vierte Ausgangssignal und die verzögerten ersten Ausgangssignale zu empfangen, und die dazu konfiguriert sind, die fünften Ausgangssignale (201-206) basierend auf Differenzen zwischen dem vierten Ausgangssignal und die verzögerten ersten Ausgangssig-

nalen zu erzeugen.

9. Tonaufnahmesystem nach Anspruch 1, wobei der Auswertestromkreis (200) Folgendes umfasst:

5 dritte Verzögerungswege (903; 904), die dazu konfiguriert sind, die zweiten Ausgangssignale zu empfangen und die zweiten Ausgangssignale zu verzögern, um verzögerte zweite Ausgangssignale zu erzeugen; und
10 dritte Subtraktionsknoten (905; 906), die jeweils dazu konfiguriert sind, das zweite Ausgangssignal eines Mikrofons eines Paares zweiter Mikrofone (109-114) und das verzögerte zweite Ausgangssignal des anderen Mikrofons des entsprechenden Paares zweiter Mikrofone (109-114) zu empfangen, und die dazu konfiguriert sind, die dritten Ausgangssignale (201-206) basierend auf der Differenz zwischen dem zweiten Ausgangssignal des einen Mikrofons eines Paares zweiter Mikrofone (109-114) und dem verzögerten zweiten Ausgangssignal des anderen Mikrofons des entsprechenden Paares zweiter Mikrofone (109-114) zu erzeugen.

Revendications

1. Système de capture de son comprenant :

30 un premier nombre de premiers microphones omnidirectionnels (103 à 108) qui fournissent des premiers signaux de sortie avec un schéma de réponse omnidirectionnelle et qui sont disposés sur une première sphère ouverte (115) à des positions différentes à une première équidistance (d1) d'un point de symétrie (101) ;
35 un second nombre de seconds microphones omnidirectionnels (109 à 114) qui fournissent des deuxièmes signaux de sortie avec un schéma de réponse omnidirectionnelle et qui sont disposés sur une seconde sphère ouverte (116) à des positions différentes à une seconde équidistance (d2) du point de symétrie (101) ;
40 un microphone omnidirectionnel central (102) qui fournit un quatrième signal de sortie avec une réponse omnidirectionnelle et qui est disposé au point de symétrie (101) ;
un circuit d'évaluation (200) qui est conçu pour recevoir les premiers signaux de sortie et les
45 deuxièmes signaux de sortie, et pour produire à partir des premiers signaux de sortie des cinquièmes signaux de sortie (201 à 206) avec un schéma de réponse directionnelle et pour produire à partir des deuxièmes signaux de sortie
50 des troisièmes signaux de sortie (207 à 212) avec un schéma de réponse directionnelle, dans lequel le circuit d'évaluation (200) est en outre

conçu pour recevoir le quatrième signal de sortie depuis le microphone central (102) et pour superposer le signal de sortie de chacun des premiers microphones (103 à 108) au quatrième signal de sortie depuis le microphone central (102) pour produire, en réponse à celui-ci, les cinquièmes signaux de sortie (201 à 206) avec un schéma de réponse directionnelle ; et
un agencement de formeur de faisceau modal (1101 à 1108) comprenant une première chaîne de signal (1101 à 1104), une seconde chaîne de signal (1105 à 1108), un filtre passe-bas (1109), un filtre passe-haut (1110), et un additionneur (1111), l'additionneur (1111) étant couplé avec la première chaîne de signal (1101 à 1104) via le filtre passe-bas (1109) et avec la seconde chaîne de signal (1105 à 1108) via le filtre passe-haut (1110), la première chaîne de signal (1101 à 1104) étant fournie avec les troisièmes signaux de sortie (207 à 212) et la seconde chaîne de signal (1105 à 1108) étant fournie avec les cinquièmes signaux de sortie (201 à 206) ; où

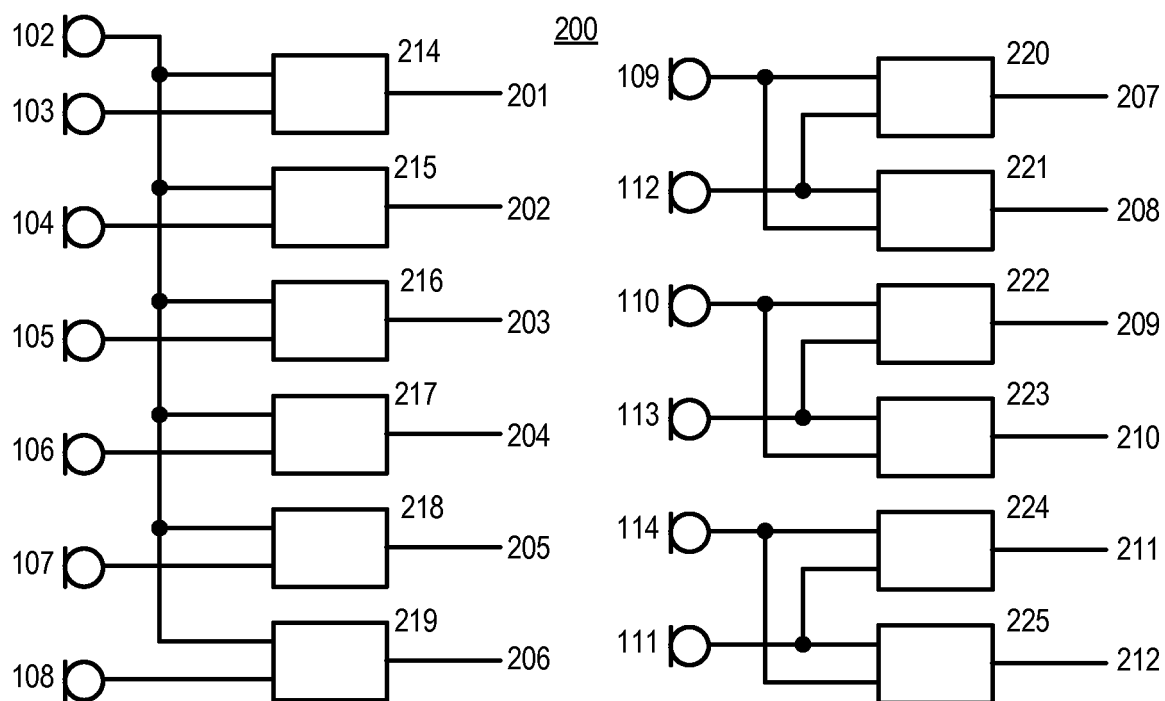
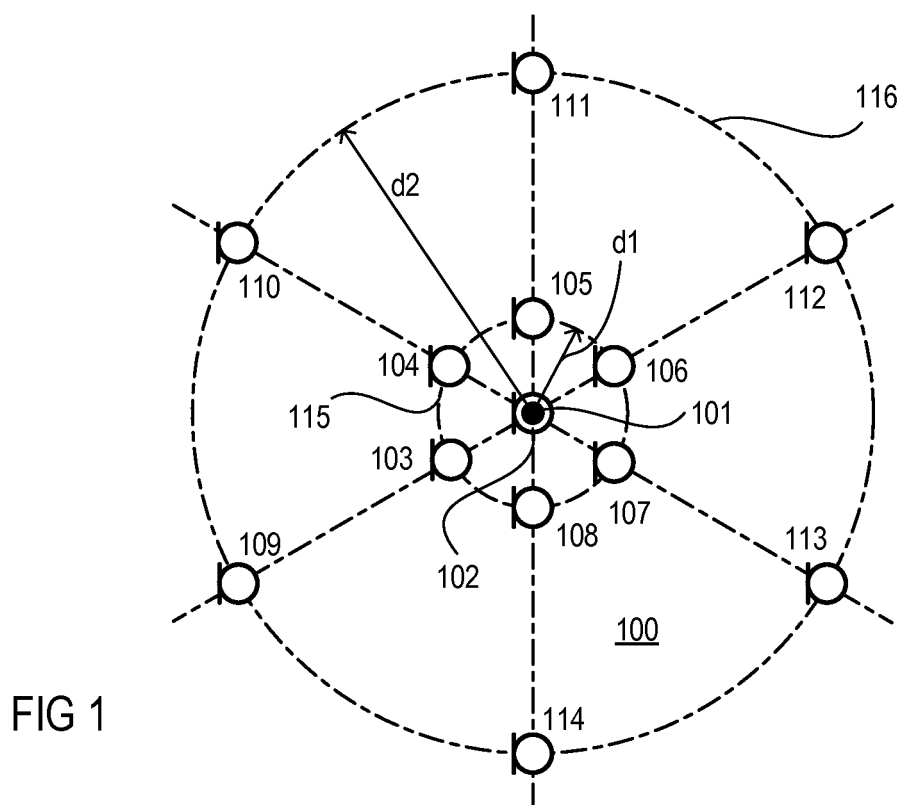
le second nombre est un multiple de deux ;
la première équidistance (d1) est plus petite que la seconde équidistance (d2) ;
chacun des seconds microphones (109 à 114) forme avec un autre des seconds microphones (109 à 114) une paire de seconds microphones (109 à 114), les microphones d'une paire de seconds microphones (109 à 114) étant disposés en ligne l'un avec l'autre et avec le point de symétrie (101) ; et
la première chaîne de signal (1101 à 1104) et la seconde chaîne de signal (1105 à 1108) comprennent chacune un module de matricage (1101, 1105), un module de rotation (1102, 1106), un module de pondération modale (1103, 1107) et un module de sommation (1104, 1108).

2. Système de capture de son selon la revendication 1, où la différence entre la première équidistance (d1) et le point de symétrie (101) est plus petite que la différence entre la seconde équidistance (d2) et la première équidistance (d1).
3. Système de capture de son selon la revendication 2, où la différence entre la première équidistance (d1) et le point de symétrie (101) est comprise entre 0,5 cm et 1,5 cm et la différence entre la seconde équidistance (d2) et la première équidistance (d1) est comprise entre 9 cm et 11 cm.
4. Système de capture de son selon la revendication 1, où le premier nombre est un multiple de deux ; et chacun des premiers microphones (103 à 108) forme avec un autre des premiers microphones une paire de premiers microphones (103 à 108), les micropho-

nes d'une paire de premiers microphones (103 à 108) étant disposés en ligne l'un avec l'autre et avec le point de symétrie (101).

5. Système de capture de son selon la revendication 1, où le nombre de premiers microphones (103 à 108) et le nombre de seconds microphones (109 à 114) sont identiques. 5
6. Système de capture de son selon la revendication 5, où le nombre de premiers microphones (103 à 108) et le nombre de seconds microphones (109 à 114) sont de six et les six premiers microphones (103 à 108) et les six seconds microphones (109 à 114) sont disposés en une structure d'hexaèdre. 10
15
7. Système de capture de son selon la revendication 1, où le circuit d'évaluation (200) comprend :
 - au moins une première voie de retard (1001) conçue pour recevoir le quatrième signal de sortie et pour retarder le quatrième signal de sortie afin de générer un quatrième signal de sortie retardé ; et 20
 - des premiers nœuds de soustraction (1002 à 1007) conçus pour recevoir les premiers signaux de sortie depuis les premiers microphones (103 à 108) et le quatrième signal de sortie retardé, et conçus pour générer les cinquièmes signaux de sortie (201 à 206) sur la base de différences entre les premiers signaux de sortie et le quatrième signal de sortie retardé. 25
30
8. Système de capture de son selon la revendication 1, où le circuit d'évaluation (200) comprend en outre : 35
 - des deuxième voies de retard (806) conçues pour recevoir les premiers signaux de sortie et conçues pour retarder les premiers signaux de sortie afin de générer les premiers signaux de sortie retardés ; et 40
 - des deuxième nœuds de soustraction (805) conçus pour recevoir le quatrième signal de sortie et les premiers signaux de sortie retardés, et conçus pour générer les cinquièmes signaux de sortie (201 à 206) sur la base de différences entre le quatrième signal de sortie et les premiers signaux de sortie retardés. 45
9. Système de capture de son selon la revendication 1, où le circuit d'évaluation (200) comprend : 50
 - des troisième voies de retard (903 ; 904) conçues pour recevoir les deuxième signaux de sortie et pour retarder les deuxième signaux de sortie afin de générer des deuxième signaux de sortie retardés ; et 55
 - des troisième nœuds de soustraction (905 ;

906) conçus chacun pour recevoir le deuxième signal de sortie d'un microphone d'une paire de seconds microphones (109 à 114) et le deuxième signal de sortie retardé de l'autre microphone de la paire respective de seconds microphones (109 à 114), et conçus pour générer les troisième signaux de sortie (201 à 206) sur la base de la différence entre le deuxième signal de sortie d'un microphone d'une paire de seconds microphones (109 à 114) et le deuxième signal de sortie retardé de l'autre microphone de la paire respective de seconds microphones (109 à 114).



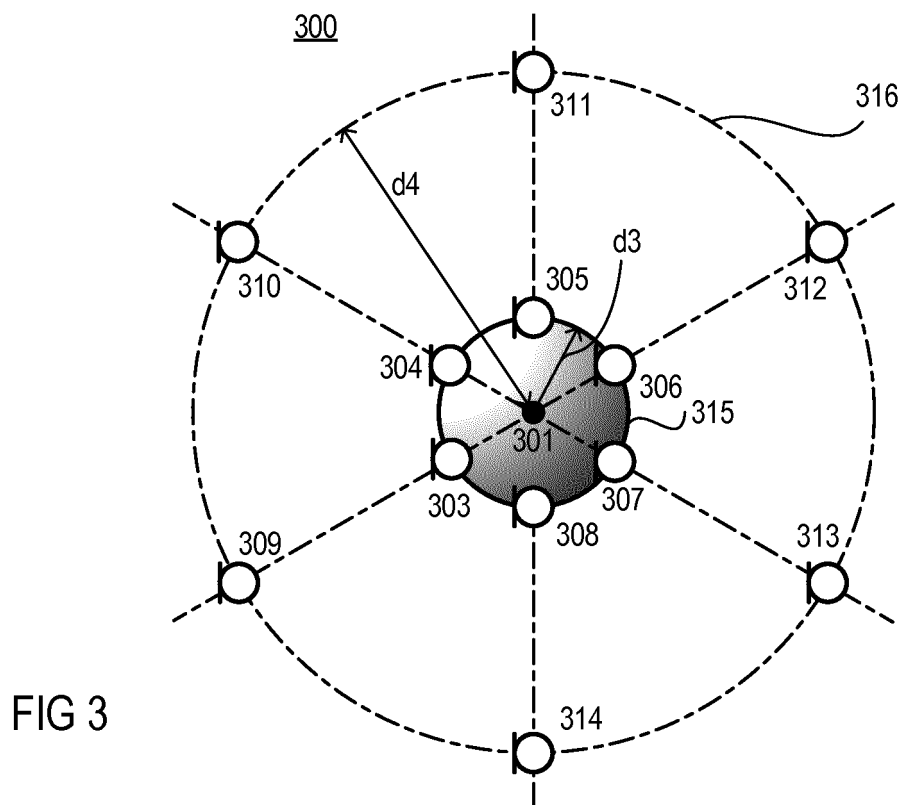


FIG 3

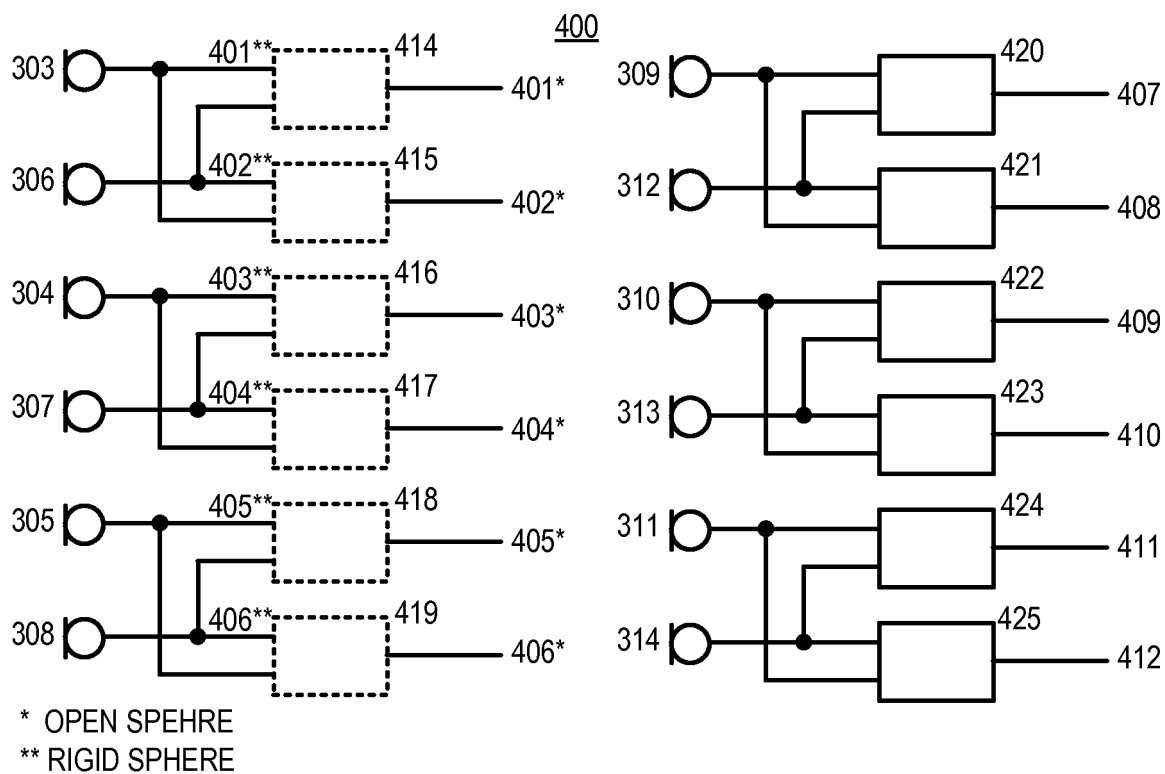


FIG 4

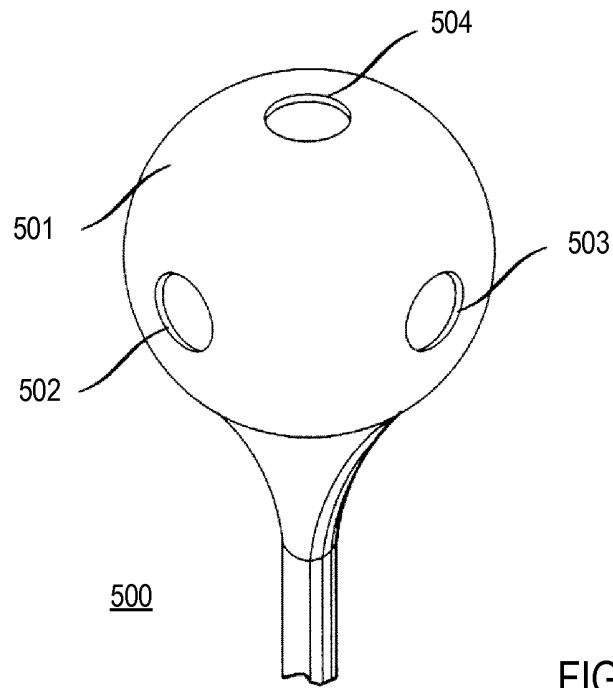


FIG 5

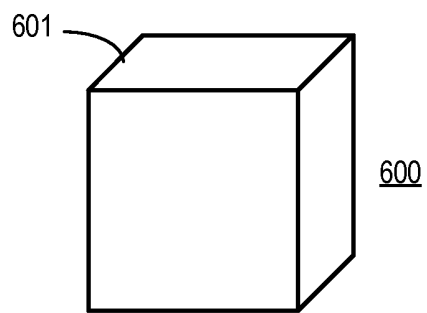


FIG 6

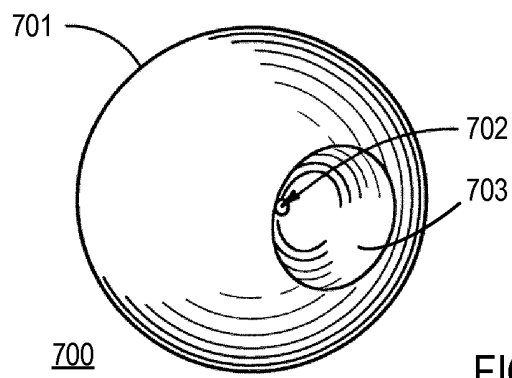


FIG 7

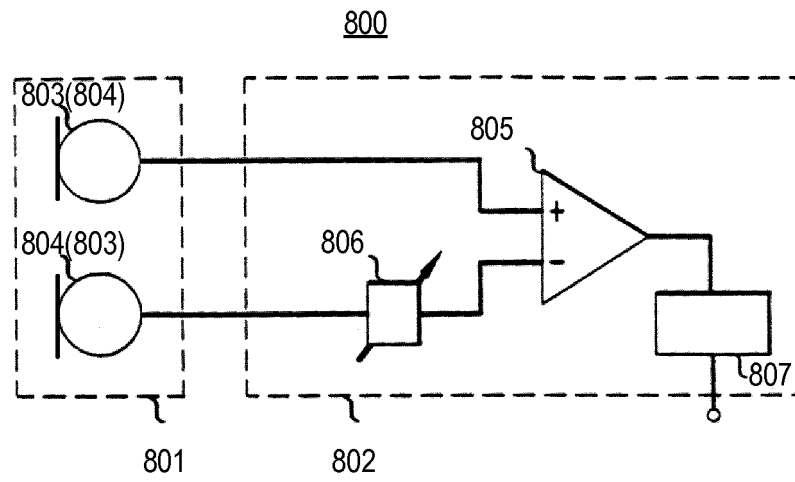


FIG 8

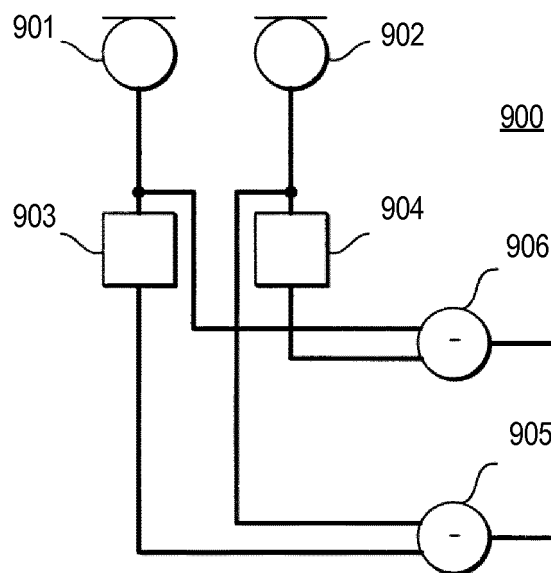


FIG 9

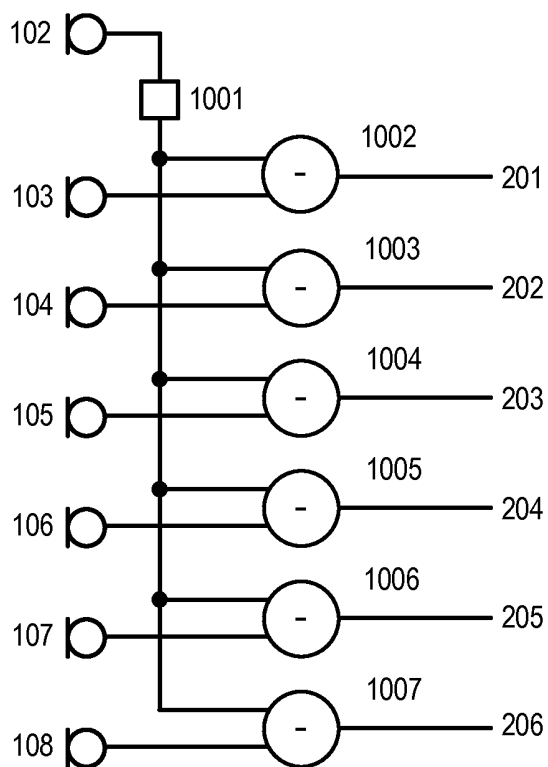


FIG 10

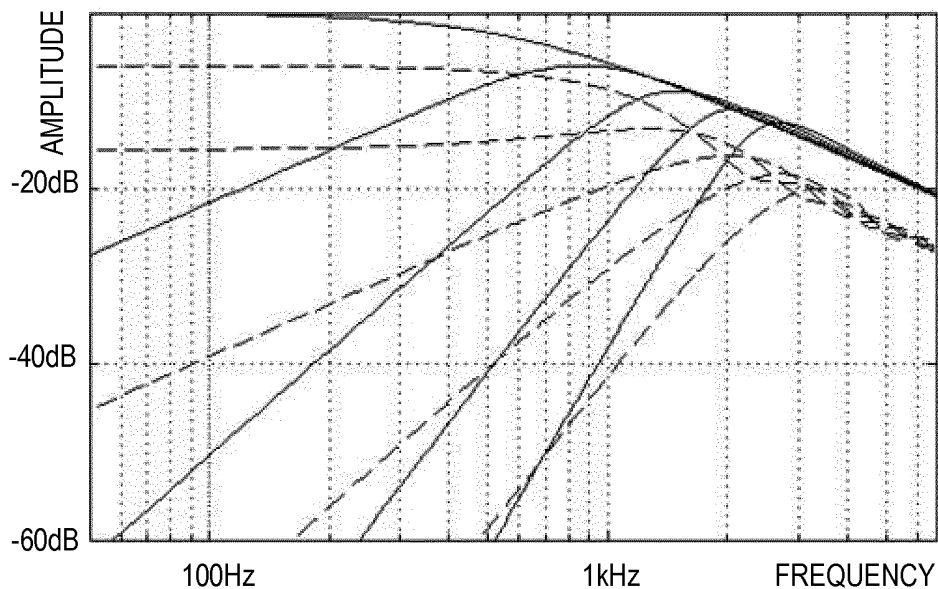


FIG 12

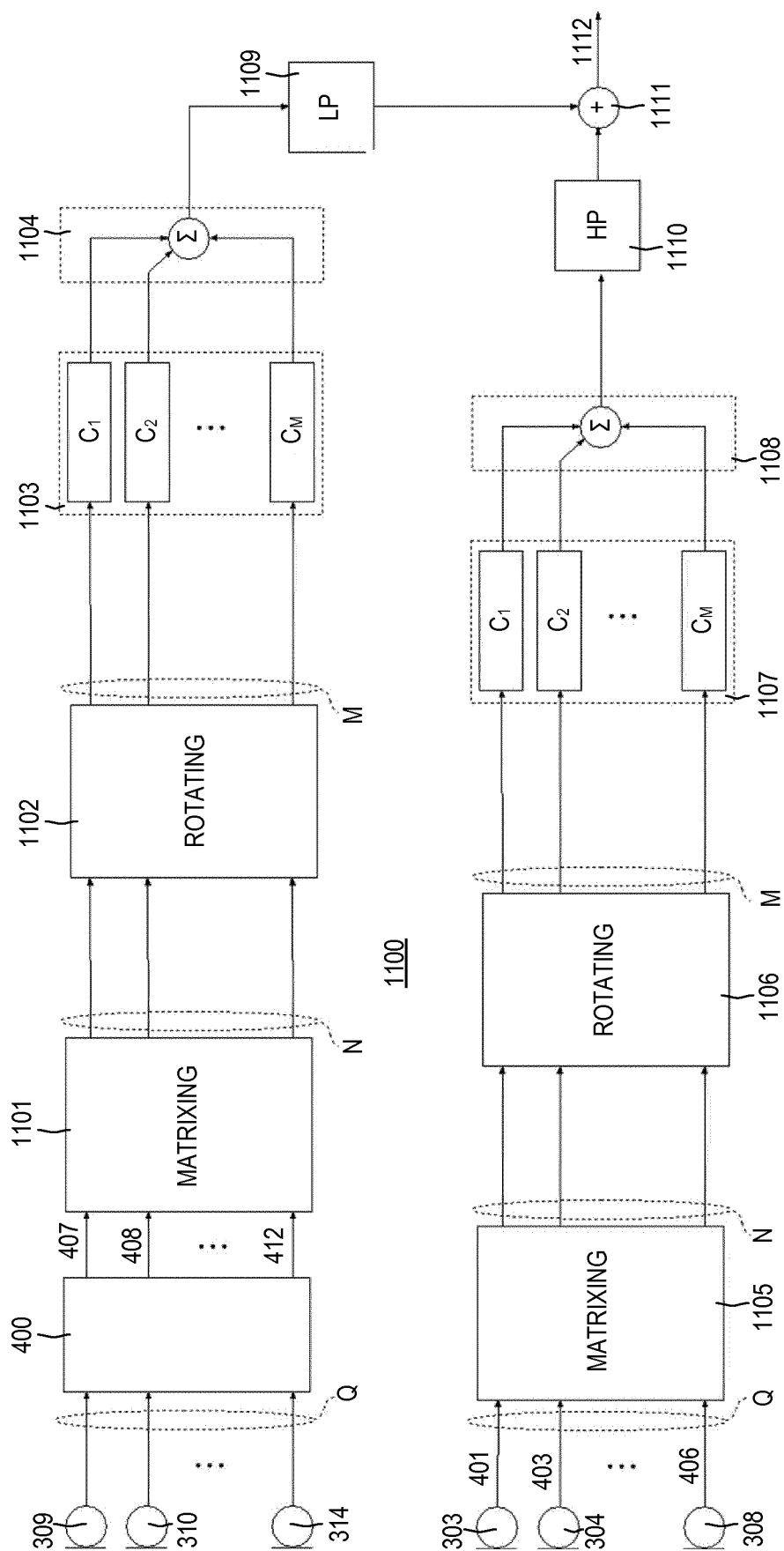


FIG 11

REFERENCES CITED IN THE DESCRIPTION

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