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NOTICE OF ALLOWANCE AND FEE(S) DUE

7590 08/13/2014

ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

EXAMINER	
ZHAO, XUEJUN	
ART UNIT	PAPER NUMBER
2653	

DATE MAILED: 08/13/2014

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
13/049,877	03/16/2011	Manli Zhu	CreativeTech_01NP_US	1547

TITLE OF INVENTION: Microphone Array System

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0	$0	$480	11/13/2014

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED. THIS NOTICE OF ALLOWANCE IS NOT A GRANT OF PATENT RIGHTS. THIS APPLICATION IS SUBJECT TO WITHDRAWAL FROM ISSUE AT THE INITIATIVE OF THE OFFICE OR UPON PETITION BY THE APPLICANT. SEE 37 CFR 1.313 AND MPEP 1308.

THE ISSUE FEE AND PUBLICATION FEE (IF REQUIRED) MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED. SEE 35 U.S.C. 151. THE ISSUE FEE DUE INDICATED ABOVE DOES NOT REFLECT A CREDIT FOR ANY PREVIOUSLY PAID ISSUE FEE IN THIS APPLICATION. IF AN ISSUE FEE HAS PREVIOUSLY BEEN PAID IN THIS APPLICATION (AS SHOWN ABOVE), THE RETURN OF PART B OF THIS FORM WILL BE CONSIDERED A REQUEST TO REAPPLY THE PREVIOUSLY PAID ISSUE FEE TOWARD THE ISSUE FEE NOW DUE.

HOW TO REPLY TO THIS NOTICE:

I. Review the ENTITY STATUS shown above. If the ENTITY STATUS is shown as SMALL or MICRO, verify whether entitlement to that entity status still applies.

If the ENTITY STATUS is the same as shown above, pay the TOTAL FEE(S) DUE shown above.

If the ENTITY STATUS is changed from that shown above, on PART B - FEE(S) TRANSMITTAL, complete section number 5 titled "Change in Entity Status (from status indicated above)".

For purposes of this notice, small entity fees are 1/2 the amount of undiscounted fees, and micro entity fees are 1/2 the amount of small entity fees.

II. PART B - FEE(S) TRANSMITTAL, or its equivalent, must be completed and returned to the United States Patent and Trademark Office (USPTO) with your ISSUE FEE and PUBLICATION FEE (if required). If you are charging the fee(s) to your deposit account, section "4b" of Part B - Fee(s) Transmittal should be completed and an extra copy of the form should be submitted. If an equivalent of Part B is filed, a request to reapply a previously paid issue fee must be clearly made, and delays in processing may occur due to the difficulty in recognizing the paper as an equivalent of Part B.

III. All communications regarding this application must give the application number. Please direct all communications prior to issuance to Mail Stop ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: Utility patents issuing on applications filed on or after Dec. 12, 1980 may require payment of maintenance fees. It is patentee's responsibility to ensure timely payment of maintenance fees when due.

IPR PETITION
US RE48,371
Sonos Ex. 1006

PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), to: **Mail** **Mail Stop ISSUE FEE**
Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313-1450
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INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE and PUBLICATION FEE (if required). Blocks 1 through 5 should be completed where appropriate. All further correspondence including the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

CURRENT CORRESPONDENCE ADDRESS (Note: Use Block 1 for any change of address)

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36 GREENLEIGH DRIVE
SEWELL, NJ 08080

Note: A certificate of mailing can only be used for domestic mailings of the Fee(s) Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing or transmission.

Certificate of Mailing or Transmission

I hereby certify that this Fee(s) Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Mail Stop ISSUE FEE address above, or being facsimile transmitted to the USPTO (571) 273-2885, on the date indicated below.

(Depositor's name)
(Signature)
(Date)

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nonprovisional	SMALL	$480	$0	$0	$480	11/13/2014

EXAMINER	ART UNIT	CLASS-SUBCLASS
ZHAO, XUEJUN	2653	381-300000

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363).
   - ❑ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.
   - ❑ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47; Rev 03-02 or more recent) attached. **Use of a Customer Number is required.**

2. For printing on the patent front page, list
   (1) The names of up to 3 registered patent attorneys or agents OR, alternatively,
   (2) The name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.
   1 ____________
   2 ____________
   3 ____________

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)

PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. If an assignee is identified below, the document has been filed for recordation as set forth in 37 CFR 3.11. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE

(B) RESIDENCE: (CITY and STATE OR COUNTRY)

Please check the appropriate assignee category or categories (will not be printed on the patent) : ❑ Individual ❑ Corporation or other private group entity ❑ Government

4a. The following fee(s) are submitted:
- ❑ Issue Fee
- ❑ Publication Fee (No small entity discount permitted)
- ❑ Advance Order - # of Copies ____________

4b. Payment of Fee(s): **(Please first reapply any previously paid issue fee shown above)**
- ❑ A check is enclosed.
- ❑ Payment by credit card. Form PTO-2038 is attached.
- ❑ The Director is hereby authorized to charge the required fee(s), any deficiency, or credits any overpayment, to Deposit Account Number ____________ (enclose an extra copy of this form).

5. **Change in Entity Status** (from status indicated above)
- ❑ Applicant certifying micro entity status. See 37 CFR 1.29 — NOTE: Absent a valid certification of Micro Entity Status (see forms PTO/SB/15A and 15B), issue fee payment in the micro entity amount will not be accepted at the risk of application abandonment.
- ❑ Applicant asserting small entity status. See 37 CFR 1.27 — NOTE: If the application was previously under micro entity status, checking this box will be taken to be a notification of loss of entitlement to micro entity status.
- ❑ Applicant changing to regular undiscounted fee status. — NOTE: Checking this box will be taken to be a notification of loss of entitlement to small or micro entity status, as applicable.

NOTE: This form must be signed in accordance with 37 CFR 1.31 and 1.33. See 37 CFR 1.4 for signature requirements and certifications.

Authorized Signature ____________ Date ____________

Typed or printed name ____________ Registration No. ____________

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Determination of Patent Term Adjustment under 35 U.S.C. 154 (b)

(Applications filed on or after May 29, 2000)

The Office has discontinued providing a Patent Term Adjustment (PTA) calculation with the Notice of Allowance.

Section 1(h)(2) of the AIA Technical Corrections Act amended 35 U.S.C. 154(b)(3)(B)(i) to eliminate the requirement that the Office provide a patent term adjustment determination with the notice of allowance. See Revisions to Patent Term Adjustment, 78 Fed. Reg. 19416, 19417 (Apr. 1, 2013). Therefore, the Office is no longer providing an initial patent term adjustment determination with the notice of allowance. The Office will continue to provide a patent term adjustment determination with the Issue Notification Letter that is mailed to applicant approximately three weeks prior to the issue date of the patent, and will include the patent term adjustment on the patent. Any request for reconsideration of the patent term adjustment determination (or reinstatement of patent term adjustment) should follow the process outlined in 37 CFR 1.705.

Any questions regarding the Patent Term Extension or Adjustment determination should be directed to the Office of Patent Legal Administration at (571)-272-7702. Questions relating to issue and publication fee payments should be directed to the Customer Service Center of the Office of Patent Publication at 1-(888)-786-0101 or (571)-272-4200.

OMB Clearance and PRA Burden Statement for PTOL-85 Part B

The Paperwork Reduction Act (PRA) of 1995 requires Federal agencies to obtain Office of Management and Budget approval before requesting most types of information from the public. When OMB approves an agency request to collect information from the public, OMB (i) provides a valid OMB Control Number and expiration date for the agency to display on the instrument that will be used to collect the information and (ii) requires the agency to inform the public about the OMB Control Number's legal significance in accordance with 5 CFR 1320.5(b).

The information collected by PTOL-85 Part B is required by 37 CFR 1.311. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, Virginia 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, Virginia 22313-1450. Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

Privacy Act Statement

The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether disclosure of these records is required by the Freedom of Information Act.
2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.
3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.
4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).
5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.
6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).
7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.
8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspection or an issued patent.
9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Notice of Allowability	Application No. 13/049,877	Applicant(s) ZHU ET AL.	
	Examiner EUGENE ZHAO	Art Unit 2653	AIA (First Inventor to File) Status No

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address--

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance (PTOL-85) or other appropriate communication will be mailed in due course. **THIS NOTICE OF ALLOWABILITY IS NOT A GRANT OF PATENT RIGHTS.** This application is subject to withdrawal from issue at the initiative of the Office or upon petition by the applicant. See 37 CFR 1.313 and MPEP 1308.

1. ☒ This communication is responsive to *Applicant's amendment dated 5/9/2014*.
   ☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on______.

2. ☐ An election was made by the applicant in response to a restriction requirement set forth during the interview on _____; the restriction requirement and election have been incorporated into this action.

3. ☒ The allowed claim(s) is/are *1, 2 and 5-23*. As a result of the allowed claim(s), you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see http://www.uspto.gov/patents/init_events/pph/index.jsp or send an inquiry to PPHfeedback@uspto.gov.

4. ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

   Certified copies:

   a) ☐ All b) ☐ Some *c) ☐ None of the:

   1. ☐ Certified copies of the priority documents have been received.
   2. ☐ Certified copies of the priority documents have been received in Application No. _____ .
   3. ☐ Copies of the certified copies of the priority documents have been received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

   * Certified copies not received: _____.

Applicant has THREE MONTHS FROM THE "MAILING DATE" of this communication to file a reply complying with the requirements noted below. Failure to timely comply will result in ABANDONMENT of this application.
THIS THREE-MONTH PERIOD IS NOT EXTENDABLE.

5. ☐ CORRECTED DRAWINGS (as "replacement sheets") must be submitted.

   ☐ including changes required by the attached Examiner's Amendment / Comment or in the Office action of Paper No./Mail Date _____.

   Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the drawings in the front (not the back) of each sheet. Replacement sheet(s) should be labeled as such in the header according to 37 CFR 1.121(d).

6. ☐ DEPOSIT OF and/or INFORMATION about the deposit of BIOLOGICAL MATERIAL must be submitted. Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Attachment(s)

1. ☒ Notice of References Cited (PTO-892)
2. ☐ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date _____
3. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
4. ☐ Interview Summary (PTO-413), Paper No./Mail Date _____ .
5. ☐ Examiner's Amendment/Comment
6. ☒ Examiner's Statement of Reasons for Allowance
7. ☐ Other _____.

/EUGENE ZHAO/
Examiner, Art Unit 2653

DETAILED ACTION

1. The present application is being examined under the pre-AIA first to invent provisions.

Reasons for Allowance

2. Claims 1, 2 and 5-23 are allowed.

3. The following is an examiner's statement of reasons for allowance:

The primary reference, Kim et al. (US 2009/0141907, hereinafter Kim) describes a method for canceling noise from a sound signal input through a microphone; Another close reference, Chol et al. (US 2004/0161121, hereinafter Chol), teaches an adaptive beamforming apparatus. However, neither Kim nor Chol teaches the method of improving sound quality by combining various units such as sound source localization unit, adaptive beamforming unit and noise reduction unit, where in the arbitrary configuration of the microphone array is determined by the delays, predefined angle, reference axis, azimuth angle, etc. So it is agreed that the prior art of record does not teach the amended independent claims.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled “Comments on Statement of Reasons for Allowance.”

Conclusion

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to EUGENE ZHAO whose telephone number is (571)270-1649. The examiner can normally be reached on 8:00AM-4:30PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Fan Tsang can be reached on (571)272-7547. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/EUGENE ZHAO/
Examiner, Art Unit 2653
Work phone: (571) 270-1649

/FAN TSANG/
Supervisory Patent Examiner, Art Unit 2653

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re. application of:
Application No.: 13/049,877
Filed: 03/16/2011
Applicant: Manli Zhu
Title: Microphone Array System

Examiner: Zhao, Xuejun
Art Unit: 2653
Atty. Docket No.: CreativeTech_01NP_US

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Response to Non Final Office Action

Examiner Zhao:

In response to the non-final office action dated January 13, 2014, please amend the above-referenced application as follows:

Amendments to the Claims are listed on page 2 of this response.

Remarks begin on page 11 of this response.

Attachments:
1. Transmittal Form, PTO/SB/21;
2. Petition for 1 month time extension, Form PTO/SB/22;
3. Payment of the following fee:
   -$210 for 1 new independent claim in excess of 3;
   -$40 for 1 new claim in excess of 20;
   -$100 for 1 month time extension;

The Director is hereby authorized to charge any underpayment of fee or any other fee that may be required to deposit account # 503291.

Amendment to the Claims

Claim 1 (currently amended): A method for enhancing a target sound signal from a plurality of sound signals, comprising:

> providing a microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, wherein said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are in operative communication with said array of said sound sensors;
>
> receiving said sound signals from a plurality of disparate sound sources by said sound sensors, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;
>
> <u>determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a two dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors;</u>
>
> estimating a spatial location of said target sound signal from said received sound signals by said sound source localization unit;
>
> performing adaptive beamforming for steering a directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound

signal by said adaptive beamforming unit, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

suppressing said ambient noise signals by said noise reduction unit for further enhancing said target sound signal.

Claim 2 (previously presented): The method of claim 1, wherein said spatial location of said target sound signal from said target sound source is estimated using a steered response power-phase transform by said sound source localization unit.

Claims 3-4 (canceled).

Claim 5 (previously presented): The method of claim 1, wherein said adaptive beamforming comprises:

providing a fixed beamformer, a blocking matrix, and an adaptive filter in said adaptive beamforming unit;

steering said directivity pattern of said array of said sound sensors in said direction of said spatial location of said target sound signal from said target sound source by said fixed beamformer for enhancing said target sound signal, when said target sound source is in motion;

feeding said ambient noise signals to said adaptive filter by blocking said target sound signal received from said target sound source using said blocking matrix; and

adaptively filtering said ambient noise signals by said adaptive filter in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources.

Claim 6 (previously presented): The method of claim 5, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 7 (previously presented): The method of claim 5, wherein said adaptive filtering comprises sub-band adaptive filtering performed by said adaptive filter, wherein said sub-band adaptive filtering comprises:

providing an analysis filter bank, an adaptive filter matrix, and a synthesis filter bank in said adaptive filter;

splitting said enhanced target sound signal from said fixed beamformer and said ambient noise signals from said blocking matrix into a plurality of frequency sub-bands by said analysis filter bank;

adaptively filtering said ambient noise signals in each of said frequency sub-bands by said adaptive filter matrix in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources; and

synthesizing a full-band sound signal using said frequency sub-bands of said enhanced target sound signal by said synthesis filter bank.

Claim 8 (previously presented): The method of claim 5, wherein said adaptive beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said adaptive beamforming unit and adjusting a step size for said adaptive filtering in response to detecting one of said presence and said absence of said target sound signal in said sound signals received from said disparate sound sources.

Claim 9 (previously presented): The method of claim 1, wherein said noise reduction unit performs noise reduction by using one of a Wiener-filter based noise reduction algorithm, a spectral subtraction noise reduction algorithm, an auditory transform based noise reduction algorithm, and a model based noise reduction algorithm.

Claim 10 (previously presented): The method of claim 1, wherein said noise reduction unit performs noise reduction in a plurality of frequency sub-bands, wherein said frequency sub-bands are employed by an analysis filter bank of said adaptive beamforming unit for sub-band adaptive beamforming.

Claim 11 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

> an array of sound sensors positioned in an arbitrary configuration, wherein said sound sensors receive said sound signals from a plurality of disparate sound sources, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;
>
> a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by <u>determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a two dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors;</u>

an adaptive beamforming unit that steers directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals for further enhancing said target sound signal.

Claim 12 (previously presented): The system of claim 11, wherein said sound source localization unit estimates said spatial location of said target sound signal from said target sound source using a steered response power-phase transform.

Claim 13 (previously presented): The system of claim 11, wherein said adaptive beamforming unit comprises:

a fixed beamformer that steers said directivity pattern of said array of said sound sensors in said direction of said spatial location of said target sound signal from said target sound source for enhancing said target sound signal, when said target sound source is in motion;

a blocking matrix that feeds said ambient noise signals to an adaptive filter by blocking said target sound signal received from said target sound source; and

said adaptive filter that adaptively filters said ambient noise signals in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources.

Claim 14 (previously presented): The system of claim 13, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 15 (previously presented): The system of claim 13, wherein said adaptive filter comprises a set of sub-band adaptive filters comprising:

an analysis filter bank that splits said enhanced target sound signal from said fixed beamformer and said ambient noise signals from said blocking matrix into a plurality of frequency sub-bands;

an adaptive filter matrix that adaptively filters said ambient noise signals in each of said frequency sub-bands in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources; and

a synthesis filter bank that synthesizes a full-band sound signal using said frequency sub-bands of said enhanced target sound signal.

Claim 16 (previously presented): The system of claim 11, wherein said adaptive beamforming unit further comprises an adaptation control unit that detects said presence of said target sound signal and adjusts a step size for said adaptive filtering in response to detecting one of said presence and said absence of said target sound signal in said sound signals received from said disparate sound sources.

Claim 17 (previously presented): The system of claim 11, wherein said noise reduction unit is one of a Wiener-filter based noise reduction unit, a spectral subtraction noise reduction unit, an auditory transform based noise reduction unit, and a model based noise reduction unit.

Claim 18 (previously presented): The system of claim 11, further comprising one or more audio codecs that convert said sound signals in an analog form of said sound signals into digital sound signals and reconverts said digital sound signals into said analog form of said sound signals.

Claim 19 (previously presented): The system of claim 11, wherein said noise reduction unit performs noise reduction in a plurality of frequency sub-bands employed by an analysis filter bank of said adaptive beamforming unit for sub-band adaptive beamforming.

Claim 20 (previously presented): The system of claim 11, wherein said array of said sound sensors is one of a linear array of said sound sensors, a circular array of said sound sensors, and an arbitrarily distributed coplanar array of said sound sensors.

Claim 21 (new): The method of claim 1, wherein said delay (τ) is determined by a formula $\tau = f_s * t$, wherein f_s is a sampling frequency and t is a time delay.

Claim 22 (new): A method for enhancing a target sound signal from a plurality of sound signals, comprising:

providing a microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, wherein said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are in operative communication with said array of said sound sensors;

receiving said sound signals from a plurality of disparate sound sources by said sound sensors, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;

determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a first reference axis, an elevation angle between a second reference axis and said target sound signal, and an azimuth angle between said first reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a three dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors;

estimating a spatial location of said target sound signal from said received sound signals by said sound source localization unit;

performing adaptive beamforming for steering a directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal by said adaptive beamforming unit, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

suppressing said ambient noise signals by said noise reduction unit for further enhancing said target sound signal.

Claim 23 (new): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors positioned in an arbitrary configuration, wherein said sound sensors receive said sound signals from a plurality of disparate sound sources, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a first reference axis, an elevation angle between a second reference axis and said target sound signal, and an azimuth angle between said first reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a three dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors ;

an adaptive beamforming unit that steers directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals for further enhancing said target sound signal.

Remarks

The Present invention and the Pending Claims

The method and system disclosed herein, in general, relates to microphone array design. More particularly, the method and system disclosed herein relates to enhancing a target sound signal from a plurality of sound signals.

Claims 1-20 are pending from the original application. Claims 3-4 are now canceled. Claims 21-23 are newly added. Therefore Claims 1, 2, 3-23 are currently pending. Reconsideration and allowance of the pending claims is respectfully requested.

Summary of the Office Action

Claims 1-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kim et al. (US 2009/0141907, hereinafter Kim) in view of Choi et al (US 2004/0161121, hereinafter Choi).

Amendments to the Claims

Claims 1 and 11 are currently amended.

Claims 3-4 are canceled.

Claims 21-23 are newly added.

Support for the amendment: "*determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is* *in a two dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors*" in amended claims 1 and 11 is found in FIGS. 3-5, and paragraph [0009] of applicant's original specification.

Support for new claim 21 is found in paragraph [0063] of applicant's original specification.

Support for new claims 22-23 is found in FIGS. 7A-7C, and paragraph [0009] of applicant's original specification.

The office action states: "**Claims 1-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kim et al. (US 2009/0141907, hereinafter Kim) in view of Choi et al (US 2004/0161121, hereinafter Choi).**"

In response to the above rejection, applicant submits that Kim, in view of Chol, does not disclose all the limitations in applicant's amended claim 1.

Applicant discloses a microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, is provided. The sound source localization unit, the adaptive beamforming unit, and the noise reduction unit are in operative communication with the array of sound sensors. The array of sound sensors is, for example, a linear array of sound sensors, a circular array of sound sensors, or an arbitrarily distributed coplanar array of sound sensors. The array of sound sensors (microphone array) receive sound signals from multiple disparate sound sources. The method disclosed by the applicant can be applied on a microphone array with an arbitrary number of sound sensors having, for example, an arbitrary two dimensional (2D) configuration. The sound signals received by the sound sensors in the microphone array comprise the target sound signal from the target sound source among the disparate sound sources, and ambient noise signals. The sound source localization unit estimates a spatial location of the target sound signal from the received sound signals, for example, using a steered response power-phase transform. The adaptive beamforming unit performs adaptive beamforming for steering a directivity pattern of the microphone array in a direction of the spatial location of the target sound signal. The adaptive beamforming unit thereby enhances the target sound signal from the target sound source and partially suppresses the ambient noise signals. The noise reduction unit suppresses the ambient noise signals for further enhancing the target sound signal received from the target sound source (see FIGS. 2-5, and paragraphs [0007] – [0008] of applicant's original specification).

In the limitation "*determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a two dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors*" in amended claim 1, applicant discloses a method of **determining the delay (τ) between each of the sound sensors and the origin of the microphone array as a function of the distance between each of the sound sensors and the origin, a predefined angle between each of the sound sensors and a reference axis, and an azimuth angle between the reference axis and the target sound signal**. Further, in paragraph [0063] of his original application, applicant discloses that the **delay (τ) can be represented as the product of the sampling frequency (f_s) and the time delay (t). The formula disclosed is, $\tau = f_s * t$.** Therefore, the **distance between the sound sensors in the microphone array corresponds to the time used for the target sound signal to travel the distance and is measured by the number of samples within that time period**. Once the delay is determined, the microphone array can be aligned to enhance the target sound signal from a specific direction.

Kim discloses a method, medium and apparatus for canceling noise whereby it is possible to solve a conventional problem that unnecessary noise cannot be appropriately canceled from a sound obtained through a microphone array because of a small size of a digital sound obtaining apparatus having the microphone array and to overcome a conventional limitation that a target sound source signal cannot be accurately obtained due to the problem (see Kim Paragraph [0008]). A full search of Kim application **does not reveal** any teaching or disclosure of determining the delay (τ) between each of the sound sensors and the origin of the microphone array as a function of the distance between each of the sound sensors and the origin, a predefined angle between each of the sound sensors and a reference axis, and an azimuth angle between the reference axis and the target sound signal. Further, Kim **does not teach or disclose** a formula for determining the delay. Further, Kim **does not teach or disclose** that the distance between the sound sensors in the microphone array corresponds to the time used for the target sound signal to travel the distance. Furthermore, Kim **does not teach or disclose** measurement of delay (τ) by the number of samples within the time period corresponding to sampling frequency (f_s).

Chol discloses an adaptive beamforming apparatus and method including a fixed beamformer that compensates for time delays of M noise-containing speech signals input via a microphone array having M microphones (M is an integer greater than or equal to 2), and generates a sum signal of the M compensated noise-containing speech signals; and a multi-channel signal separator that extracts pure noise components from the M compensated noise-containing speech signals using M adaptive blocking filters that are connected to M adaptive canceling filters in a feedback structure and extracts pure speech components from the added signal using the M adaptive canceling filters that are connected to the M adaptive blocking filters in the feedback structure (see Chol abstract). A full search of Chol publication **does not reveal** any teaching or disclosure of determining the delay (τ) between each of the sound sensors and the origin of the microphone array as a function of the distance between each of the sound sensors and the origin, a predefined angle between each of the sound sensors and a reference axis, and an azimuth angle between the reference axis and the target sound signal. Further, Chol **does not disclose** a formula for determining the delay. Furthermore, Chol **does not teach or disclose** that the distance between the sound sensors in the microphone array corresponds to the time used for the target sound signal to travel the distance. Furthermore, Chol **does not teach or disclose** measurement of delay (τ) by the number of samples within the time period corresponding to sampling frequency (f_s).

Therefore, applicant submits that Kim, Chol, or a combination of Kim and Chol, does not teach or disclose the following limitation in amended claim 1:

> "*determining a delay between each of said sound sensors and an origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a two dimensional plane, wherein said delay is represented in terms of number of samples, and wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors;*"

Applicant therefore submits that amended claim 1 is novel and non-obvious over a combination of Kim and Chol, and respectfully requests that the rejection of claim 1 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Amended claim 11 is a system claim corresponding to the amended method claim 1 and has the same functionality. Since amended claim 1 is novel and non-obvious over a combination of Kim and Chol, amended claim 11 is also novel and non-obvious over a combination of Kim and Chol. Applicant therefore respectfully requests that the rejection of claim 11 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Claims 2, 5-10, and 21 are dependent on amended claim 1. Since amended claim 1 is novel and non-obvious over a combination of Kim and Chol, dependent claims 2, 5-

10, and 21 are also novel and non-obvious over a combination of Kim and Chol. Applicant therefore respectfully requests that the rejection of claims 2, 5-10, and 21 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Claims 12-20 are dependent on amended claim 11. Since amended claim 11 is novel and non-obvious over a combination of Kim and Chol, dependent claims 12-20 are also novel and non-obvious over a combination of Kim and Chol. Applicant therefore respectfully requests that the rejection of claims 12-20 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

New claims 22 and 23 are functionally the same as amended claims 1 and 11 respectively except that the target sound source that emits said target sound signal is in a three dimensional plane. Since amended claims 1 and 11 are novel and non-obvious over a combination of Kim and Chol, new claims 22 and 23 are also novel and non-obvious over a combination of Kim and Chol.

Conclusion

Applicant respectfully requests that a timely Notice of Allowance be issued in this case. In the interest of compact prosecution, if the prosecution of the application can be advanced or if a claim may be made potentially allowable by an Examiner's amendment, applicant requests Examiner Zhao to call the undersigned with the proposed amendment.

Respectfully submitted,

Date: May 09, 2014

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
13/049,877	03/16/2011	Manli Zhu	CreativeTech_01NP_US	1547

7590 01/13/2014

Ashok Tankha
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EXAMINER	
ZHAO, XUEJUN	
ART UNIT	**PAPER NUMBER**
2653	

MAIL DATE	DELIVERY MODE
01/13/2014	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 13/049,877	Applicant(s) ZHU ET AL.	
	Examiner EUGENE ZHAO	Art Unit 2653	AIA (First Inventor to File) Status No

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTHS FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *3/6/2011*.
☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ An election was made by the applicant in response to a restriction requirement set forth during the interview on _____; the restriction requirement and election have been incorporated into this action.

4)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims*

5)☒ Claim(s) *1-20* is/are pending in the application.
5a) Of the above claim(s) _____ is/are withdrawn from consideration.
6)☐ Claim(s) _____ is/are allowed.
7)☒ Claim(s) *1-20* is/are rejected.
8)☐ Claim(s) _____ is/are objected to.
9)☐ Claim(s) _____ are subject to restriction and/or election requirement.

* If any claims have been determined allowable, you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see http://www.uspto.gov/patents/init_events/pph/index.jsp or send an inquiry to PPHfeedback@uspto.gov.

Application Papers

10)☐ The specification is objected to by the Examiner.
11)☒ The drawing(s) filed on *4/18/2011* is/are: a)☒ accepted or b)☐ objected to by the Examiner.
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
Certified copies:
a)☐ All b)☐ Some** c)☐ None of the:
1.☐ Certified copies of the priority documents have been received.
2.☐ Certified copies of the priority documents have been received in Application No. _____.
3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

** See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☒ Information Disclosure Statement(s) (PTO/SB/08a and/or PTO/SB/08b) Paper No(s)/Mail Date _____.
3) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
4) ☐ Other: _____.

DETAILED ACTION

1. The present application is being examined under the pre-AIA first to invent provisions.

Information Disclosure Statement

2. The information disclosure statement (IDS) was submitted on March 16, 2011, and is in compliance with the provisions of 37 CFR 1.97. Accordingly, the information disclosure statement is being considered by the examiner.

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kim et al. (US 2009/0141907, hereinafter Kim) in view of Chol et al. (US 2004/0161121, hereinafter Chol).

Regarding claim 11, Kim teaches

A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors *(figure 6, reference 600 of Kim)* positioned in an arbitrary configuration, wherein said sound sensors receive said sound signals from a plurality of disparate sound sources, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals *(figure 2, reference 210 of Kim)*;

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals *(figure 6, reference 640 of Kim)*;

an adaptive beamforming unit that steers directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals *(figure 6, references 621 and 622 of Kim)*; and

a noise reduction unit that suppresses said ambient noise signals for further enhancing said target sound signal *(figure 6, reference 630 of Kim)*.

However, though Kim teaches a detailed method and apparatus for canceling noise from a sound signal input through microphones, it doesn't specifically teach that the sound source localization unit is directly connected to the microphone array, which is an obvious choice for the designers.

Chol teaches an adaptive beamforming apparatus and method that includes a fixed beamformer that compensates for time delays of M noise-containing speech signals input via a microphone array having M microphones. Specifically, Chol teaches that the time delay estimator 413 obtains the correlation of the speech signals x.sub.1(k), x.sub.2(k) and x.sub.M(k) directly from the microphone array. Here, the time delay estimator 413 may calculate time delays of speech signals using various methods besides the calculation of the correlation *(figure 4, reference 413 as the localization unit; see paragraph 36 of Chol)*.

Therefore it would have been obvious for one of ordinary skill in the art at the time of invention to utilize the teachings as in Kim with the teachings as in Chol. The motivation for doing so would have been to directly locate the position of the sound source *(paragraph 5 of Chol).*

Regarding claim 12, Kim further teaches that the signal synthesis unit 500 includes a window function 510, a signal transformation unit 520, a synthesis unit 530, an inverse signal transformation unit 540, and a frame accumulation unit 550. While the generated high-frequency target signal is a signal of the time domain, the low-frequency target signal generated using a phase difference is a signal of the frequency domain. Thus, it is necessary to transform the high-frequency target signal into a signal of the frequency domain *(figure 5; paragraph 65 of Kim).*

Regarding claims 13-15, Kim further teaches that the high-frequency target signal generation unit 221 obtains a high-frequency target signal by canceling a noise signal from the filtered high-frequency signal using a beamforming method. As described previously, beamforming is used to amplify or extract a sound source signal, i.e., a target signal, radiated from a sound source located in a particular direction through a microphone array. To this end, a beam pattern formed through the microphone array and signal information input to each individual microphone of the microphone array are used. Various beamforming methods such as a fixed beamforming method or an adaptive beamforming method have been introduced to obtain the signal information, and various algorithms for extracting a target signal from an input signal using the beamforming methods have been developed *(paragraph 38 of Kim).*

Regarding claim 16, Kim further teaches that the microphone array 200 obtains sound source signals. A way to control the microphone array 200, e.g., the direction of a sound source or the magnitude of a sound source signal, can be designed variously according to a situation in which and a goal for which the current embodiment of the present invention is implemented *(paragraph 30 of Kim).*

Regarding claim 17, Kim further teaches that a method of recognizing the dominant signal characteristic may be executed by specifying a direction of a sound source having a large objective measurement value, such as a large signal to noise ratio (SNR), of a sound source signal, as a target sound source direction. For the measurement, various sound source position searching methods such as a time delay of arrival (TDOA) method, a beamforming method, and a high-resolution spectral analysis method have been introduced. Hereinafter, the sound source position searching methods will be described in brief *(paragraph 71 of Kim).*

Regarding claims 18-19, Kim further teaches that it is possible to accurately obtain a target sound source signal by minimizing signal distortion occurring in a low-frequency band in a digital sound obtaining apparatus having a small-size microphone array and accurately canceling or attenuating unnecessary noise *(paragraph 84 of Kim).*

Regarding claim 20, Kim further teaches that the direction detection unit 640 detects a direction of a sound source from which input signals obtained through a microphone array 600 are radiated. In order to obtain a direction of each of sound sources for sound source signals input from the sound sources, input directions of the sound source signals are detected using time delays between the input signals. In other words, the direction detection unit 640 searches for a sound source signal having a dominant signal characteristic that a gain or a sound pressure is large from neighboring scattered sound sources, in order to detect a direction of a corresponding sound source *(paragraph 71 of Kim).*

Regarding claims 1-10, method claims 1-10 are rejected for the same reasons as the apparatus claims previously discussed, since the recited elements would perform the claimed steps.

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Applicant is required under 37 C.F.R. § 1.111(c) to consider these references fully when responding to this action. It is noted that any citation to specific pages, columns, lines, or figures in the prior art references and any interpretation of the references should not be considered to be limiting in any way. A reference is relevant for all it contains and may be relied upon for all that it would have reasonably suggested to one having ordinary skill in the art. In re Heck, 699 F.2d 1331, 1332-33, 216 U.S.P.Q. 1038, 1039 (Fed. Cir. 1983) (quoting In re Lemelson, 397 F.2d 1006, 1009, 158 U.S.P.Q. 275, 277 (C.C.P.A. 1968)).

Amiri et al. (US 2003/0204397, hereinafter Amiri) teaches a beamformer for outputting an enhanced signal to a speech recognition system. Specifically, Amiri teaches a plurality (n) of microphone signals from array 1 are applied to the localization algorithm 3, which immediately begins to calculate the position of a person talking. The microphone signals are also fed into FIFO buffers 5, which introduce an equal delay to all channels before the signals are transmitted to the beamformer 7. The FIFO buffers 5 are preferably implemented in DSP software using a circular buffer in RAM. This well-known method requires that two pointers are provided: one points to the next input sample and the other points to the next output sample. DSP code manages the pointers to ensure that the pointers do not cross, thereby avoiding an overflow or underflow condition, as is well understood in the art. As discussed above, the delay conforms to the maximum amount of time needed by the localization algorithm 3 to find the position of the talker. Thus, the localized signal output from beamformer 7 is enhanced for application to a speech detection algorithm (not shown). As discussed above, this configuration should only be used when speech recognition is being applied to the handsfree telephone output (i.e. the output of beamformer 7). It is desirable that there should be no unnecessary added delay (i.e. the microphone signals should be routed directly to the beamformer 7) during normal (human) handsfree conversation. As discussed below, the FIFO delay can be reduced to zero during periods of silence *(figure 1, reference 3; paragraph 12 of Amiri).*

Any inquiry concerning this communication or earlier communications from the examiner should be directed to EUGENE ZHAO whose telephone number is (571)270-1649 **(Email: xuejun.zhao@uspto.gov).** The examiner can normally be reached on 8:00AM-4:30PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Fan Tsang can be reached on (571)272-7547. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/EUGENE ZHAO/
Examiner, Art Unit 2653
Work phone: (571) 270-1649
/FAN TSANG/
Supervisory Patent Examiner, Art Unit 2653

Notice of References Cited	Application/Control No. 13/049,877	Applicant(s)/Patent Under Reexamination ZHU ET AL.	
	Examiner EUGENE ZHAO	Art Unit 2653	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	Classification
*	A	US-2009/0141907	06-2009	Kim et al.	381/71.7
*	B	US-2004/0161121	08-2004	Chol et al.	381/092
*	C	US-2003/0204397	10-2003	Amiri et al.	704/231
	D	US-			
	E	US-			
	F	US-			
	G	US-			
	H	US-			
	I	US-			
	J	US-			
	K	US-			
	L	US-			
	M	US-			

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	Classification
	N					
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	P					
	Q					
	R					
	S					
	T					

NON-PATENT DOCUMENTS

*		Include as applicable: Author, Title Date, Publisher, Edition or Volume, Pertinent Pages)
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	X	

*A copy of this reference is not being furnished with this Office action. (See MPEP § 707.05(a).)
Dates in MM-YYYY format are publication dates. Classifications may be US or foreign.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99**)		
	Application Number	
	Filing Date	
	First Named Inventor	Manli Zhu
	Art Unit	
	Examiner Name	
	Attorney Docket Number	CreativeTech_01NP_US

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code1	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	7039199	B2	2006-05-02	Yong Rui	

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Examiner Initial*	Cite No	Publication Number	Kind Code1	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	20070076898	A1	2007-04-05	Bahaa Eddine Sarroukh et al.	
	2	20090279714	A1	2009-11-12	Hyun-Soo Kim et al.	
	3	20090304200	A1	2009-12-10	Hyun Soo Kim et al.	

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Examiner Initial*	Cite No	Foreign Document Number3	Country Code2 i	Kind Code4	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T5
	1	WO2008041878	RS	A2	2008-04-10	Zoran Sarik et al., Micronas NIT		☐

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /X.Z./

UTILITY PATENT APPLICATION TRANSMITTAL

(Only for new nonprovisional applications under 37 CFR 1.53(b))

Attorney Docket No.	CreativeTech_01NP_US
First Inventor	Manli Zhu
Title	Microphone Array System
Express Mail Label No.	

APPLICATION ELEMENTS

See MPEP chapter 600 concerning utility patent application contents.

ADDRESS TO: **Commissioner for Patents, P.O. Box 1450, Alexandria VA 22313-1450**

1. ☐ **Fee Transmittal Form** (e.g., PTO/SB/17)
2. ☑ **Applicant claims small entity status.** See 37 CFR 1.27.
3. ☑ **Specification** [*Total Pages* 43]
   Both the claims and abstract must start on a new page
   (For information on the preferred arrangement, see MPEP 608.01(a))
4. ☑ **Drawing(s)** (*35 U.S.C. 113*) [*Total Sheets* 33]
5. **Oath or Declaration** [*Total Sheets* 4]
   a. ☑ Newly executed (original or copy)
   b. ☐ A copy from a prior application (37 CFR 1.63(d))
   (for continuation/divisional with Box 18 completed)
   i. ☐ DELETION OF INVENTOR(S)
   Signed statement attached deleting inventor(s) name in the prior application, see 37 CFR 1.63(d)(2) and 1.33(b).
6. ☐ **Application Data Sheet.** See 37 CFR 1.76
7. ☐ **CD-ROM or CD-R** in duplicate, large table or Computer Program *(Appendix)*
   ☐ Landscape Table on CD
8. **Nucleotide and/or Amino Acid Sequence Submission**
   (if applicable, items a. – c. are required)
   a. ☐ Computer Readable Form (CRF)
   b. Specification Sequence Listing on:
   i. ☐ CD-ROM or CD-R (2 copies); or
   ii. ☐ Paper
   c. ☐ Statements verifying identity of above copies

ACCOMPANYING APPLICATION PARTS

9. ☐ **Assignment Papers** (cover sheet & document(s))
   Name of Assignee_______________
10. ☐ **37 CFR 3.73(b) Statement** *(when there is an assignee)* ☑ **Power of Attorney**
11. ☐ **English Translation Document** *(if applicable)*
12. ☑ **Information Disclosure Statement** (PTO/SB/08 or PTO-1449)
    ☐ Copies of citations attached
13. ☐ **Preliminary Amendment**
14. ☐ **Return Receipt Postcard** (MPEP 503) (*Should be specifically itemized)*
15. ☐ **Certified Copy of Priority Document(s)** *(if foreign priority is claimed)*
16. ☐ **Nonpublication Request** under 35 U.S.C. 122(b)(2)(B)(i). Applicant must attach form PTO/SB/35 or equivalent.
17. ☐ Other:_______________

18. If a CONTINUING APPLICATION, *check appropriate box, and supply the requisite information below and in the first sentence of the specification following the title, or in an Application Data Sheet under 37 CFR 1.76:*

☐ Continuation ☐ Divisional ☐ Continuation-in-part (CIP) of prior application No.: ....

Prior application information: *Examiner* _______________ *Art Unit:* _______________

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MICROPHONE ARRAY SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application claims the benefit of provisional patent application number 61/403,952 titled "Microphone array design and implementation for telecommunications and handheld devices", filed on September 24, 2010 in the United States Patent and Trademark Office.

[0002] The specification of the above referenced patent application is incorporated herein by reference in its entirety.

BACKGROUND

[0003] Microphones constitute an important element in today's speech acquisition devices. Currently, most of the hands-free speech acquisition devices, for example, mobile devices, lapels, headsets, etc., convert sound into electrical signals by using a microphone embedded within the speech acquisition device. However, the paradigm of a single microphone often does not work effectively because the microphone picks up many ambient noise signals in addition to the desired sound, specifically when the distance between a user and the microphone is more than a few inches. Therefore, there is a need for a microphone system that operates under a variety of different ambient noise conditions and that places fewer constraints on the user with respect to the microphone, thereby eliminating the need to wear the microphone or be in close proximity to the microphone.

[0004] To mitigate the drawbacks of the single microphone system, there is a need for a microphone array that achieves directional gain in a preferred spatial direction while suppressing ambient noise from other directions. Conventional microphone arrays include arrays that are typically developed for applications such as radar and sonar, but are generally not suitable for hands-free or handheld speech acquisition devices. The main reason is that the desired sound signal has an extremely wide bandwidth relative to its center frequency, thereby rendering conventional narrowband techniques employed in the conventional microphone arrays unsuitable. In order to cater to such broadband speech applications, the array size needs to be vastly increased, making the conventional microphone arrays large and bulky, and precluding the conventional microphone arrays from having broader applications, for example, in mobile and handheld communication devices. There is a need for a microphone array system that provides an effective response over a wide spectrum of frequencies while being unobtrusive in terms of size.

[0005] Hence, there is a long felt but unresolved need for a broadband microphone array and broadband beamforming system that enhances acoustics of a desired sound signal while suppressing ambient noise signals.

SUMMARY OF THE INVENTION

[0006] This summary is provided to introduce a selection of concepts in a simplified form that are further described in the detailed description of the invention. This summary is not intended to identify key or essential inventive concepts of the claimed subject matter, nor is it intended for determining the scope of the claimed subject matter.

[0007] The method and system disclosed herein addresses the above stated need for enhancing acoustics of a target sound signal received from a target sound source, while suppressing ambient noise signals. As used herein, the term "target sound signal" refers to a sound signal from a desired or target sound source, for example, a person's speech that needs to be enhanced. A microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, is provided. The sound source localization unit, the adaptive beamforming unit, and the noise reduction unit are in operative communication with the array of sound sensors. The array of sound sensors is, for example, a linear array of sound sensors, a circular array of sound sensors, or an arbitrarily distributed coplanar array of sound sensors. The array of sound sensors herein referred to as a "microphone array" receives sound signals from multiple disparate sound sources. The method disclosed herein can be applied on a microphone array with an arbitrary number of sound sensors having, for example, an arbitrary two dimensional (2D) configuration. The sound signals received by the sound sensors in the microphone array comprise the target sound signal from the target sound source among the disparate sound sources, and ambient noise signals.

[0008] The sound source localization unit estimates a spatial location of the target sound signal from the received sound signals, for example, using a steered response power-phase transform. The adaptive beamforming unit performs adaptive beamforming for steering a directivity pattern of the microphone array in a direction of the spatial location of the target sound signal. The adaptive beamforming unit thereby enhances the target sound signal from the target sound source and partially suppresses the ambient noise signals. The noise reduction unit suppresses the ambient noise signals for further enhancing the target sound signal received from the target sound source.

[0009] In an embodiment where the target sound source that emits the target sound signal is in a two dimensional plane, a delay between each of the sound sensors and an origin of the microphone array is determined as a function of distance between each of the sound sensors and the origin, a predefined angle between each of the sound sensors and a reference axis, and an azimuth angle between the reference axis and the target sound signal. In another embodiment where the target sound source that emits the target sound signal is in a three dimensional plane, the delay between each of the sound sensors and the origin of the microphone array is determined as a function of distance between each of the sound sensors and the origin, a predefined angle between each of the sound sensors and a first reference axis, an elevation angle between a second reference axis and the target sound signal, and an azimuth angle between the first reference axis and the target sound signal. This method of determining the delay enables beamforming for arbitrary numbers of sound sensors and multiple arbitrary microphone array configurations. The delay is determined, for example, in terms of number of samples.

Once the delay is determined, the microphone array can be aligned to enhance the target sound signal from a specific direction.

[0010] The adaptive beamforming unit comprises a fixed beamformer, a blocking matrix, and an adaptive filter. The fixed beamformer steers the directivity pattern of the microphone array in the direction of the spatial location of the target sound signal from the target sound source for enhancing the target sound signal, when the target sound source is in motion. The blocking matrix feeds the ambient noise signals to the adaptive filter by blocking the target sound signal from the target sound source. The adaptive filter adaptively filters the ambient noise signals in response to detecting the presence or absence of the target sound signal in the sound signals received from the disparate sound sources. The fixed beamformer performs fixed beamforming, for example, by filtering and summing output sound signals from the sound sensors.

[0011] In an embodiment, the adaptive filtering comprises sub-band adaptive filtering. The adaptive filter comprises an analysis filter bank, an adaptive filter matrix, and a synthesis filter bank. The analysis filter bank splits the enhanced target sound signal from the fixed beamformer and the ambient noise signals from the blocking matrix into multiple frequency sub-bands. The adaptive filter matrix adaptively filters the ambient noise signals in each of the frequency sub-bands in response to detecting the presence or absence of the target sound signal in the sound signals received from the disparate sound sources. The synthesis filter bank synthesizes a full-band sound signal using the frequency sub-bands of the enhanced target sound signal. In an embodiment, the adaptive beamforming unit further comprises an adaptation control unit for detecting the presence of the target sound signal and adjusting a step size for the adaptive filtering in response to detecting the presence or the absence of the target sound signal in the sound signals received from the disparate sound sources.

[0012] The noise reduction unit suppresses the ambient noise signals for further enhancing the target sound signal from the target sound source. The noise reduction unit performs noise reduction, for example, by using a Wiener-filter based noise reduction algorithm, a spectral subtraction noise reduction algorithm, an auditory transform based noise reduction algorithm, or a model based noise reduction algorithm. The noise reduction unit performs noise reduction in multiple frequency sub-bands employed for sub-band adaptive beamforming by the analysis filter bank of the adaptive beamforming unit.

[0013] The microphone array system disclosed herein comprising the microphone array with an arbitrary number of sound sensors positioned in arbitrary configurations can be implemented in handheld devices, for example, the iPad® of Apple Inc., the iPhone® of Apple Inc., smart phones, tablet computers, laptop computers, etc. The microphone array system disclosed herein can further be implemented in conference phones, video conferencing applications, or any device or equipment that needs better speech inputs.

BRIEF DESCRIPTION OF THE DRAWINGS

[0014] The foregoing summary, as well as the following detailed description of the invention, is better understood when read in conjunction with the appended drawings. For the purpose of illustrating the invention, exemplary constructions of the invention are shown in the drawings. However, the invention is not limited to the specific methods and instrumentalities disclosed herein.

[0015] FIG. **1** illustrates a method for enhancing a target sound signal from multiple sound signals.

[0016] FIG. **2** illustrates a system for enhancing a target sound signal from multiple sound signals.

[0017] FIG. **3** exemplarily illustrates a microphone array configuration showing a microphone array having N sound sensors arbitrarily distributed on a circle.

[0018] FIG. **4** exemplarily illustrates a graphical representation of a filter-and-sum beamforming algorithm for determining output of the microphone array having N sound sensors.

[0019] FIG. **5** exemplarily illustrates distances between an origin of the microphone array and sound sensor M_1 and sound sensor M_3 in the circular microphone array configuration, when the target sound signal is at an angle θ from the Y-axis.

[0020] FIG. **6A** exemplarily illustrates a table showing the distance between each sound sensor in a circular microphone array configuration from the origin of the microphone array, when the target sound source is in the same plane as that of the microphone array.

[0021] FIG. **6B** exemplarily illustrates a table showing the relationship of the position of each sound sensor in the circular microphone array configuration and its distance to the origin of the microphone array, when the target sound source is in the same plane as that of the microphone array.

[0022] FIG. **7A** exemplarily illustrates a graphical representation of a microphone array, when the target sound source is in a three dimensional plane.

[0023] FIG. **7B** exemplarily illustrates a table showing delay between each sound sensor in a circular microphone array configuration and the origin of the microphone array, when the target sound source is in a three dimensional plane.

[0024] FIG. **7C** exemplarily illustrates a three dimensional working space of the microphone array, where the target sound signal is incident at an elevation angle $\Psi < \Omega$.

[0025] FIG. **8** exemplarily illustrates a method for estimating a spatial location of the target sound signal from the target sound source by a sound source localization unit using a steered response power-phase transform.

[0026] FIG. **9A** exemplarily illustrates a graph showing the value of the steered response power-phase transform for every 10^{o}.

[0027] FIG. **9B** exemplarily illustrates a graph representing the estimated target sound signal from the target sound source.

[0028] FIG. **10** exemplarily illustrates a system for performing adaptive beamforming by an adaptive beamforming unit.

[0029] FIG. **11** exemplarily illustrates a system for sub-band adaptive filtering.

[0030] FIG. **12** exemplarily illustrates a graphical representation showing the performance of a perfect reconstruction filter bank.

[0031] FIG. **13** exemplarily illustrates a block diagram of a noise reduction unit that performs noise reduction using a Wiener-filter based noise reduction algorithm.

[0032] FIG. **14** exemplarily illustrates a hardware implementation of the microphone array system.

[0033] FIGS. **15A-15C** exemplarily illustrate a conference phone comprising an eight-sensor microphone array.

[0034] FIG. **16A** exemplarily illustrates a layout of an eight-sensor microphone array for a conference phone.

[0035] FIG. **16B** exemplarily illustrates a graphical representation of eight spatial regions to which the eight-sensor microphone array of FIG. **16A** responds.

[0036] FIGS. **16C**-**16D** exemplarily illustrate computer simulations showing the steering of the directivity patterns of the eight-sensor microphone array of FIG. **16A** in the directions of 15° and 60° respectively, in the frequency range 300 Hz to 5 kHz.

[0037] FIGS. **16E**-**16L** exemplarily illustrate graphical representations showing the directivity patterns of the eight-sensor microphone array of FIG. **16A** in each of the eight spatial regions, where each directivity pattern is an average response from 300Hz to 5000Hz.

[0038] FIG. **17A** exemplarily illustrates a graphical representation of four spatial regions to which a four-sensor microphone array for a wireless handheld device responds.

[0039] FIGS. **17B**-**17I** exemplarily illustrate computer simulations showing the directivity patterns of the four-sensor microphone array of FIG. **17A** with respect to azimuth and frequency.

[0040] FIGS. **18A-18B** exemplarily illustrate a microphone array configuration for a tablet computer.

[0041] FIG. **18C** exemplarily illustrates an acoustic beam formed using the microphone array configuration of FIGS. **18A-18B** according to the method and system disclosed herein.

[0042] FIGS. **18D**-**18G** exemplarily illustrate graphs showing processing results of the adaptive beamforming unit and the noise reduction unit for the microphone array configuration of FIG. **18B**, in both a time domain and a spectral domain for the tablet computer.

[0043] FIGS. **19A-19F** exemplarily illustrate tables showing different microphone array configurations and the corresponding values of delay τ_n for the sound sensors in each of the microphone array configurations.

DETAILED DESCRIPTION OF THE INVENTION

[0044] FIG. **1** illustrates a method for enhancing a target sound signal from multiple sound signals. As used herein, the term "target sound signal" refers to a desired sound signal from a desired or target sound source, for example, a person's speech that needs to be enhanced. The method disclosed herein provides **101** a microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit. The sound source localization unit, the adaptive beamforming unit, and the noise reduction unit are in operative communication with the array of sound sensors. The microphone array system disclosed herein employs the array of sound sensors positioned in an arbitrary configuration, the sound source localization unit, the adaptive beamforming unit, and the noise reduction unit for enhancing a target sound signal by acoustic beam forming in the direction of the target sound signal in the presence of ambient noise signals.

[0045] The array of sound sensors herein referred to as a "microphone array" comprises multiple or an arbitrary number of sound sensors, for example, microphones, operating in tandem. The microphone array refers to an array of an arbitrary number of sound sensors positioned in an arbitrary configuration. The sound sensors are transducers that detect sound and convert the sound into electrical signals. The sound sensors are, for example, condenser microphones, piezoelectric microphones, etc.

[0046] The sound sensors receive **102** sound signals from multiple disparate sound sources and directions. The target sound source that emits the target sound signal is one of the disparate sound sources. As used herein, the term "sound signals" refers to composite sound energy from multiple disparate sound sources in an environment of the microphone array. The sound signals comprise the target sound signal from the target sound source and the ambient noise signals. The sound sensors are positioned in an arbitrary planar configuration herein referred to as a "microphone array configuration", for example, a linear configuration, a circular configuration, any arbitrarily distributed coplanar array configuration, etc. By employing beamforming according to the method disclosed herein, the microphone array provides a higher response to the target sound signal received from a particular direction than to the sound signals from other directions. A plot of the response of the microphone array versus frequency and direction of arrival of the sound signals is referred to as a directivity pattern of the microphone array.

[0047] The sound source localization unit estimates **103** a spatial location of the target sound signal from the received sound signals. In an embodiment, the sound source localization unit estimates the spatial location of the target sound signal from the target sound source, for example, using a steered response power-phase transform as disclosed in the detailed description of FIG. **8**.

[0048] The adaptive beamforming unit performs adaptive beamforming **104** by steering the directivity pattern of the microphone array in a direction of the spatial location of the target sound signal, thereby enhancing the target sound signal, and partially suppressing the ambient noise signals. Beamforming refers to a signal processing technique used in the microphone array for directional signal reception, that is, spatial filtering. This spatial filtering is achieved by using adaptive or fixed methods. Spatial filtering refers to separating two signals with overlapping frequency content that originate from different spatial locations.

[0049] The noise reduction unit performs noise reduction by further suppressing **105** the ambient noise signals and thereby further enhancing the target sound signal. The noise reduction unit performs the noise reduction, for example, by using a Wiener-filter based noise reduction algorithm, a spectral subtraction noise reduction algorithm, an auditory transform based noise reduction algorithm, or a model based noise reduction algorithm.

[0050] FIG. **2** illustrates a system **200** for enhancing a target sound signal from multiple sound signals. The system **200**, herein referred to as a "microphone array system", comprises the array **201** of sound sensors positioned in an arbitrary configuration, the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207**.

[0051] The array **201** of sound sensors, herein referred to as the "microphone array" is in operative communication with the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207**. The microphone array **201** is, for example, a linear array of sound sensors, a circular array of sound sensors, or an arbitrarily distributed coplanar array of sound sensors. The microphone array **201** achieves directional gain in any preferred spatial direction and frequency band while suppressing signals from other spatial directions and frequency bands. The sound sensors receive the sound signals comprising the target sound signal and ambient noise signals from multiple disparate sound sources, where one of the disparate sound sources is the target sound source that emits the target sound signal.

[0052] The sound source localization unit **202** estimates the spatial location of the target sound signal from the received sound signals. In an embodiment, the sound source localization unit **202** uses, for example, a steered response power-phase transform, for estimating the spatial location of the target sound signal from the target sound source.

[0053] The adaptive beamforming unit **203** steers the directivity pattern of the microphone array **201** in a direction of the spatial location of the target sound signal, thereby enhancing the target sound signal and partially suppressing the ambient noise signals. The adaptive beamforming unit **203** comprises a fixed beamformer **204**, a blocking matrix **205**, and an adaptive filter **206** as disclosed in the detailed description of FIG. **10**. The fixed beamformer **204** performs fixed beamforming by filtering and summing output sound signals from each of the sound sensors in the microphone array **201** as disclosed in the detailed description of FIG. **4**. In an embodiment, the adaptive filter **206** is implemented as a set of sub-band adaptive filters. The adaptive filter **206** comprises an analysis filter bank **206a**, an adaptive filter matrix **206b**, and a synthesis filter bank **206c** as disclosed in the detailed description of FIG. **11**.

[0054] The noise reduction unit **207** further suppresses the ambient noise signals for further enhancing the target sound signal. The noise reduction unit **207** is, for example, a Wiener-filter based noise reduction unit, a spectral subtraction noise reduction unit, an auditory transform based noise reduction unit, or a model based noise reduction unit.

[0055] FIG. **3** exemplarily illustrates a microphone array configuration showing a microphone array **201** having N sound sensors **301** arbitrarily distributed on a circle **302** with a diameter "d", where "N" refers to the number of sound sensors **301** in the microphone array **201**. Consider an example where N = 4, that is, there are four sound sensors **301** M_0, M_1, M_2, and M_3 in the microphone array **201**. Each of the sound sensors **301** is positioned at an acute angle "Φ_n" from a Y-axis, where $\Phi_n \geq 0$ and n=0, 1, 2, …N-1. In an example, the sound sensor **301** M_0 is positioned at an acute angle Φ_0 from the Y-axis; the sound sensor **301** M_1 is positioned at an acute angle Φ_1 from the Y-axis; the sound sensor **301** M_2 is positioned at an acute angle Φ_2 from the Y-axis; and the sound sensor **301** M_3 is positioned at an acute angle Φ_3 from the Y-axis. A filter-and-sum beamforming algorithm determines the output "y" of the microphone array **201** having N sound sensors **301** as disclosed in the detailed description of FIG. **4**.

[0056] FIG. **4** exemplarily illustrates a graphical representation of the filter-and-sum beamforming algorithm for determining the output of the microphone array **201** having N sound sensors **301**. Consider an example where the target sound signal from the target sound source is at an angle θ with a normalized frequency ω. The microphone array configuration is arbitrary in a two dimensional plane, for example, a circular array configuration where the sound sensors **301** M_0, M_1, M_2,…, M_N, M_{N-1} of the microphone array **201** are arbitrarily positioned on a circle **302**. The sound signals received by each of the sound sensors **301** in the microphone array **201** are inputs to the microphone array **201**. The adaptive beamforming unit **203** employs the filter-and-sum beamforming algorithm that applies independent weights to each of the inputs to the microphone array **201** such that directivity pattern of the microphone array **201** is steered to the spatial location of the target sound signal as determined by the sound source localization unit **202**.

[0057] The output "y" of the microphone array **201** having N sound sensors **301** is the filter-and-sum of the outputs of the N sound sensors **301**. That is, $y = \sum_{n=0}^{N-1} w_n^T x_n$, where x_n is the output of the $(n+1)^{th}$ sound sensor **301**, and w_n^T denotes a transpose of a length-L filter applied to the $(n+1)^{th}$ sound sensor **301**.

[0058] The spatial directivity pattern $H(\omega, \theta)$ for the target sound signal from angle θ with normalized frequency ω is defined as:

$$H(\omega,\theta) = \frac{Y(\omega,\theta)}{\overline{X}(\omega,\theta)} = \frac{\sum_{n=0}^{N-1} W_n(\omega) X_n(\omega,\theta)}{\overline{X}(\omega,\theta)} \qquad (1)$$

where $\overline{X}$ is the signal received at the origin of the circular microphone array **201** and W is the frequency response of the real-valued finite impulse response (FIR) filter w. If the target sound source is far enough away from the microphone array **201**, the difference between the signal received by the $(n+1)^{th}$ sound sensor **301** "x_n" and the origin of the microphone array **201** is a delay τ_n; that is, $X_n(\omega,\tau) = \overline{X}(\omega,\theta) e^{-j\omega\tau_n}$.

[0059] FIG. **5** exemplarily illustrates distances between an origin of the microphone array **201** and the sound sensor **301** M_1 and the sound sensor **301** M_3 in the circular microphone array configuration, when the target sound signal is at an angle θ from the Y-axis. The microphone array system **200** disclosed herein can be used with an arbitrary directivity pattern for arbitrarily distributed sound sensors **301**. For any specific microphone array configuration, the parameter that is defined to achieve beamformer coefficients is the value of delay τ_n for each sound sensor **301**. To define the value of τ_n, an origin or a reference point of the microphone array **201** is defined; and then the distance d_n between each sound sensor **301** and the origin is measured, and then the angle Φ_n of each sound sensor **301** biased from a vertical axis is measured.

[0060] For example, the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_0 is $\Phi_{0,}$ the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_1 is $\Phi_{1,}$ the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_2 is $\Phi_{2,}$ and the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_3 is $\Phi_{3.}$ The distance between the origin O and the sound sensor **301** $M_{1,}$ and the origin O and the sound sensor **301** M_3 when the incoming target sound signal from the target sound source is at an angle θ from the Y-axis is denoted as τ_1 and τ_3, respectively.

[0061] For purposes of illustration, the detailed description refers to a circular microphone array configuration; however, the scope of the microphone array system **200** disclosed herein is not limited to the circular microphone array configuration but may be extended to include a linear array configuration, an arbitrarily distributed coplanar array configuration, or a microphone array configuration with any arbitrary geometry.

[0062] FIG. **6A** exemplarily illustrates a table showing the distance between each sound sensor **301** in a circular microphone array configuration from the origin of the microphone array **201**, when the target sound source is in the same plane as that of the microphone array **201**. The distance measured in meters and the corresponding delay (τ) measured in number of samples is exemplarily illustrated in FIG. **6A**. In an embodiment where the target sound source that emits the target sound signal is in a two dimensional plane, the delay (τ) between each of the sound sensors **301** and the origin of the microphone array **201** is determined as a function of distance (d) between each of the sound sensors **301** and the origin, a predefined angle (Φ) between each of the sound sensors **301** and a reference axis (Y) as exemplarily illustrated in FIG. **5**, and an azimuth angle (θ) between the reference axis (Y) and the target sound signal. The determined delay (τ) is represented in terms of number of samples.

[0063] If the target sound source is far enough from the microphone array **201**, the time delay between the signal received by the $(n+1)^{\text{th}}$ sound sensor **301** "x_n" and the origin of the microphone array **201** is herein denoted as "t" measured in seconds. The sound signals received by the microphone array **201**, which are in analog form are converted into digital sound signals by sampling the analog sound signals at a particular frequency, for example, 8000 Hz. That is, the number of samples in each second is 8000. The delay τ can be represented as the product of the sampling frequency (f_s) and the time delay (t). That is, $\tau = f_s * t$. Therefore, the distance between the sound sensors **301** in the microphone array **201** corresponds to the time used for the target sound signal to travel the distance and is measured by the number of samples within that time period.

[0064] Consider an example where "d" is the radius of the circle **302** of the circular microphone array configuration, "f_s" is the sampling frequency, and "c" is the speed of sound. FIG. **6B** exemplarily illustrates a table showing the relationship of the position of each sound sensor **301** in the circular microphone array configuration and its distance to the origin of the microphone array **201**, when the target sound source is in the same plane as that of the microphone array **201**. The distance measured in meters and the corresponding delay (τ) measured in number of samples is exemplarily illustrated in FIG. **6B**.

[0065] The method of determining the delay (τ) enables beamforming for arbitrary numbers of sound sensors **301** and multiple arbitrary microphone array configurations. Once the delay (τ) is determined, the microphone array **201** can be aligned to enhance the target sound signal from a specific direction.

[0066] Therefore, the spatial directivity pattern H can be re-written as:

$$H(\omega,\theta) = \sum_{n=0}^{N-1} W_n(\omega) e^{-j\omega\tau_n(\theta)} = \mathbf{w}^T \mathbf{g}(\omega,\theta) \qquad (2)$$

where $\mathbf{w}^T = [w_0^T, w_1^T, w_2^T, w_3^T, \ldots, w_{N-1}^T]$ and $\mathbf{g}(\omega,\theta) = \{g^i(\omega,\theta)\}_{i=1...NL} = \{e^{-j\omega(k+\tau_n(\theta))}\}_{i=1...NL}$ is the steering vector, i=1…NL, and k=mod(i-1,L) and n=floor((i-1)/L).

[0067] FIGS. **7A-7C** exemplarily illustrate an embodiment of a microphone array **201** when the target sound source is in a three dimensional plane. In an embodiment where the target sound source that emits the target sound signal is in a three dimensional plane, the delay (τ) between each of the sound sensors **301** and the origin of the microphone array **201** is determined as a function of distance (d) between each of the sound sensors **301** and the origin, a predefined angle (Φ) between each of the sound sensors **301** and a first reference axis (Y), an elevation angle (Ψ) between a second reference axis (Z) and the target sound signal, and an azimuth angle (θ) between the first reference axis (Y) and the target sound signal. The determined delay (τ) is represented in terms of number of samples. The determination of the delay enables beamforming for arbitrary numbers of the sound sensors **301** and multiple arbitrary configurations of the microphone array **201**.

[0068] Consider an example of a microphone array configuration with four sound sensors **301** M_0, M_1, M_2, and M_3. FIG. **7A** exemplarily illustrates a graphical representation of a microphone array **201**, when the target sound source in a three dimensional plane. As exemplarily illustrated in FIG. **7A**, the target sound signal from the target sound source is received from the direction (Ψ, θ) with reference to the origin of the microphone array **201**, where Ψ is the elevation angle and θ is the azimuth.

[0069] FIG. **7B** exemplarily illustrates a table showing delay between each sound sensor **301** in a circular microphone array configuration and the origin of the microphone array **201**, when the target sound source is in a three dimensional plane. The target sound source in a three dimensional plane emits a target sound signal from a spatial location (Ψ, θ). The distances between the origin O and the sound sensors **301** M_0, M_1, M_2, and M_3 when the incoming target sound signal from the target sound source is at an angle (Ψ, θ) from the Z-axis and the Y-axis respectively, are denoted as τ_0, τ_1, τ_2, and τ_3 respectively. When the spatial location of the target sound signal moves from the location $\Psi = 90^0$ to a location $\Psi = 0^0$, sin (Ψ) changes from 1 to 0, and as a result, the difference between each sound sensor **301** in the microphone array **201** becomes smaller and smaller. When $\Psi=0^0$, there is no difference between the sound sensors **301**, which implies that the target sound signal reaches each sound sensor **301** at the same time. Taking into account that the sample delay between the sound sensors **301** can only be an integer, the range where all the sound sensors **301** are identical is determined.

[0070] FIG. **7C** exemplarily illustrates a three dimensional working space of the microphone array **201** , where the target sound signal is incident at an elevation angle $\Psi < \Omega$, where Ω is a specific angle and Ψ is a variable representing the elevation angle. When the target sound signal is incident at an elevation angle $\Psi < \Omega$, all four sound sensors **301** M_0, M_1, M_2, and M_3 receive the same target sound signal for $0^0<\theta<360^0$. The delay τ is a function of both the elevation angle Ψ and the azimuth angle θ. That is, $\tau = \tau(\theta, \Psi)$. As used herein, Ω refers to the elevation angle such that all $\tau_i(\theta, \Omega)$ are equal to each other, where i=0, 1, 2, 3, etc. The value of Ω is determined by the sample delay between each of the sound sensors **301** and the origin of the microphone array **201**. The adaptive beamforming unit **203** enhances sound from this range and suppresses sound signals from other directions, for example, S_1 and S_2 treating them as ambient noise signals.

[0071] Consider a least mean square solution for beamforming according to the method disclosed herein. Let the spatial directivity pattern be 1 in the passband and 0 in the stopband. The least square cost function is defined as:

$$J(w)=\int_{\Omega_p}\int_{\Theta_p}|H(\omega,\theta)-1|^2\,d\omega d\theta+\alpha\int_{\Omega_s}\int_{\Theta_s}|H(\omega,\theta)|^2\,d\omega d\theta$$
$$=\int_{\Omega_p}\int_{\Theta_p}|H(\omega,\theta)|^2\,d\omega d\theta+\alpha\int_{\Omega_S}\int_{\Theta_X}|H(\omega,\theta)|^2\,d\omega d\theta-2\int_{\Omega_P}\int_{\Theta_P}\mathrm{Re}(H(\omega,\theta))d\omega d\theta+\int_{\Omega_P}\int_{\Theta_P}1\,d\omega d\theta$$

(3)

Replacing

$|H(\omega,\theta)|^2=w^T g(\omega,\theta)g^H(\omega,\theta)w=w^T G(\omega,\theta)w=w^T(G_R(\omega,\theta)+jG_I(\omega,\theta))w=w^T G_R(\omega,\theta)w$

and $\mathrm{Re}(H(\omega,\theta))=w^T g_R(\omega,\theta)$, $J(\omega)$ becomes

$$J(\omega) = w^T Q w - 2w^T a + d \text{, where}$$

$$Q = \int_{\Omega_P}\int_{\Theta_P} G_R(\omega,\theta)d\omega d\theta + \alpha\int_{\Omega_S}\int_{\Theta_S} G_R(\omega,\theta)d\omega d\theta$$

$$a = \int_{\Omega_P}\int_{\Theta_P} g_R(\omega,\theta)d\omega d\theta$$

$$d = \int_{\Omega_P}\int_{\Theta_P} 1\ d\omega d\theta \qquad (4)$$

where $g_R(\omega,\theta) = \cos[\omega(k+\tau_n)]$ and $G_R(\omega,\theta) = \cos[\omega(k-l+\tau_n-\tau_m)]$.

When $\partial J/\partial w = 0$, the cost function J is minimized. The least-square estimate of w is obtained by:

$$w = Q^{-1}a \qquad (5)$$

[0072] Applying linear constrains $Cw = b$, the spatial response is further constrained to a predefined value b at angle θ_f using following equation:

$$\begin{bmatrix} \mathbf{g}_R^T(\omega_{start},\theta_f) \\ \cdots \\ \mathbf{g}_R^T(\omega_{end},\theta_f) \end{bmatrix} w = \begin{bmatrix} b_{start} \\ \cdots \\ b_{end} \end{bmatrix} \qquad (6)$$

Now, the design problem becomes:

$$\min_w w^T Q w - 2w^T a + d \text{ subject to } Cw = b \qquad (7)$$

and the solution of the constrained minimization problem is equal to:

$$w = Q^{-1}C^T(CQ^{-1}C^T)^{-1}(b - CQ^{-1}a) + Q^{-1}a \qquad (8)$$

where w is the filter parameter for the designed adaptive beamforming unit **203**.

[0073] In an embodiment, the beamforming is performed by a delay-sum method. In another embodiment, the beamforming is performed by a filter-sum method.

[0074] FIG. **8** exemplarily illustrates a method for estimating a spatial location of the target sound signal from the target sound source by the sound source localization unit **202** using a steered response power-phase transform (SRP-PHAT). The SRP-PHAT combines the advantages of sound source localization methods, for example, the time difference of arrival (TDOA) method and the steered response power (SRP) method. The TDOA method performs the time delay estimation of the sound signals relative to a pair of spatially separated sound sensors **301**. The estimated time delay is a function of both the location of the target sound source and the position of each of the sound sensors **301** in the microphone array **201**. Because the position of each of the sound sensors **301** in the microphone array **201** is predefined, once the time delay is estimated, the location of the target sound source can be determined. In the SRP method, a filter-and-sum beamforming algorithm is applied to the microphone array **201** for sound signals in the direction of each of the disparate sound sources. The location of the target sound source corresponds to the direction in which the output of the filter-and-sum beamforming has the largest response power. The TDOA based localization is suitable under low to moderate reverberation conditions. The SRP method requires shorter analysis intervals and exhibits an elevated insensitivity to environmental conditions while not allowing for use under excessive multi-path. The SRP-PHAT method disclosed herein combines the advantages of the TDOA method and the SRP method, has a decreased sensitivity to noise and reverberations compared to the TDOA method, and provides more precise location estimates than existing localization methods.

[0075] For direction *i* ($0 \leq i \leq 360$), the delay D_{it} is calculated **801** between the t^{th} pair of the sound sensors **301** (t=1: all pairs). The correlation value $corr(D_{it})$ between the t^{th} pair of the sound sensors **301** corresponding to the delay of D_{it} is then calculated **802**. For the direction *i* ($0 \leq i \leq 360$), the correlation value is given **803** by:

$$CORR_i = \sum_{t=1}^{ALL\ PAIR} corr(D_{it})$$

Therefore, the spatial location of the target sound signal is given **804** by:

$$S = \underset{0 \leq i \leq 360}{\operatorname{argmax}} CORR_i$$
.

[0076] FIGS. **9A-9B** exemplarily illustrate graphs showing the results of sound source localization performed using the steered response power-phase transform (SRP-PHAT). FIG. **9A** exemplarily illustrates a graph showing the value of the SRP-PHAT for every $10^{o.}$ The maximum value corresponds to the location of the target sound signal from the target sound source. FIG. **9B** exemplarily illustrates a graph representing the estimated target sound signal from the target sound source and a ground truth.

[0077] FIG. **10** exemplarily illustrates a system for performing adaptive beamforming by the adaptive beamforming unit **203**. The algorithm for fixed beamforming is disclosed with reference to equations (3) through (8) in the detailed description of FIG. **4**, FIGS. **6A-6B**, and FIGS. **7A-7C**, which is extended herein to adaptive beamforming. Adaptive beamforming refers to a beamforming process where the directivity pattern of the microphone array **201** is adaptively steered in the direction of a target sound signal emitted by a target sound source in motion. Adaptive beamforming achieves better ambient noise suppression than fixed beamforming. This is because the target direction of arrival, which is assumed to be stable in fixed beamforming, changes with the movement of the target sound source. Moreover, the gains of the sound sensors **301** which are assumed uniform in fixed beamforming, exhibit significant distribution. All these factors reduce speech quality. On the other hand, adaptive beamforming adaptively performs beam steering and null steering; therefore, the adaptive beamforming method is more robust against steering error caused by the array imperfection mentioned above.

[0078] As exemplarily illustrated in FIG. **10**, the adaptive beamforming unit **203** disclosed herein comprises a fixed beamformer **204**, a blocking matrix **205**, an adaptation control unit **208**, and an adaptive filter **206**. The fixed beamformer **204** adaptively steers the directivity pattern of the microphone array **201** in the direction of the spatial location of the target sound signal from the target sound source for enhancing the target sound signal, when the target sound source is in motion. The sound sensors **301** in the microphone array **201** receive the sound signals $S_1, \ldots, S_4$, which comprise both the target sound signal from the target sound source and the ambient noise signals. The received sound signals are fed as input to the fixed beamformer **204** and the blocking matrix **205**.

The fixed beamformer **204** outputs a signal "b". In an embodiment, the fixed beamformer **204** performs fixed beamforming by filtering and summing output sound signals from the sound sensors **301**. The blocking matrix **205** outputs a signal "z" which primarily comprises the ambient noise signals. The blocking matrix **205** blocks the target sound signal from the target sound source and feeds the ambient noise signals to the adaptive filter **206** to minimize the effect of the ambient noise signals on the enhanced target sound signal.

[0079] The output "z" of the blocking matrix **205** may contain some weak target sound signals due to signal leakage. If the adaptation is active when the target sound signal, for example, speech is present, the speech is cancelled out with the noise. Therefore, the adaptation control unit **208** determines when the adaptation should be applied. The adaptation control unit **208** comprises a target sound signal detector **208a** and a step size adjusting module **208b**. The target sound signal detector **208a** of the adaptation control unit **208** detects the presence or absence of the target sound signal, for example, speech. The step size adjusting module **208b** adjusts the step size for the adaptation process such that when the target sound signal is present, the adaptation is slow for preserving the target sound signal, and when the target sound signal is absent, adaptation is quick for better cancellation of the ambient noise signals.

[0080] The adaptive filter **206** is a filter that adaptively updates filter coefficients of the adaptive filter **206** so that the adaptive filter **206** can be operated in an unknown and changing environment. The adaptive filter **206** adaptively filters the ambient noise signals in response to detecting presence or absence of the target sound signal in the sound signals received from the disparate sound sources. The adaptive filter **206** adapts its filter coefficients with the changes in the ambient noise signals, thereby eliminating distortion in the target sound signal, when the target sound source and the ambient noise signals are in motion. In an embodiment, the adaptive filtering is performed by a set of sub-band adaptive filters using sub-band adaptive filtering as disclosed in the detailed description of FIG. **11**.

[0081] FIG. **11** exemplarily illustrates a system for sub-band adaptive filtering. Sub-band adaptive filtering involves separating a full-band signal into different frequency ranges called sub-bands prior to the filtering process. The sub-band adaptive filtering using sub-band adaptive filters lead to a higher convergence speed compared to using a full-band adaptive filter. Moreover, the noise reduction unit **207** disclosed herein is developed in a sub-band, whereby applying sub-band adaptive filtering provides the same sub-band framework for both beamforming and noise reduction, and thus saves on computational cost.

[0082] As exemplarily illustrated in FIG. **11**, the adaptive filter **206** comprises an analysis filter bank **206a**, an adaptive filter matrix **206b**, and a synthesis filter bank **206c**. The analysis filter bank **206a** splits the enhanced target sound signal (b) from the fixed beamformer **204** and the ambient noise signals (z) from the blocking matrix **205** exemplarily illustrated in FIG. **10** into multiple frequency sub-bands. The analysis filter bank **206a** performs an analysis step where the outputs of the fixed beamformer **204** and the blocking matrix **205** are split into frequency sub bands. The sub-band adaptive filter **206** typically has a shorter impulse response than its full band counterpart. The step size of the sub-bands can be adjusted individually for each sub-band by the step-size adjusting module **208b**, which leads to a higher convergence speed compared to using a full band adaptive filter.

[0083] The adaptive filter matrix **206b** adaptively filters the ambient noise signals in each of the frequency sub-bands in response to detecting the presence or absence of the target sound signal in the sound signals received from the disparate sound sources. The adaptive filter matrix **206b** performs an adaptation step, where the adaptive filter **206** is adapted such that the filter output only contains the target sound signal, for example, speech. The synthesis filter bank **206c** synthesizes a full-band sound signal using the frequency sub-bands of the enhanced target sound signal. The synthesis filter bank **206c** performs a synthesis step where the sub-band sound signal is synthesized into a full-band sound signal. Since the noise reduction and the beamforming are performed in the same sub-band framework, the noise reduction as disclosed in the detailed description of FIG.

13, by the noise reduction unit **207** is performed prior to the synthesis step, thereby reducing computation.

[0084] In an embodiment, the analysis filter bank **206a** is implemented as a perfect-reconstruction filter bank, where the output of the synthesis filter bank **206c** after the analysis and synthesis steps perfectly matches the input to the analysis filter bank **206a**. That is, all the sub-band analysis filter banks **206a** are factorized to operate on prototype filter coefficients and a modulation matrix is used to take advantage of the fast Fourier transform (FFT). Both analysis and synthesize steps require performing frequency shifts in each sub-band, which involves complex value computations with cosines and sinusoids. The method disclosed herein employs the FFT to perform the frequency shifts required in each sub-band, thereby minimizing the amount of multiply-accumulate operations. The implementation of the sub-band analysis filter bank **206a** as a perfect-reconstruction filter bank ensures the quality of the target sound signal by ensuring that the sub-band analysis filter banks **206a** do not distort the target sound signal itself.

[0085] FIG. **12** exemplarily illustrates a graphical representation showing the performance of a perfect-reconstruction filter bank. The solid line represents the input signal to the analysis filter bank **206a**, and the circles represent the output of the synthesis filter bank **206c** after analysis and synthesis. As exemplarily illustrated in FIG. **12**, the output of the synthesis filter bank **206c** perfectly matches the input, and is therefore referred to as the perfect-reconstruction filter bank.

[0086] FIG. **13** exemplarily illustrates a block diagram of a noise reduction unit **207** for performing noise reduction using, for example, a Wiener-filter based noise reduction algorithm. The noise reduction unit **207** performs noise reduction for further suppressing the ambient noise signals after adaptive beamforming, for example, by using a Wiener-filter based noise reduction algorithm, a spectral subtraction noise reduction algorithm, an auditory transform based noise reduction algorithm, or a model based noise reduction algorithm. In an embodiment, the noise reduction unit **207** performs noise reduction in multiple frequency sub-bands employed by an analysis filter bank **206a** of the adaptive beamforming unit **203** for sub-band adaptive beamforming.

[0087] In an embodiment, the noise reduction is performed using the Wiener-filter based noise reduction algorithm. The noise reduction unit **207** explores the short-term and long-term statistics of the target sound signal, for example, speech, and the ambient noise signals, and the wide-band and narrow-band signal-to-noise ratio (SNR) to support a Wiener gain filtering. The noise reduction unit **207** comprises a target sound signal statistics analyzer **207a**, a noise statistics analyzer **207b**, a signal-to-noise ratio (SNR) analyzer **207c**, and a Wiener filter **207d**. The target sound signal statistics analyzer **207a** explores the short-term and long-term statistics of the target sound signal, for example, speech. Similarly, the noise statistics analyzer **207b** explores the short-term and long-term statistics of the ambient noise signals. The SNR analyzer **207c** of the noise reduction unit **207** explores the wide-band and narrow-band signal-to-noise ratio (SNR). After the spectrum of noisy-speech passes through the Wiener filter **207d**, an estimation of the clean-speech spectrum is generated. The synthesis filter bank **206c**, by an inverse process of the analysis filter bank **206a**, reconstructs the signals of the clean speech into a full-band signal, given the estimated spectrum of the clean speech.

[0088] FIG. **14** exemplarily illustrates a hardware implementation of the microphone array system **200** disclosed herein. The hardware implementation of the microphone array system **200** disclosed in the detailed description of FIG. **2** comprises the microphone array **201** having an arbitrary number of sound sensors **301** positioned in an arbitrary configuration, multiple microphone amplifiers **1401**, one or more audio codecs **1402**, a digital signal processor (DSP) **1403**, a flash memory **1404**, one or more power regulators **1405** and **1406**, a battery **1407**, a loudspeaker or a headphone **1408**, and a communication interface **1409**. The microphone array **201** comprises, for example, four or eight sound sensors **301** arranged in a linear or a circular microphone array configuration. The microphone array **201** receives the sound signals.

[0089] Consider an example where the microphone array **201** comprises four sound sensors **301** that pick up the sound signals. Four microphone amplifiers **1401** receive the output sound signals from the four sound sensors **301**. The microphone amplifiers **1401** also referred to as preamplifiers provide a gain to boost the power of the received sound signals for enhancing the sensitivity of the sound sensors **301**. In an example, the gain of the preamplifiers is 20 dB.

[0090] The audio codec **1402** receives the amplified output from the microphone amplifiers **1401**. The audio codec **1402** provides an adjustable gain level, for example, from about -74dB to about 6dB. The received sound signals are in an analog form. The audio codec **1402** converts the four channels of the sound signals in the analog form into digital sound signals. The pre-amplifiers may not be required for some applications. The audio codec **1402** then transmits the digital sound signals to the DSP **1403** for processing of the digital sound signals. The DSP **1403** implements the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207**.

[0091] After the processing, the DSP **1403** either stores the processed signal from the DSP **1403** in a memory device for a recording application, or transmits the processed signal to the communication interface **1409**. The recording application comprises, for example, storing the processed signal onto the memory device for the purposes of playing back the processed signal at a later time. The communication interface **1409** transmits the processed signal, for example, to a computer, the internet, or a radio for communicating the processed signal. In an embodiment, the microphone array system **200** disclosed herein implements a two-way communication device where the signal received from the communication interface **1409** is processed by the DSP **1403** and the processed signal is then played through the loudspeaker or the headphone **1408**.

[0092] The flash memory **1404** stores the code for the DSP **1403** and compressed audio signals. When the microphone array system **200** boots up, the DSP **1403** reads the code from the flash memory **1404** into an internal memory of the DSP **1403** and then starts executing the code. In an embodiment, the audio codec **1402** can be configured for encoding and decoding audio or sound signals during the start up stage by writing to registers of the DSP **1403**. For an eight-sensor microphone array **201**, two four-channel audio codec **1402** chips may be used. The power regulators **1405** and **1406**, for example, linear power regulators **1405** and switch power regulators **1406** provide appropriate voltage and current supply for all the components, for example, **201**, **1401**, **1402**, **1403**, etc., mechanically supported and electrically connected on a circuit board. A universal serial bus (USB) control is built into the DSP **1403**. The battery **1407** is used for powering the microphone array system **200**.

[0093] Consider an example where the microphone array system **200** disclosed herein is implemented on a mixed signal circuit board having a six-layer printed circuit board (PCB). Noisy digital signals easily contaminate the low voltage analog sound signals from the sound sensors **301**. Therefore, the layout of the mixed signal circuit board is carefully partitioned to isolate the analog circuits from the digital circuits. Although both the inputs and outputs of the microphone amplifiers **1401** are in analog form, the microphone amplifiers **1401** are placed in a digital region of the mixed signal circuit board because of their high power consumption **1401** and switch amplifier nature.

[0094] The linear power regulators **1405** are deployed in an analog region of the mixed signal circuit board due to the low noise property exhibited by the linear power regulators **1405**. Five power regulators, for example, **1405** are designed in the microphone array system **200** circuits to ensure quality. The switch power regulators **1406** achieve an efficiency of about 95% of the input power and have high output current capacity; however their outputs are too noisy for analog circuits. The efficiency of the linear power regulators **1405** is determined by the ratio of the output voltage to the input voltage, which is lower than that of the switch power regulators **1406** in most cases. The regulator outputs utilized in the microphone array system **200** circuits are stable, quiet, and suitable for the low power analog circuits.

[0095] In an example, the microphone array system **200** is designed with a microphone array **201** having dimensions of 10 cm x 2.5 cm x 1.5 cm, a USB interface, and an assembled PCB supporting the microphone array **201** and a DSP **1403** having a low power consumption design devised for portable devices, a four-channel codec **1402**, and a flash memory **1404**. The DSP **1403** chip is powerful enough to handle the DSP **1403** computations in the microphone array system **200** disclosed herein. The hardware configuration of this example can be used for any microphone array configuration, with suitable modifications to the software. In an embodiment, the adaptive beamforming unit **203** of the microphone array system **200** is implemented as hardware with software instructions programmed on the DSP **1403**. The DSP **1403** is programmed for beamforming, noise reduction, echo cancellation, and USB interfacing according to the method disclosed herein, and fine tuned for optimal performance.

[0096] FIGS. **15A-15C** exemplarily illustrate a conference phone **1500** comprising an eight-sensor microphone array **201**. The eight-sensor microphone array **201** comprises eight sound sensors **301** arranged in a configuration as exemplarily illustrated in FIG. **15A**. A top view of the conference phone **1500** comprising the eight-sensor microphone array **201** is exemplarily illustrated in FIG. **15A**. A front view of the conference phone **1500** comprising the eight-sensor microphone array **201** is exemplarily illustrated in FIG. **15B**. A headset **1502** that can be placed in a base holder **1501** of the conference phone **1500** having the eight-sensor microphone array **201** is exemplarily illustrated in FIG. **15C**. In addition to a conference phone **1500**, the microphone array system **200** disclosed herein with broadband beamforming can be configured for a mobile phone, a tablet computer, etc., for speech enhancement and noise reduction.

[0097] FIG. **16A** exemplarily illustrates a layout of an eight-sensor microphone array **201** for a conference phone **1500**. Consider an example of a circular microphone array **201** in which eight sound sensors **301** are mounted on the surface of the conference phone **1500** as exemplarily illustrated in FIG. **15A**. The conference phone **1500** has a removable handset **1502** on top, and hence the microphone array system **200** is configured to accommodate the handset **1502** as exemplarily illustrated in FIGS. **15A-15C**. In an example, the circular microphone array **201** has a diameter of about four inches. Eight sound sensors **301**, for example, microphones, M_0, M_1, M_2, M_3, M_4, M_5, M_6, and M_7 are distributed along a circle **302** on the conference phone **1500**. Microphones M_4-M_7 are separated by 90 degrees from each other, and microphones M_0-M_3 are rotated counterclockwise by 60 degrees from microphone M_4-M_7 respectively.

[0098] FIG. **16B** exemplarily illustrates a graphical representation of eight spatial regions to which the eight-sensor microphone array **201** of FIG. **16A** responds. The space is divided into eight spatial regions with equal spaces centered at 15°, 60°, 105°, 150°, 195°, 240°, 285°, and 330° respectively. The adaptive beamforming unit **203** configures the eight-sensor microphone array **201** to automatically point to one of these eight spatial regions according to the location of the target sound signal from the target sound source as estimated by the sound source localization unit **202**.

[0099] FIGS. **16C**-**16D** exemplarily illustrate computer simulations showing the steering of the directivity patterns of the eight-sensor microphone array **201** of FIG. **16A**, in the directions 15° and 60° respectively, in the frequency range 300 Hz to 5 kHz. FIG. **16C** exemplarily illustrates the computer simulation result showing the directivity pattern of the microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 15°.

[0100] The computer simulation for verifying the performance of the adaptive beamforming unit **203** when the target sound signal is received from the target sound source in the spatial region centered at 15° uses the following parameters:

Sampling frequency fs = 16k,
FIR filter taper length L=20

Passband (Θ_p, Ω_p) = {300-5000Hz, -5°– 35°}, designed spatial directivity pattern is 1.
Stopband (Θ_s, Ω_s) = {300~5000Hz, -180°~-15° + 45°~180°}, the designed spatial directivity pattern is 0.

[0101] It can be seen that the directivity pattern of the microphone array **201** in the spatial region centered at 15° is enhanced while the sound signals from all other spatial regions are suppressed.

[0102] FIG. **16D** exemplarily illustrates the computer simulation result showing the directivity pattern of the microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 60°. The computer simulation for verifying the performance of the adaptive beamforming unit **203** when the target sound signal is received from the target sound source in the spatial region centered at 60° uses the following parameters:

Sampling frequency fs = 16k,
FIR filter taper length L=20

Passband (Θ_p, Ω_p) = {300-5000Hz, 40°– 80°}, designed spatial directivity pattern is 1.
Stopband (Θ_s, Ω_s) = {300~5000Hz, -180°~30° + 90°~180°}, the designed spatial directivity pattern is 0.

[0103] It can be seen that the directivity pattern of the microphone array **201** in the spatial region centered at 60° is enhanced while the sound signals from all other spatial regions are suppressed. The other six spatial regions have similar parameters. Moreover, in all frequencies, the main lobe has the same level, which means the target sound signal has little distortion in frequency.

[0104] FIGS. **16E-16L** exemplarily illustrate graphical representations showing the directivity patterns of the eight-sensor microphone array **201** of FIG. **16A** in each of the eight spatial regions, where each directivity pattern is an average response from 300Hz to 5000Hz. The main lobe is about 10dB higher than the side lobe, and therefore the ambient noise signals from other directions are highly suppressed compared to the target sound signal in the pass direction. The microphone array system **200** calculates the filter coefficients for the target sound signal, for example, speech signals from each sound sensor **301** and combines the filtered signals to enhance the speech from any specific direction. Since speech covers a large range of frequencies, the method and system **200** disclosed herein covers broadband signals from 300Hz to 5000Hz.

[0105] FIG. **16E** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 15°. FIG. **16F** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 60°. FIG. **16G** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 105°. FIG. **16H** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 150°. FIG. **16I** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 195°. FIG. **16J** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 240°. FIG. **16K** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 285°. FIG. **16L** exemplarily illustrates a graphical representation showing the directivity pattern of the eight-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 330°. The microphone array system **200** disclosed herein enhances the target sound signal from each of the directions 15°, 60°, 105°, 150°, 195°, 240°, 285°, and 330°, while suppressing the ambient noise signals from the other directions.

[0106] The microphone array system **200** disclosed herein can be implemented for a square microphone array configuration and a rectangular array configuration where a sound sensor **301** is positioned in each corner of the four-cornered array. The microphone array system **200** disclosed herein implements beamforming from plane to three dimensional sound sources.

[0107] FIG. **17A** exemplarily illustrates a graphical representation of four spatial regions to which a four-sensor microphone array **201** for a wireless handheld device responds. The wireless handheld device is, for example, a mobile phone. Consider an example where the microphone array **201** comprises four sound sensors **301**, for example, microphones, uniformly distributed around a circle **302** having diameter equal to about two inches. This configuration is identical to positioning four sound sensors **301** or microphones on four corners of a square. The space is divided into four spatial regions with equal space centered at -90^{o}, 0^{o}, 90^{o}, and 180^{o} respectively. The adaptive beamforming unit **203** configures the four-sensor microphone array **201** to automatically point to one of these spatial regions according to the location of the target sound signal from the target sound source as estimated by the sound source localization unit **202**.

[0108] FIGS. **17B**-**17I** exemplarily illustrate computer simulations showing the directivity patterns of the four-sensor microphone array **201** of FIG. **17A** with respect to azimuth and frequency. The results of the computer simulations performed for verifying the performance of the adaptive beamforming unit **203** of the microphone array system **200** disclosed herein for a sampling frequency f_s = 16k and FIR filter taper length L=20, are as follows:

For the spatial region centered at 0^{o}:

Passband (Θ_p, Ω_p) = {300-4000Hz, -20^{o}– 20^{o}}, designed spatial directivity pattern is 1.

Stopband (Θ_s, Ω_s) = {300~4000Hz, -180^{o}~-30^{o} + 30^{o}~180^{o}}, the designed spatial directivity pattern is 0.

[0109] For the spatial region centered at 90^{o}:

Passband (Θ_p, Ω_p) = {300-4000Hz, 70^{o}– 110^{o}}, designed spatial directivity pattern is 1.

Stopband $(\Theta_s, \Omega_s) = \{300\text{~}4000\text{Hz}, -180^o\text{~}60^o + 120^o\text{~}180^o\}$, the designed spatial directivity pattern is 0. The directivity patterns for the spatial regions centered at -90^o and 180^o are similarly obtained.

[0110] FIG. **17B** exemplarily illustrates the computer simulation result representing a three dimensional (3D) display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at -90^o. FIG. **17C** exemplarily illustrates the computer simulation result representing a 2D display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at -90^o.

[0111] FIG. **17D** exemplarily illustrates the computer simulation result representing a 3D display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 0^o. FIG. **17E** exemplarily illustrates the computer simulation result representing a 2D display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 0^o.

[0112] FIG. **17F** exemplarily illustrates the computer simulation result representing a 3D display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 90^o. FIG. **17G** exemplarily illustrates the computer simulation result representing a 2D display of the directivity pattern of the four-sensor microphone array **201** when the target sound signal is received from the target sound source in the spatial region centered at 90^o.

[0113] FIG. **17H** exemplarily illustrates the computer simulation result representing a 3D display of the directivity pattern of the four-sensor microphone array **201** when the target sound source is received from the target sound source in the spatial region centered at 180^o. FIG. **17I** exemplarily illustrates the computer simulation result representing a 2D display of the directivity pattern of the four-sensor microphone array **201** when the target sound source is received from the target sound source in the spatial region centered at 180°. The 3D displays of the directivity patterns in FIG. **17B**, FIG. **17D**, FIG. **17F**, and FIG. **17H** demonstrate that the passbands have the same height. The 2D displays of the directivity patterns in FIG. **17C**, FIG. **17E**, FIG. **17G**, and FIG. **17I** demonstrate that the passbands have the same width along the frequency and demonstrates the broadband properties of the microphone array **201**.

[0114] FIGS. **18A**-**18B** exemplarily illustrates a microphone array configuration for a tablet computer. In this example, four sound sensors **301** of the microphone array **201** are positioned on a frame **1801** of the tablet computer, for example, the iPad® of Apple Inc. Geometrically, the sound sensors **301** are distributed on the circle **302** as exemplarily in FIG. **18B**. The radius of the circle **302** is equal to the width of the tablet computer. The angle θ between the sound sensors **301** M_2 and M_3 is determined to avoid spatial aliasing up to 4000Hz. This microphone array configuration enhances a front speaker's voice and suppresses background ambient noise. The adaptive beamforming unit **203** configures the microphone array **201** to form an acoustic beam **1802** pointing frontwards using the method and system **200** disclosed herein. The target sound signal, that is, the front speaker's voice within the range of $\Phi<30^{\circ}$ is enhanced compared to the sound signals from other directions.

[0115] FIG. **18C** exemplarily illustrates an acoustic beam **1802** formed using the microphone array configuration of FIGS. **18A-18B** according to the method and system **200** disclosed herein.

[0116] FIGS. **18D**-**18G** exemplarily illustrates graphs showing processing results of the adaptive beamforming unit **203** and the noise reduction unit **207** for the microphone array configuration of FIG. **18B**, in both a time domain and a spectral domain for the tablet computer. Consider an example where a speaker is talking in front of the tablet computer with ambient noise signals on the side. FIG. **18D** exemplarily illustrates a graph showing the performance of the microphone array **201** before performing beamforming and noise reduction with a signal-to-noise ratio (SNR) of 15 dB. FIG. **18E** exemplarily illustrates a graph showing the performance of the microphone array **201** after performing beamforming and noise reduction, according to the method disclosed herein, with an SNR of 15 dB. FIG. **18F** exemplarily illustrates a graph showing the performance of the microphone array **201** before performing beamforming and noise reduction with an SNR of 0 dB. FIG. **18G** exemplarily illustrates a graph showing the performance of the microphone array **201** after performing beamforming and noise reduction, according to the method disclosed herein, with an SNR of 0 dB.

[0117] It can be seen from FIGS. **18D**-**18G** that the performance graph is noisier for the microphone array **201** before the beamforming and noise reduction is performed. Therefore, the adaptive beamforming unit **203** and the noise reduction unit **207** of the microphone array system **200** disclosed herein suppresses ambient noise signals while maintaining the clarity of the target sound signal, for example, the speech signal.

[0118] FIGS. **19A-19F** exemplarily illustrate tables showing different microphone array configurations and the corresponding values of delay τ_n for the sound sensors **301** in each of the microphone array configurations. The broadband beamforming method disclosed herein can be used for microphone arrays **201** with arbitrary numbers of sound sensors **301** and arbitrary locations of the sound sensors **301**. The sound sensors **301** can be mounted on surfaces or edges of any speech acquisition device. For any specific microphone array configuration, the only parameter that needs to be defined to achieve the beamformer coefficients is the value of τ_n for each sound sensor **301** as disclosed in the detailed description of FIG. **5**, FIGS. **6A-6B**, and FIGS. **7A-7C** and as exemplarily illustrated in FIGS. **19A**-**19F**. In an example, the microphone array configuration exemplarily illustrated in FIG. **19F** is implemented on a handheld device for hands-free speech acquisition. In a hands-free and non-close talking scenario, a user prefers to talk in distance rather than speaking close to the sound sensor **301** and may want to talk while watching a screen of the handheld device. The microphone array system **200** disclosed herein allows the handheld device to pick up sound signals from the direction of the speaker's mouth and suppress noise from other directions. The method and system **200**

disclosed herein may be implemented on any device or equipment, for example, a voice recorder where a target sound signal or speech needs to be enhanced.

[0119] The foregoing examples have been provided merely for the purpose of explanation and are in no way to be construed as limiting of the present invention disclosed herein. While the invention has been described with reference to various embodiments, it is understood that the words, which have been used herein, are words of description and illustration, rather than words of limitation. Further, although the invention has been described herein with reference to particular means, materials and embodiments, the invention is not intended to be limited to the particulars disclosed herein; rather, the invention extends to all functionally equivalent structures, methods and uses, such as are within the scope of the appended claims. Those skilled in the art, having the benefit of the teachings of this specification, may affect numerous modifications thereto and changes may be made without departing from the scope and spirit of the invention in its aspects.

CLAIMS

We claim:

1. A method for enhancing a target sound signal from a plurality of sound signals, comprising:

    providing a microphone array system comprising an array of sound sensors positioned in an arbitrary configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, wherein said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are in operative communication with said array of said sound sensors;

    receiving said sound signals from a plurality of disparate sound sources by said sound sensors, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;

    estimating a spatial location of said target sound signal from said received sound signals by said sound source localization unit;

    performing adaptive beamforming for steering a directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal by said adaptive beamforming unit, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

    suppressing said ambient noise signals by said noise reduction unit for further enhancing said target sound signal.

2. The method of claim 1, wherein said spatial location of said target sound signal from said target sound source is estimated using a steered response power-phase transform by said sound source localization unit.

3. The method of claim 1, further comprising determining a delay between each of said sound sensors and origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a reference axis, and an azimuth angle between said reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a two dimensional plane, wherein said delay is represented in terms of number of samples, wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors.

4. The method of claim 1, further comprising determining a delay between each of said sound sensors and origin of said array of said sound sensors as a function of distance between each of said sound sensors and said origin, a predefined angle between each of said sound sensors and a first reference axis, an elevation angle between a second reference axis and said target sound signal, and an azimuth angle between said first reference axis and said target sound signal, when said target sound source that emits said target sound signal is in a three dimensional plane, wherein said delay is represented in terms of number of samples, wherein said determination of said delay enables beamforming for arbitrary numbers of said sound sensors and a plurality of arbitrary configurations of said array of said sound sensors.

5. The method of claim 1, wherein said adaptive beamforming comprises:

   providing a fixed beamformer, a blocking matrix, and an adaptive filter in said adaptive beamforming unit;

steering said directivity pattern of said array of said sound sensors in said direction of said spatial location of said target sound signal from said target sound source by said fixed beamformer for enhancing said target sound signal, when said target sound source is in motion;

feeding said ambient noise signals to said adaptive filter by blocking said target sound signal received from said target sound source using said blocking matrix; and

adaptively filtering said ambient noise signals by said adaptive filter in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources.

6. The method of claim 5, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

7. The method of claim 5, wherein said adaptive filtering comprises sub-band adaptive filtering performed by said adaptive filter, wherein said sub-band adaptive filtering comprises:

providing an analysis filter bank, an adaptive filter matrix, and a synthesis filter bank in said adaptive filter;

splitting said enhanced target sound signal from said fixed beamformer and said ambient noise signals from said blocking matrix into a plurality of frequency sub-bands by said analysis filter bank;

adaptively filtering said ambient noise signals in each of said frequency sub-bands by said adaptive filter matrix in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources; and synthesizing a full-band sound signal using said frequency sub-bands of said enhanced target sound signal by said synthesis filter bank.

8. The method of claim 5, wherein said adaptive beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said adaptive beamforming unit and adjusting a step size for said adaptive filtering in response to detecting one of said presence and said absence of said target sound signal in said sound signals received from said disparate sound sources.

9. The method of claim 1, wherein said noise reduction unit performs noise reduction by using one of a Wiener-filter based noise reduction algorithm, a spectral subtraction noise reduction algorithm, an auditory transform based noise reduction algorithm, and a model based noise reduction algorithm.

10. The method of claim 1, wherein said noise reduction unit performs noise reduction in a plurality of frequency sub-bands, wherein said frequency sub-bands are employed by an analysis filter bank of said adaptive beamforming unit for sub-band adaptive beamforming.

11. A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors positioned in an arbitrary configuration, wherein said sound sensors receive said sound signals from a plurality of disparate sound sources, wherein said received sound signals comprise said target sound signal from a target sound source among said disparate sound sources, and ambient noise signals;

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals;

an adaptive beamforming unit that steers directivity pattern of said array of said sound sensors in a direction of said spatial location of said target sound signal, wherein said adaptive beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals for further enhancing said target sound signal.

12. The system of claim 11, wherein said sound source localization unit estimates said spatial location of said target sound signal from said target sound source using a steered response power-phase transform.

13. The system of claim 11, wherein said adaptive beamforming unit comprises:

a fixed beamformer that steers said directivity pattern of said array of said sound sensors in said direction of said spatial location of said target sound signal from said target sound source for enhancing said target sound signal, when said target sound source is in motion;

a blocking matrix that feeds said ambient noise signals to an adaptive filter by blocking said target sound signal received from said target sound source; and

said adaptive filter that adaptively filters said ambient noise signals in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources.

14. The system of claim 13, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

15. The system of claim 13, wherein said adaptive filter comprises a set of sub-band adaptive filters comprising:

an analysis filter bank that splits said enhanced target sound signal from said fixed beamformer and said ambient noise signals from said blocking matrix into a plurality of frequency sub-bands;

an adaptive filter matrix that adaptively filters said ambient noise signals in each of said frequency sub-bands in response to detecting one of presence and absence of said target sound signal in said sound signals received from said disparate sound sources; and

a synthesis filter bank that synthesizes a full-band sound signal using said frequency sub-bands of said enhanced target sound signal.

16. The system of claim 11, wherein said adaptive beamforming unit further comprises an adaptation control unit that detects said presence of said target sound signal and adjusts a step size for said adaptive filtering in response to detecting one of said presence and said absence of said target sound signal in said sound signals received from said disparate sound sources.

17. The system of claim 11, wherein said noise reduction unit is one of a Wiener-filter based noise reduction unit, a spectral subtraction noise reduction unit, an auditory transform based noise reduction unit, and a model based noise reduction unit.

18. The system of claim 11, further comprising one or more audio codecs that convert said sound signals in an analog form of said sound signals into digital sound signals and reconverts said digital sound signals into said analog form of said sound signals.

19. The system of claim 11, wherein said noise reduction unit performs noise reduction in a plurality of frequency sub-bands employed by an analysis filter bank of said adaptive beamforming unit for sub-band adaptive beamforming.

20. The system of claim 11, wherein said array of said sound sensors is one of a linear array of said sound sensors, a circular array of said sound sensors, and an arbitrarily distributed coplanar array of said sound sensors.

ABSTRACT

A method and system for enhancing a target sound signal from multiple sound signals is provided. An array of an arbitrary number of sound sensors positioned in an arbitrary configuration receives the sound signals from multiple disparate sources. The sound signals comprise the target sound signal from a target sound source, and ambient noise signals. A sound source localization unit, an adaptive beamforming unit, and a noise reduction unit are in operative communication with the array of sound sensors. The sound source localization unit estimates a spatial location of the target sound signal from the received sound signals. The adaptive beamforming unit performs adaptive beamforming by steering a directivity pattern of the array of sound sensors in a direction of the spatial location of the target sound signal, thereby enhancing the target sound signal and partially suppressing the ambient noise signals, which are further suppressed by the noise reduction unit.

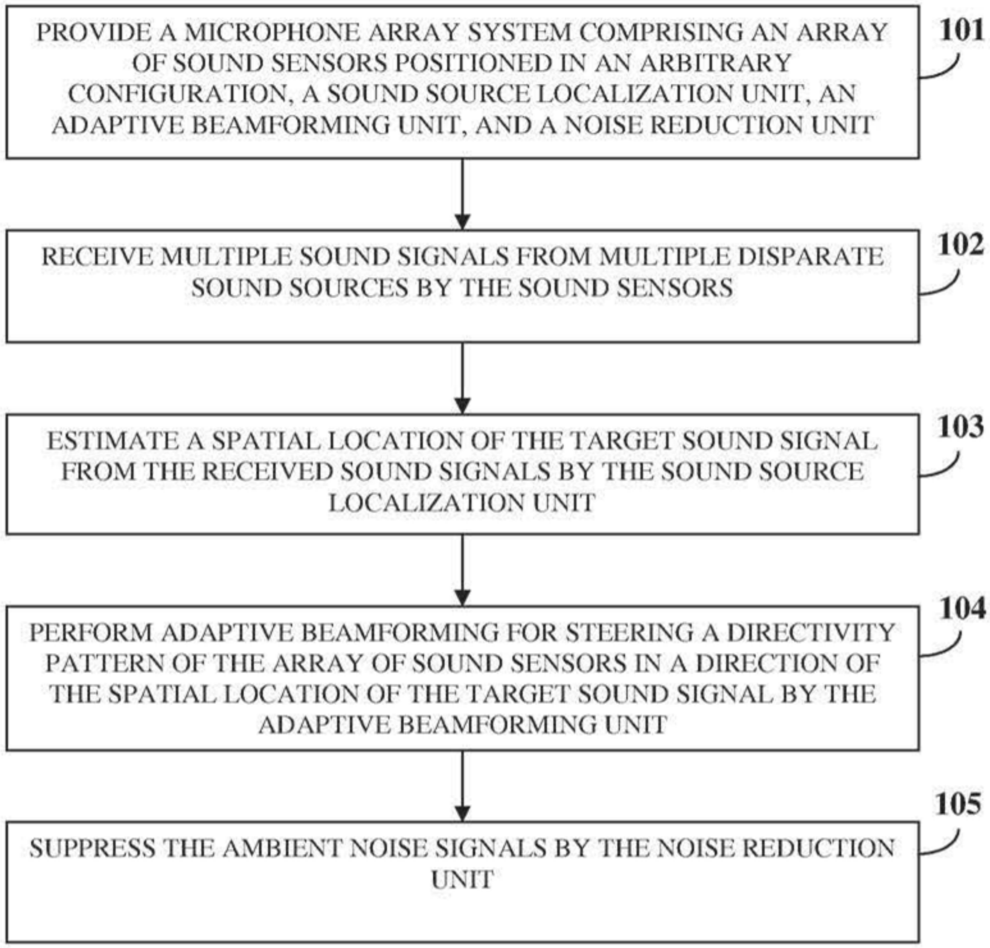

PROVIDE A MICROPHONE ARRAY SYSTEM COMPRISING AN ARRAY OF SOUND SENSORS POSITIONED IN AN ARBITRARY CONFIGURATION, A SOUND SOURCE LOCALIZATION UNIT, AN ADAPTIVE BEAMFORMING UNIT, AND A NOISE REDUCTION UNIT
101
RECEIVE MULTIPLE SOUND SIGNALS FROM MULTIPLE DISPARATE SOUND SOURCES BY THE SOUND SENSORS
102
ESTIMATE A SPATIAL LOCATION OF THE TARGET SOUND SIGNAL FROM THE RECEIVED SOUND SIGNALS BY THE SOUND SOURCE LOCALIZATION UNIT
103
PERFORM ADAPTIVE BEAMFORMING FOR STEERING A DIRECTIVITY PATTERN OF THE ARRAY OF SOUND SENSORS IN A DIRECTION OF THE SPATIAL LOCATION OF THE TARGET SOUND SIGNAL BY THE ADAPTIVE BEAMFORMING UNIT
104
SUPPRESS THE AMBIENT NOISE SIGNALS BY THE NOISE REDUCTION UNIT
105

FIG. 1

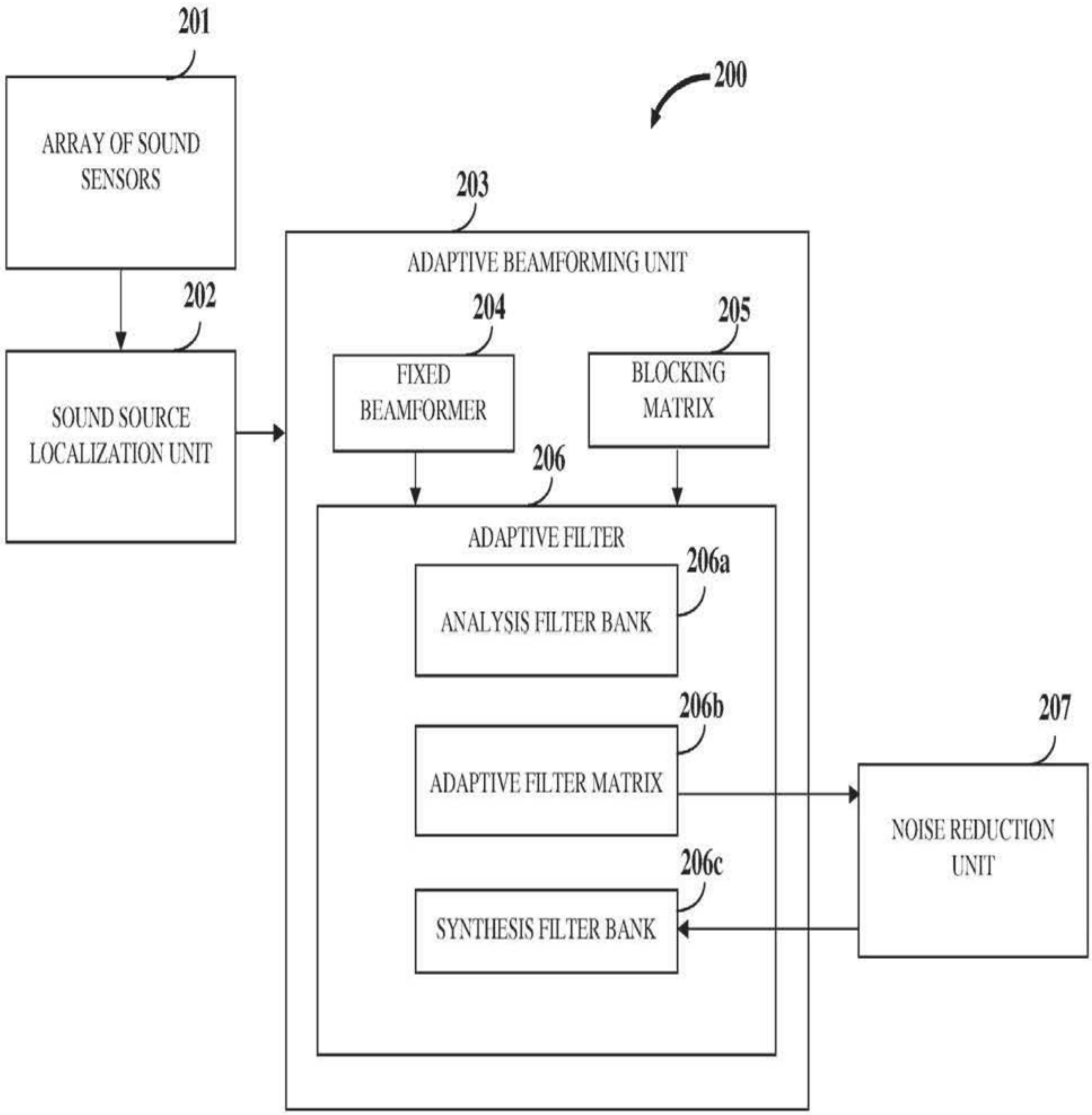

201
200
ARRAY OF SOUND SENSORS
203
ADAPTIVE BEAMFORMING UNIT
202
204
205
FIXED BEAMFORMER
BLOCKING MATRIX
SOUND SOURCE LOCALIZATION UNIT
206
ADAPTIVE FILTER
206a
ANALYSIS FILTER BANK
206b
207
ADAPTIVE FILTER MATRIX
NOISE REDUCTION UNIT
206c
SYNTHESIS FILTER BANK

FIG. 2

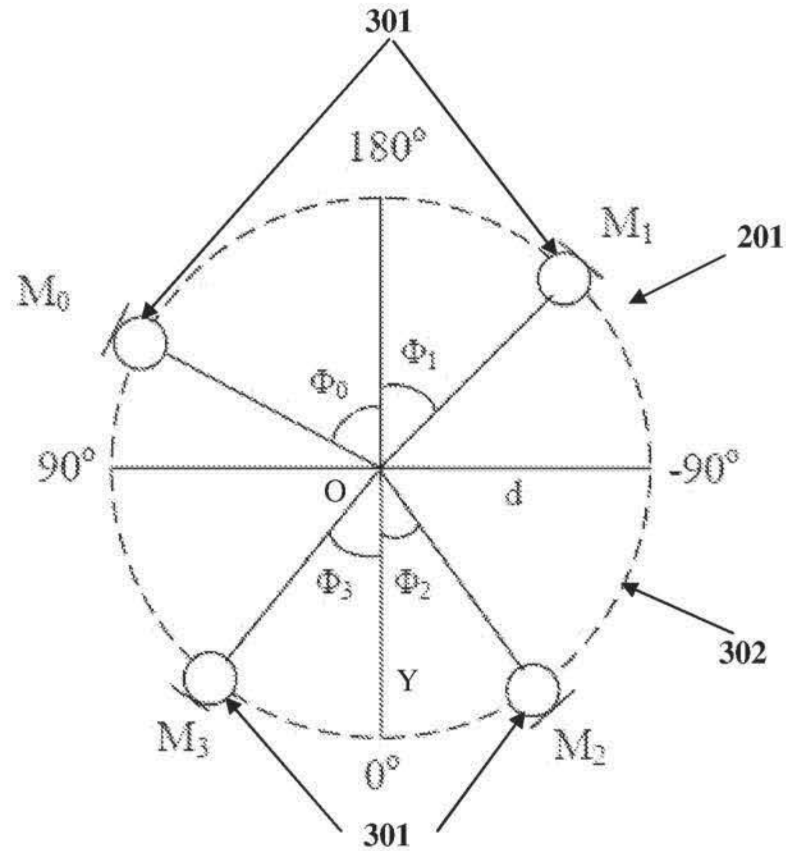


FIG. 3

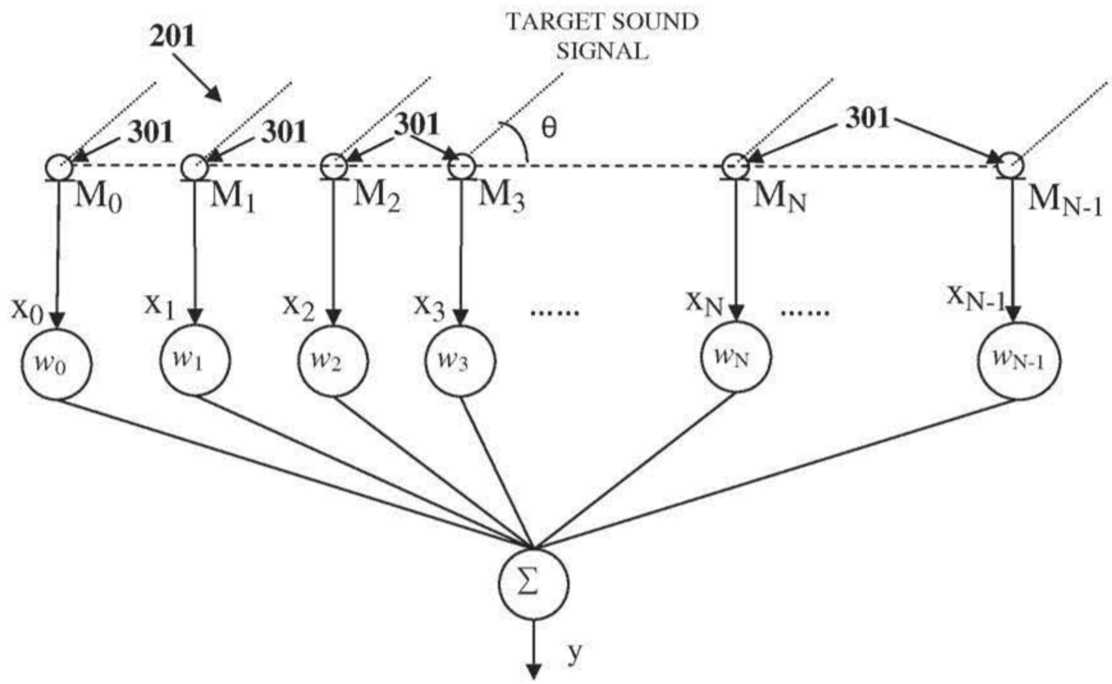

TARGET SOUND SIGNAL
201
301
301
301
301
θ
M0
M1
M2
M3
MN
MN-1
x0
x1
x2
x3
......
xN
......
xN-1
w0
w1
w2
w3
wN
wN-1
Σ
y

FIG. 4

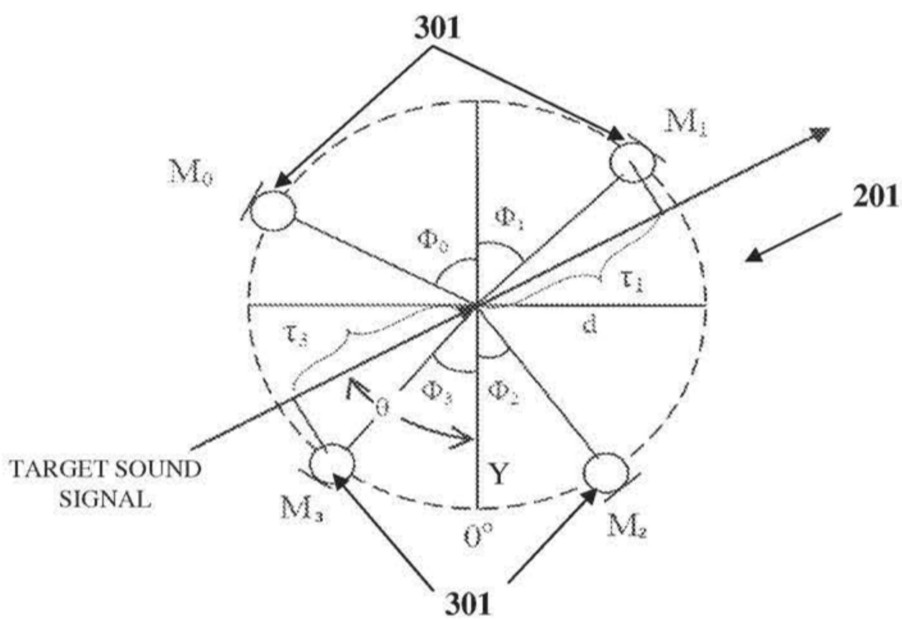

301
M1
M0
201
Φ1
Φ0
τ1
d
τ3
Φ3
Φ2
θ
TARGET SOUND SIGNAL
Y
M3
M2
0°
301

FIG. 5

Sound Sensor	Distance (m)	Delay τ (number of samples)
M0	$d*\cos(\theta+\Phi_0)$	$d*\cos(\theta+\Phi_0)*f_s/c$
M1	$d*\cos(\theta-\Phi_1)$	$d*\cos(\theta-\Phi_1)*f_s/c$
M2	$-d*\cos(\theta+\Phi_2)$	$-d*\cos(\theta+\Phi_2)*f_s/c$
M3	$-d*\cos(\theta-\Phi_3)$	$-d*\cos(\theta-\Phi_3)*f_s/c$

FIG. 6A

Sound Sensor Position	Distance (m)	Delay τ (number of samples)
0^o	$-d*\cos(\theta)$	$-d*\cos(\theta)*f_s/c$
180^o	$d*\cos(\theta)$	$d*\cos(\theta)*f_s/c$
90^o	$-d*\sin(\theta)$	$-d*\sin(\theta)*f_s/c$
-90^o	$d*\sin(\theta)$	$d*\sin(\theta)*f_s/c$
Φ clockwise away from 0^o $(0\leq\Phi\leq90^o)$	$-d*\cos(\theta-\Phi)$	$-d*\cos(\theta-\Phi)*f_s/c$
Φ anticlockwise away from 0^o $(0\leq\Phi\leq90^o)$	$-d*\cos(\theta+\Phi)$	$-d*\cos(\theta+\Phi)*f_s/c$
Φ clockwise away from 180^o $(0\leq\Phi\leq90^o)$	$d*\cos(\theta-\Phi)$	$d*\cos(\theta-\Phi)*f_s/c$
Φ anticlockwise away from 180^o $(0\leq\Phi\leq90^o)$	$d*\cos(\theta+\Phi)$	$d*\cos(\theta+\Phi)*f_s/c$

FIG. 6B

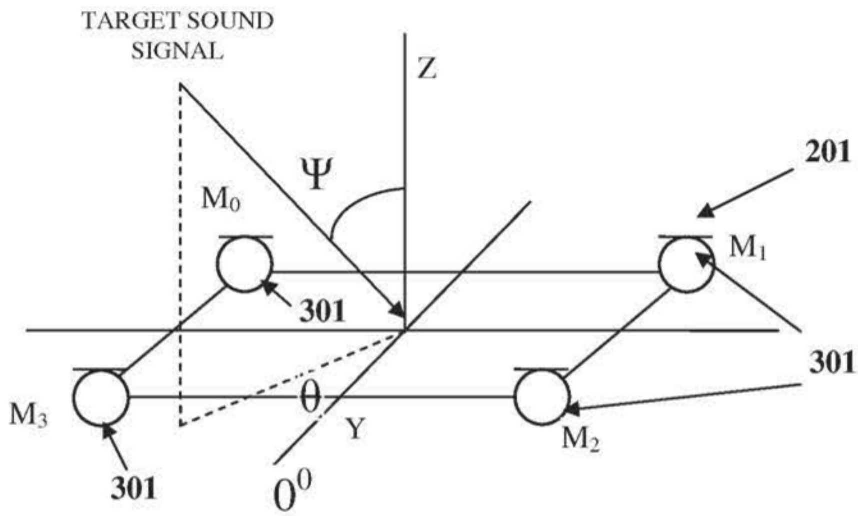


FIG. 7A

$\tau_0 =$ $d \cdot fs \cdot \cos(\theta+\Phi_0)\sin(\Psi)/c$	$\tau_1 =$ $d \cdot fs \cdot \cos(\theta-\Phi_1)\sin(\Psi)/c$	$\tau_2 =$ $-d \cdot fs \cdot \cos(\theta+\Phi_2)\sin(\Psi)/c$	$\tau_3 =$ $-d \cdot fs \cdot \cos(\theta-\Phi_3)\sin(\Psi)/c$

FIG. 7B

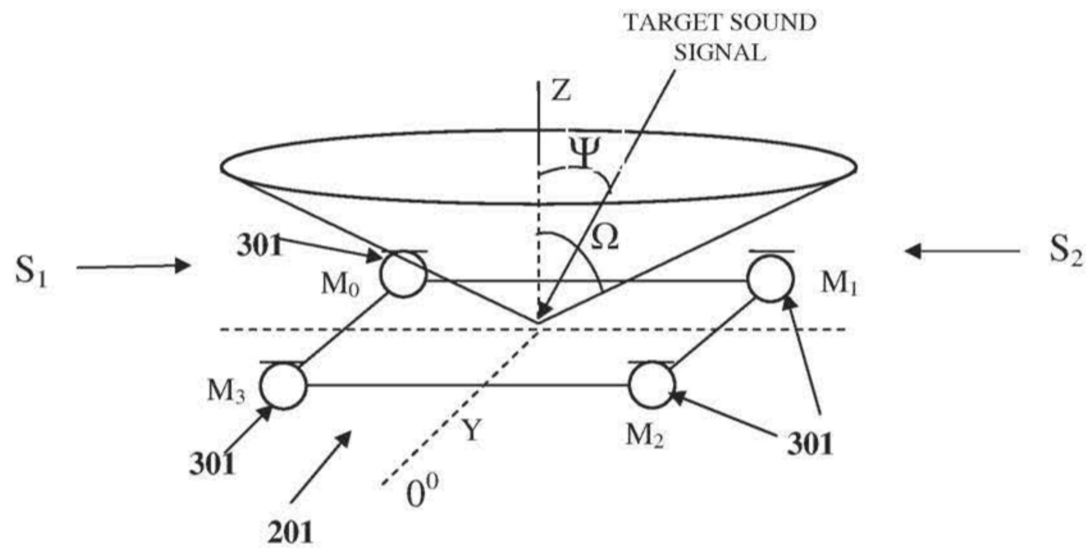


FIG. 7C

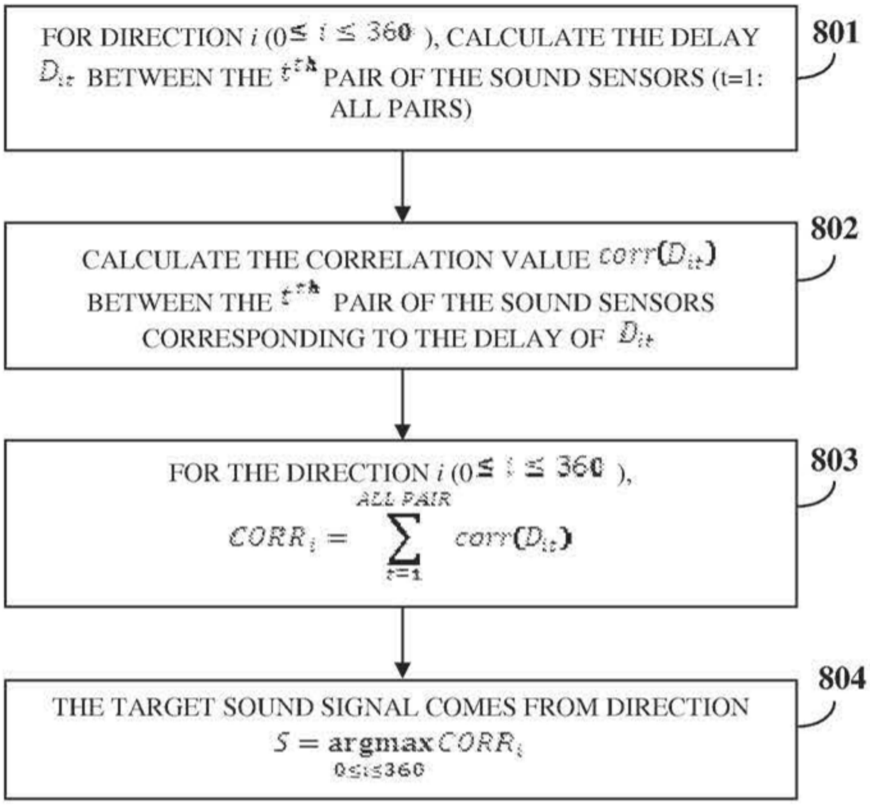
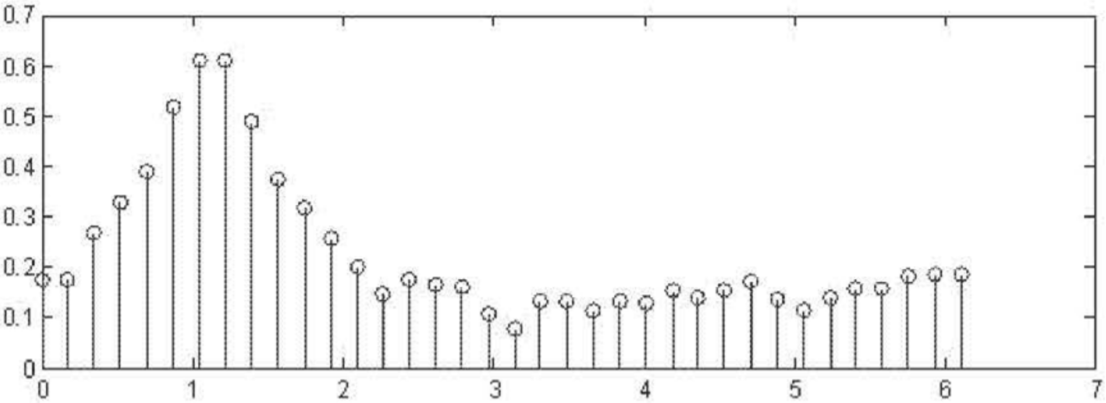
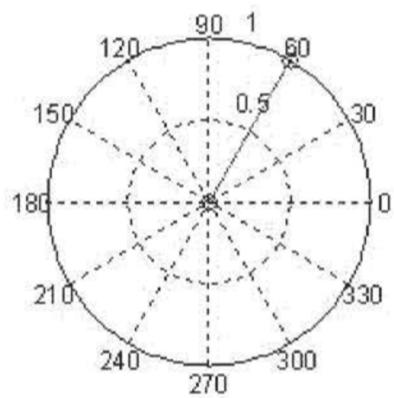

FOR DIRECTION i ($0 \leq i \leq 360$), CALCULATE THE DELAY D_{it} BETWEEN THE t^{th} PAIR OF THE SOUND SENSORS (t=1: ALL PAIRS)
801
CALCULATE THE CORRELATION VALUE $corr(D_{it})$ BETWEEN THE t^{th} PAIR OF THE SOUND SENSORS CORRESPONDING TO THE DELAY OF D_{it}
802
FOR THE DIRECTION i ($0 \leq i \leq 360$), $CORR_i = \sum_{t=1}^{ALL\ PAIR} corr(D_{it})$
803
THE TARGET SOUND SIGNAL COMES FROM DIRECTION $S = \mathbf{argmax}_{0 \leq i \leq 360} CORR_i$
804

FIG. 8



0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
1
2
3
4
5
6
7

FIG. 9A



90
1
120
60
0.5
150
30
180
0
210
330
240
300
270

FIG. 9B

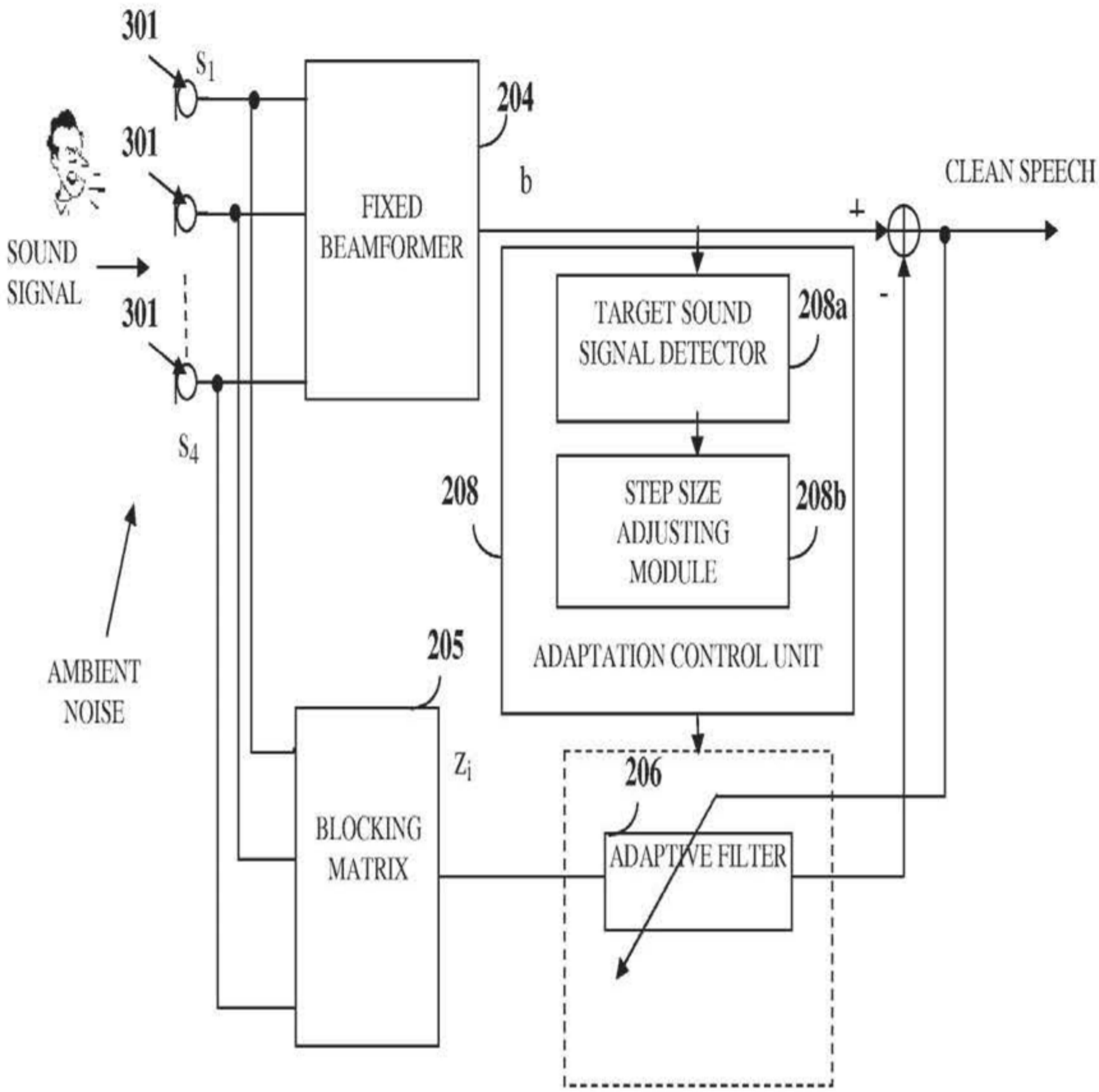

301
S_1
301
301
S_4
SOUND SIGNAL
AMBIENT NOISE
204
FIXED BEAMFORMER
b
CLEAN SPEECH
+
-
TARGET SOUND SIGNAL DETECTOR
208a
208
STEP SIZE ADJUSTING MODULE
208b
ADAPTATION CONTROL UNIT
205
BLOCKING MATRIX
z_i
206
ADAPTIVE FILTER

FIG. 10

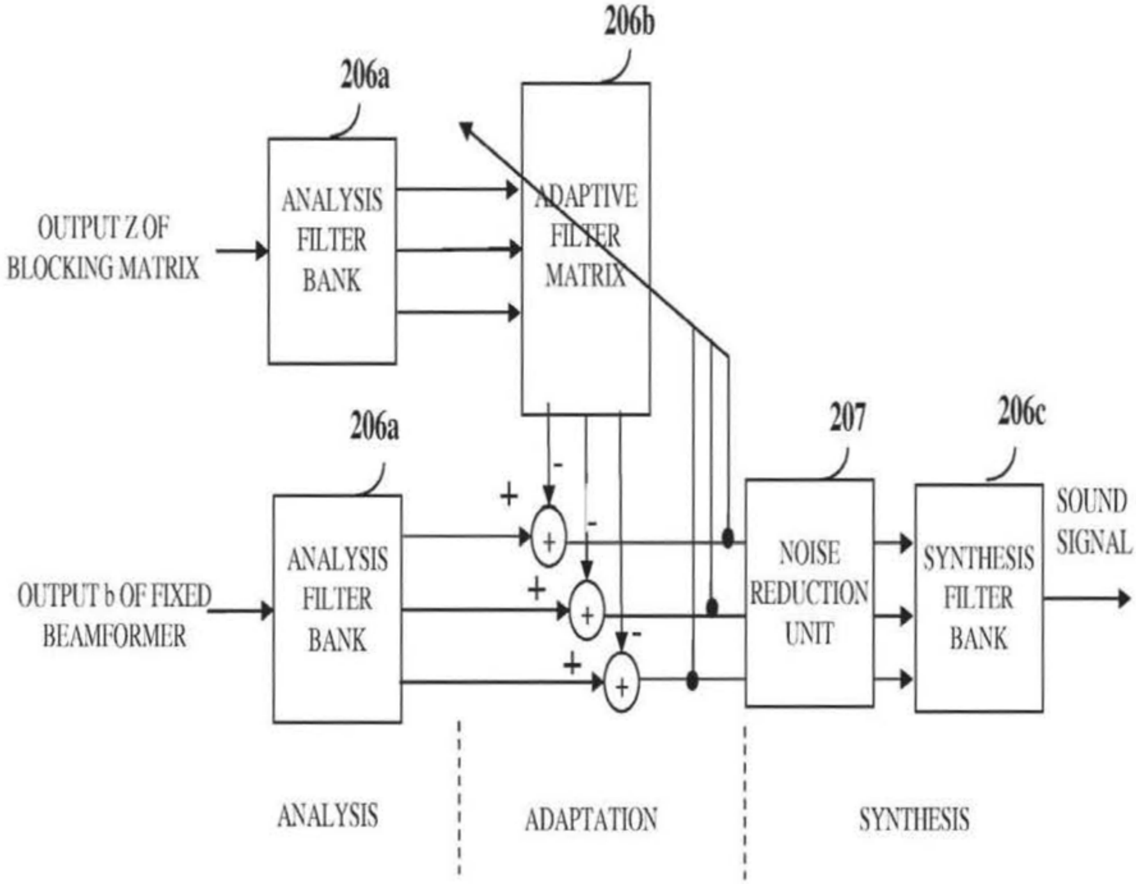

206b
206a
OUTPUT Z OF
BLOCKING MATRIX
ANALYSIS
FILTER
BANK
ADAPTIVE
FILTER
MATRIX
206a
OUTPUT b OF FIXED
BEAMFORMER
ANALYSIS
FILTER
BANK
207
NOISE
REDUCTION
UNIT
206c
SYNTHESIS
FILTER
BANK
SOUND
SIGNAL
ANALYSIS
ADAPTATION
SYNTHESIS

FIG. 11

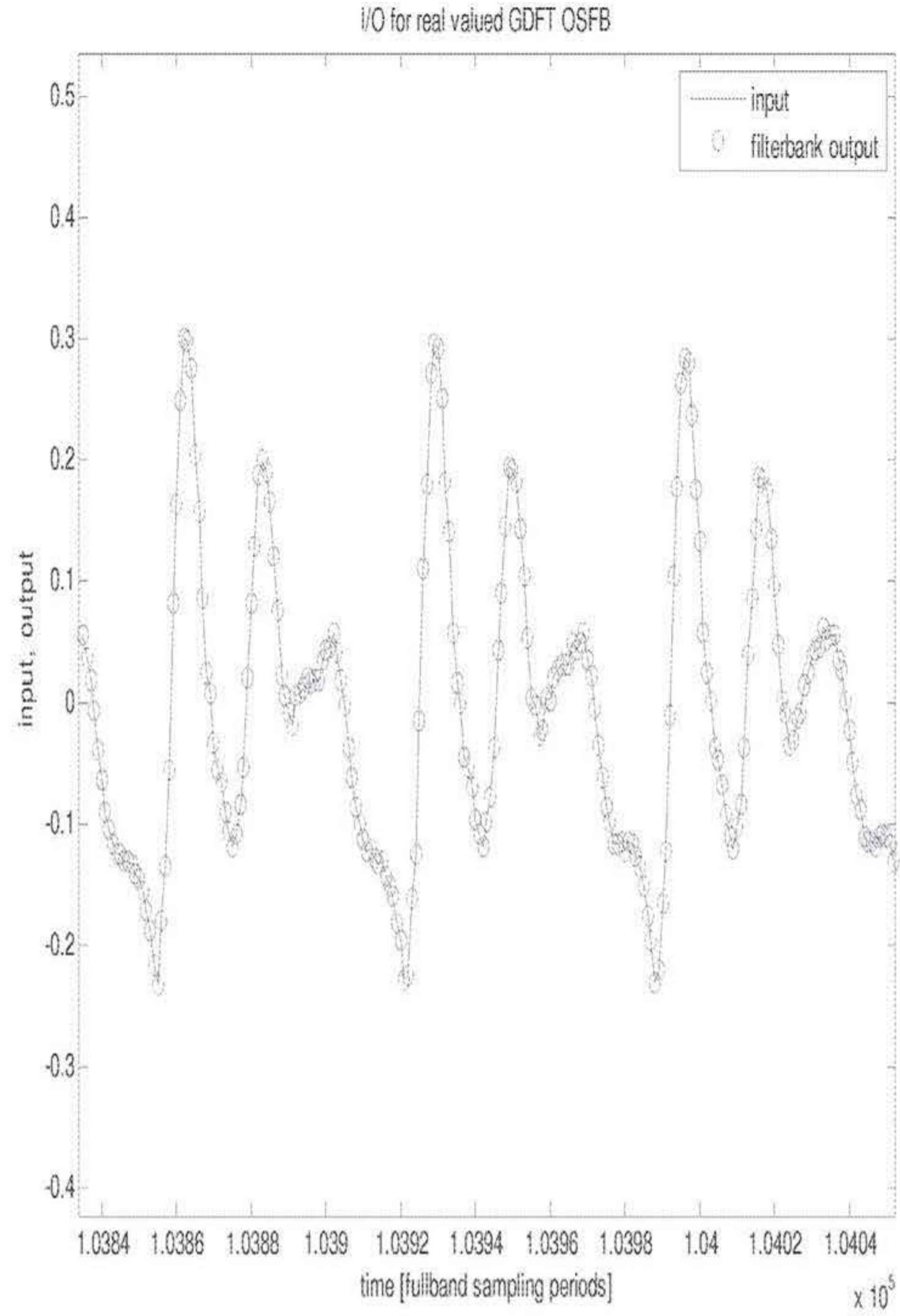

I/O for real valued GDFT OSFB
input
filterbank output
input, output
0.5
0.4
0.3
0.2
0.1
0
-0.1
-0.2
-0.3
-0.4
1.0384
1.0386
1.0388
1.039
1.0392
1.0394
1.0396
1.0398
1.04
1.0402
1.0404
time [fullband sampling periods]
x 10^5

FIG. 12

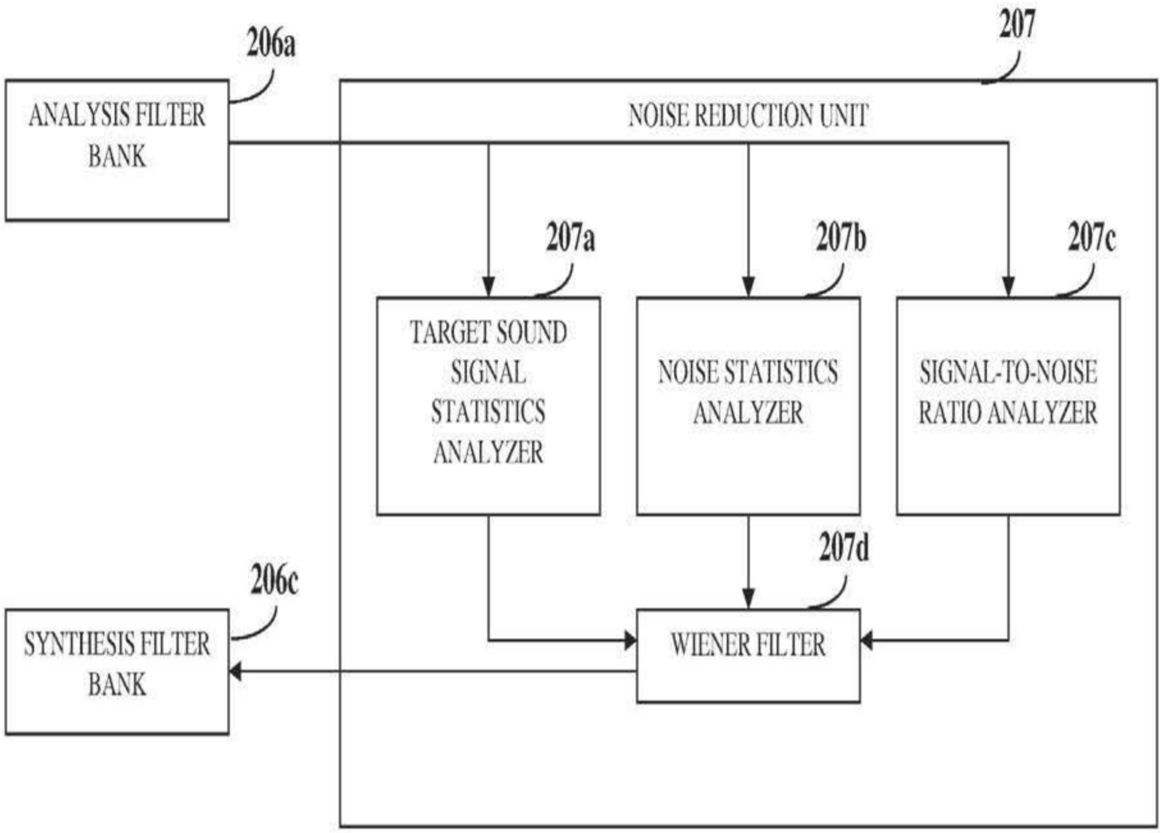

206a
ANALYSIS FILTER BANK
207
NOISE REDUCTION UNIT
207a
TARGET SOUND SIGNAL STATISTICS ANALYZER
207b
NOISE STATISTICS ANALYZER
207c
SIGNAL-TO-NOISE RATIO ANALYZER
207d
WIENER FILTER
206c
SYNTHESIS FILTER BANK

FIG. 13

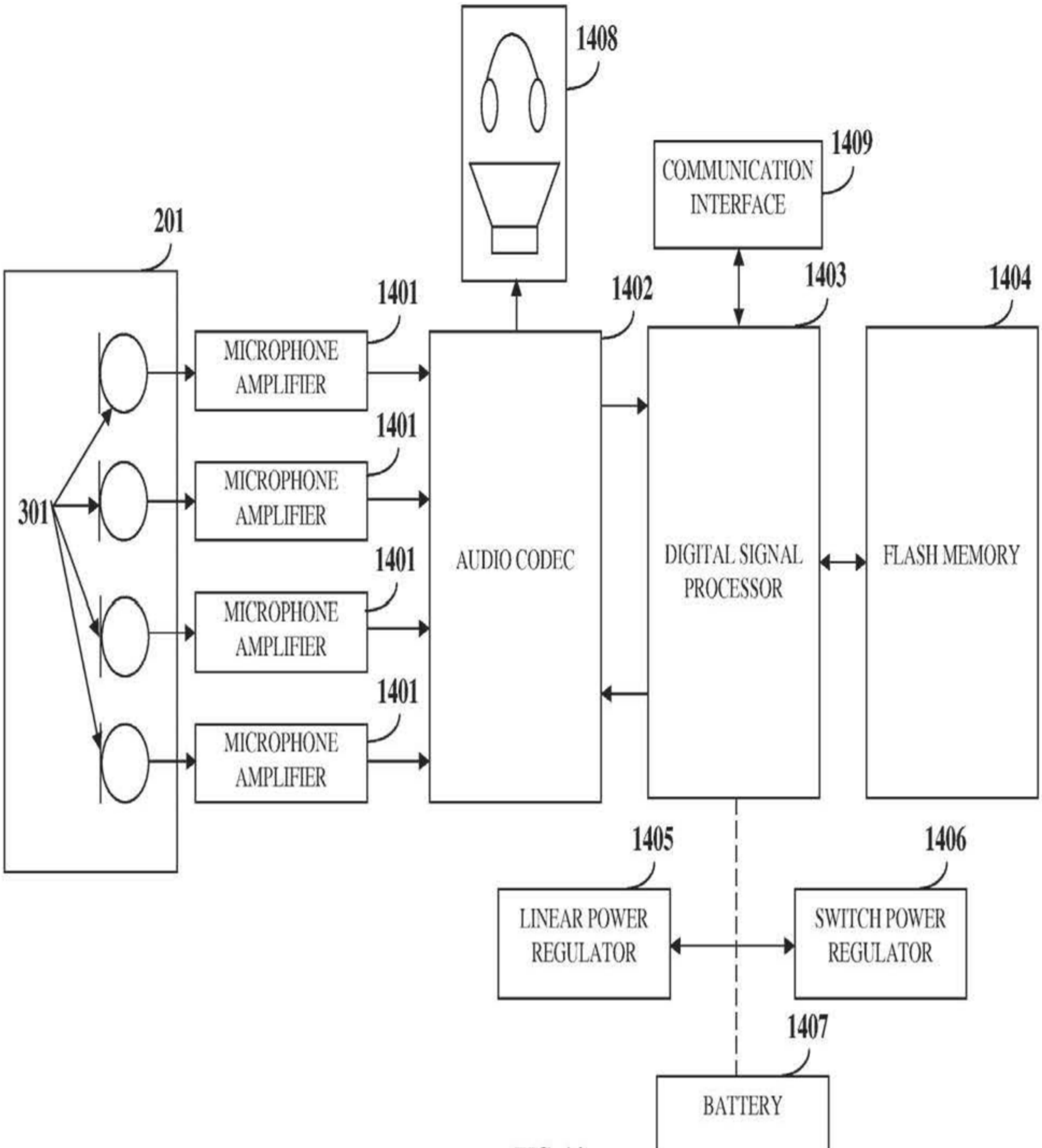

1408
1409
COMMUNICATION INTERFACE
201
1401
1402
1403
1404
MICROPHONE AMPLIFIER
1401
MICROPHONE AMPLIFIER
301
1401
AUDIO CODEC
DIGITAL SIGNAL PROCESSOR
FLASH MEMORY
MICROPHONE AMPLIFIER
1401
MICROPHONE AMPLIFIER
1405
1406
LINEAR POWER REGULATOR
SWITCH POWER REGULATOR
1407
BATTERY

FIG. 14

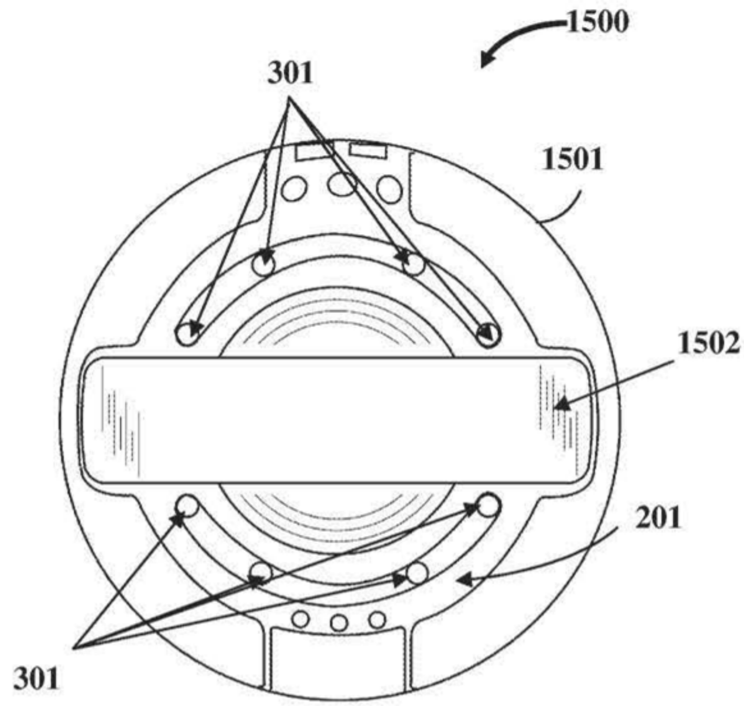


FIG. 15A

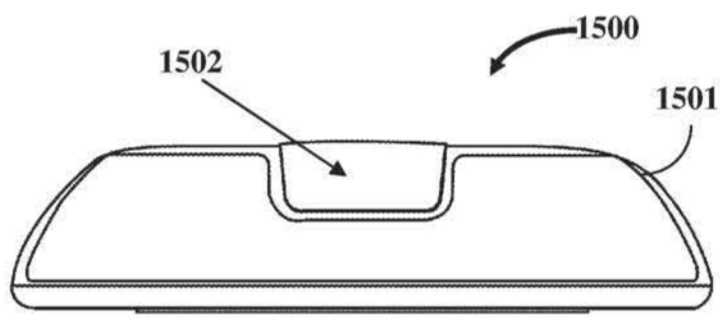


FIG. 15B

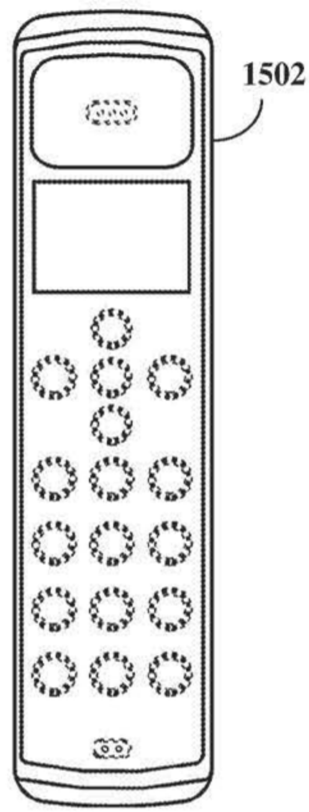


FIG. 15C

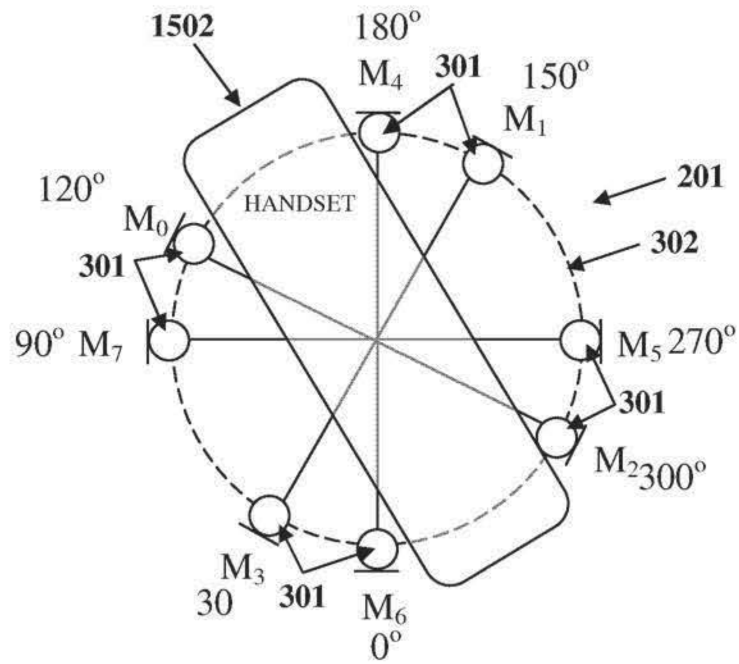


FIG. 16A

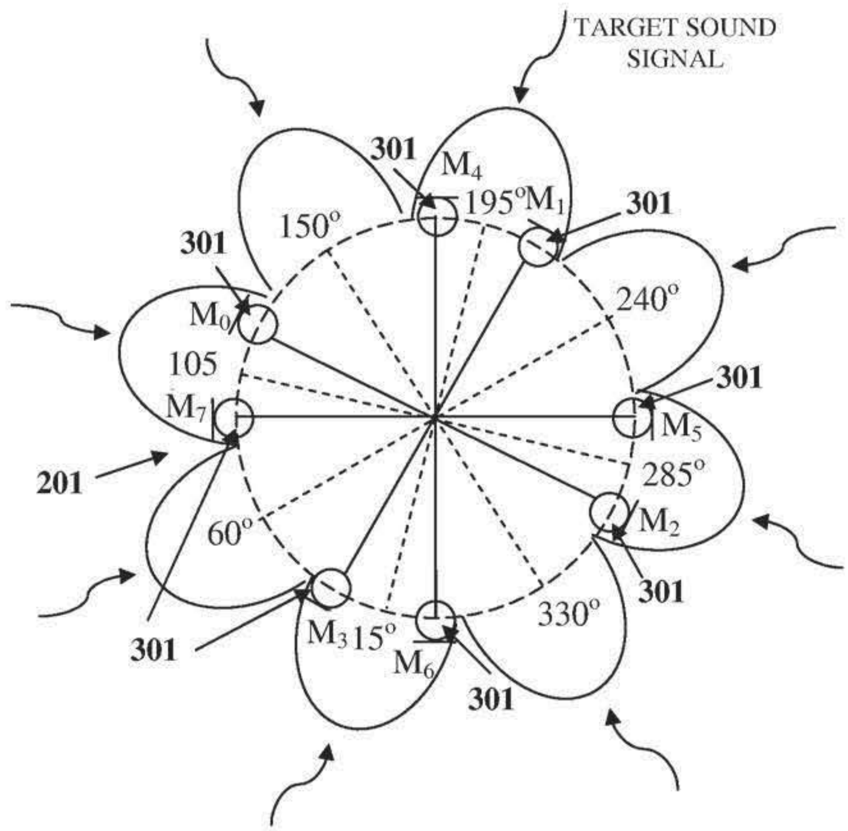


FIG. 16B

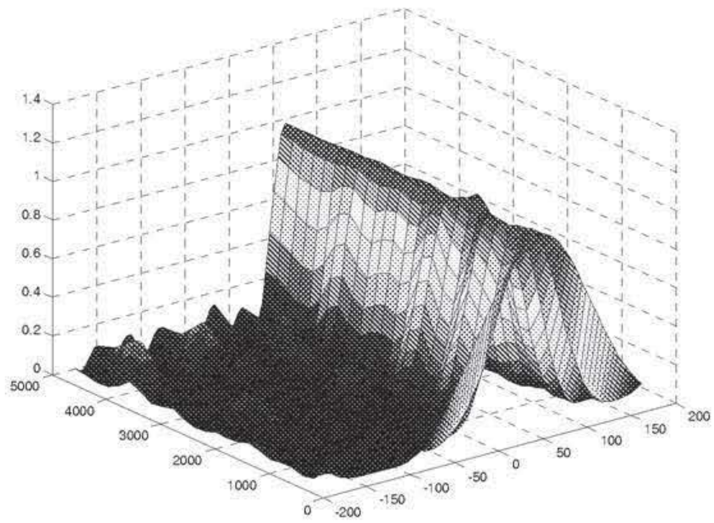


FIG. 16C

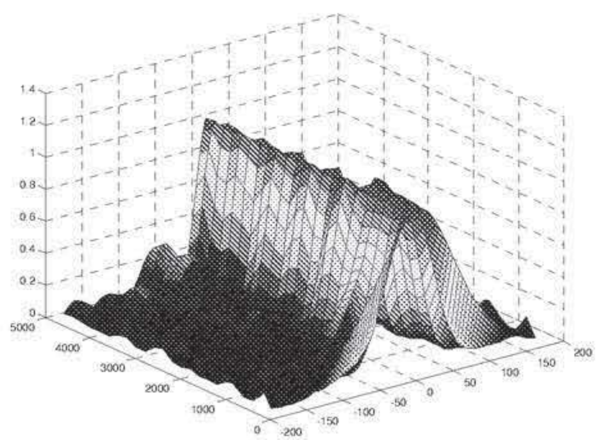


FIG. 16D

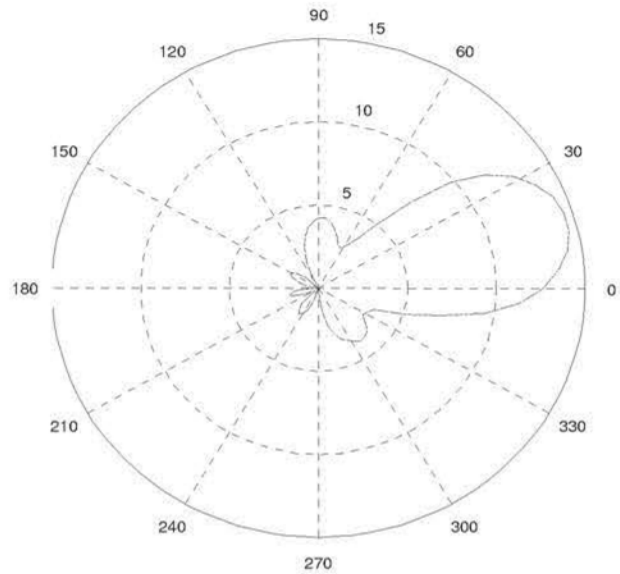


FIG. 16E

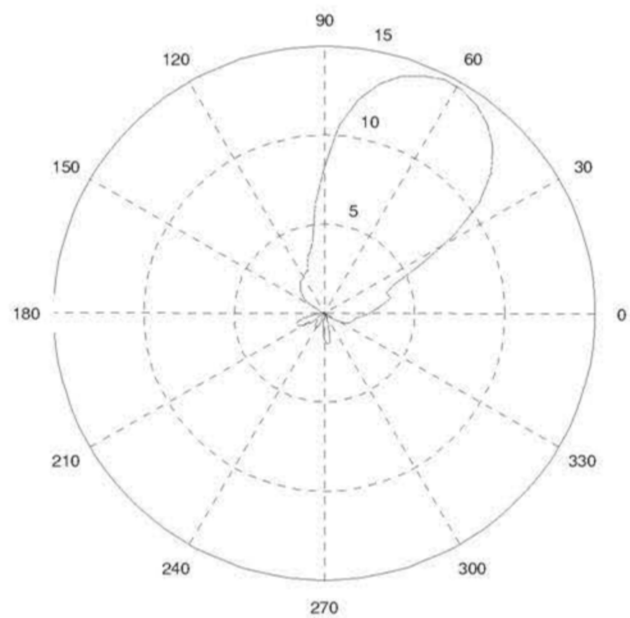


FIG. 16F

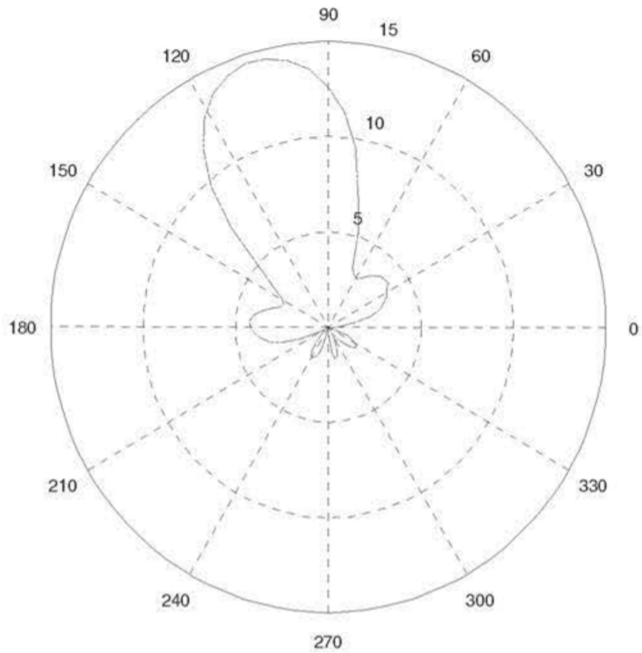


FIG. 16G

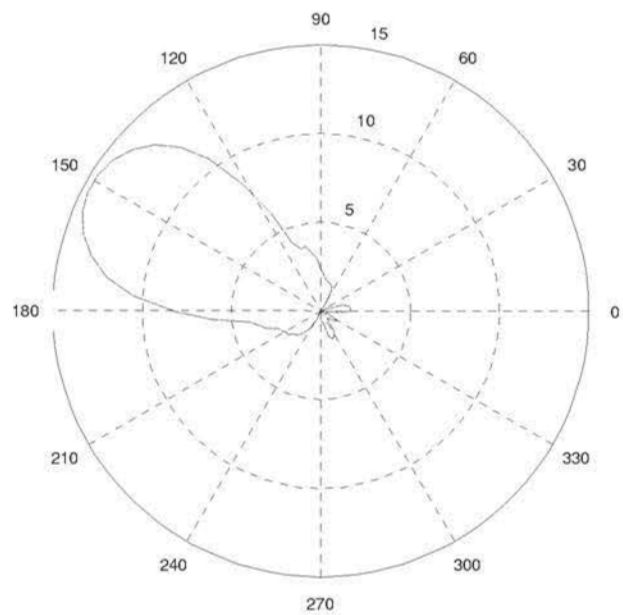


FIG. 16H

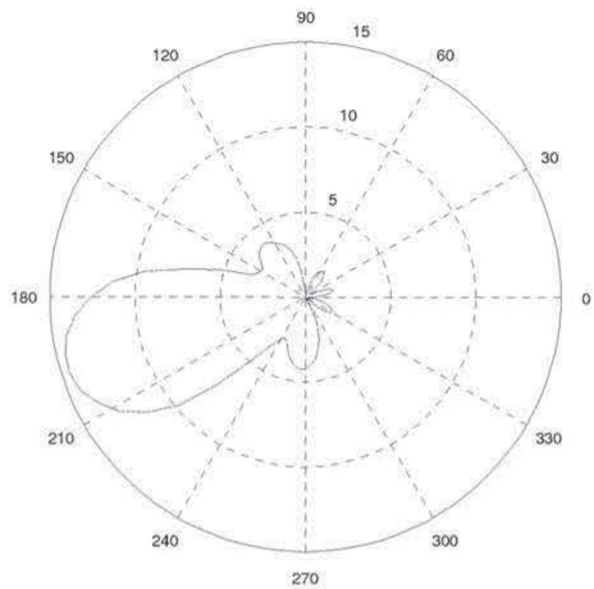


FIG. 16I

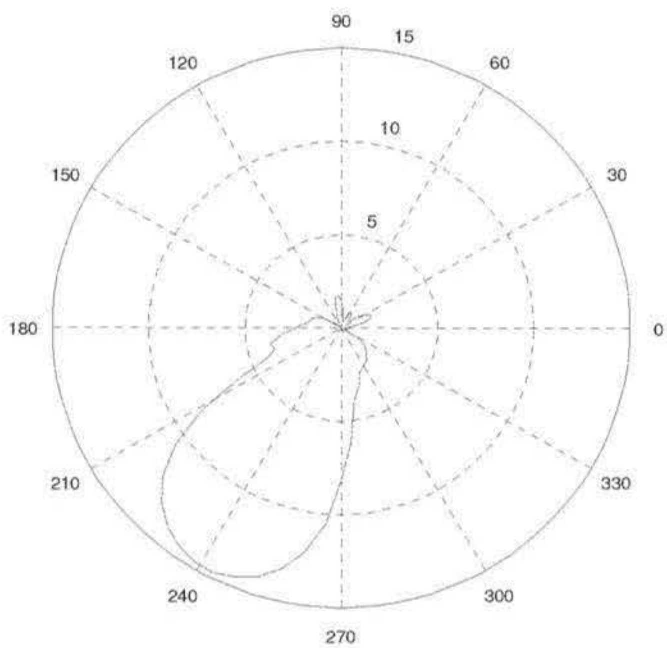


FIG. 16J

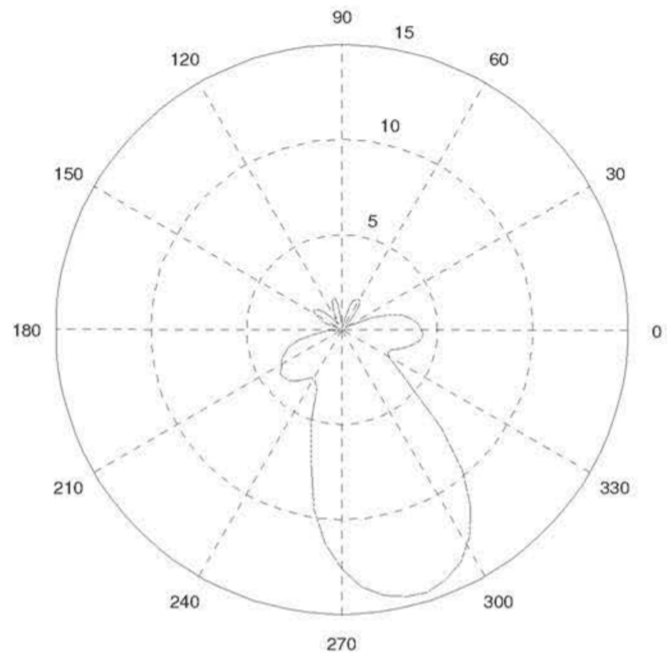


FIG. 16K

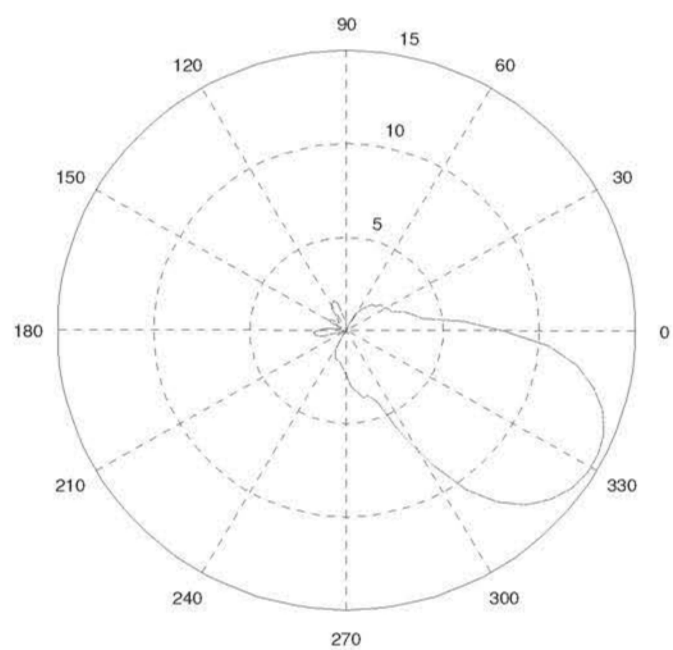


FIG. 16L

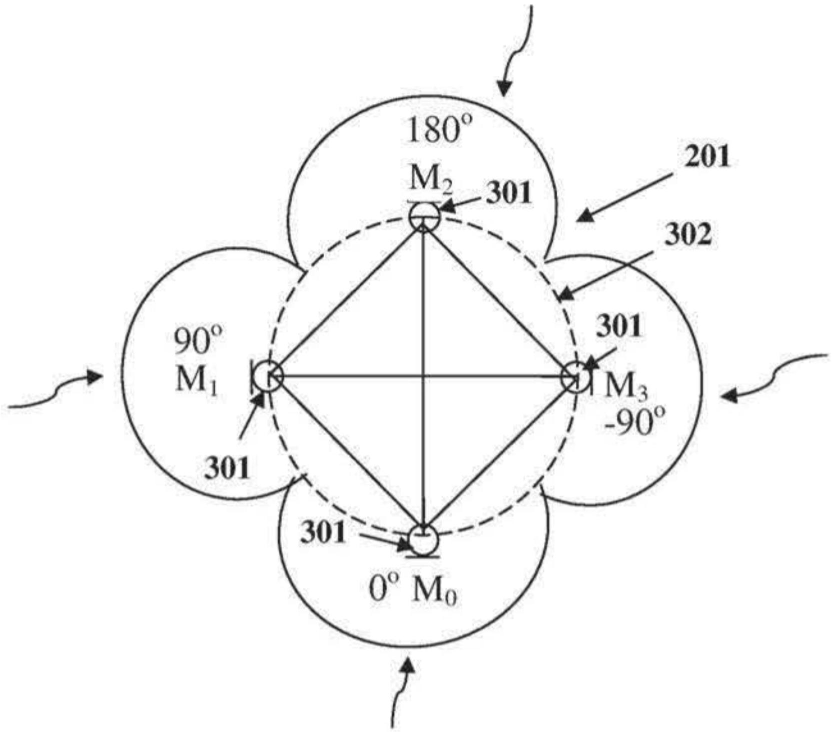


FIG. 17A

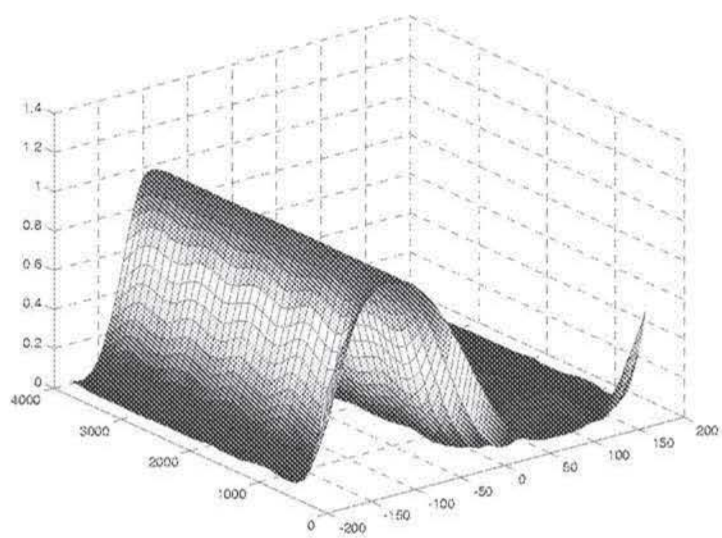


FIG. 17B

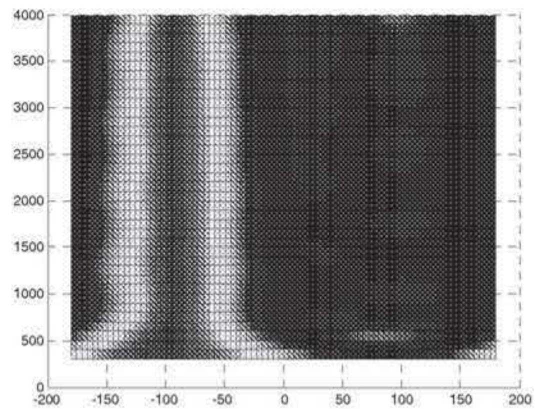


FIG. 17C

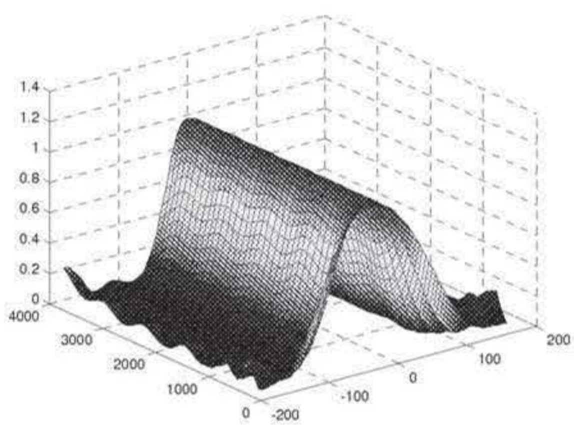


FIG. 17D

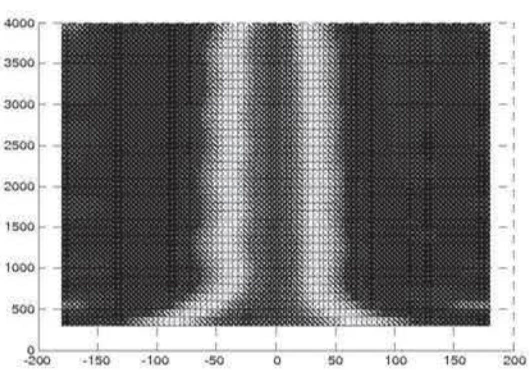


FIG. 17E

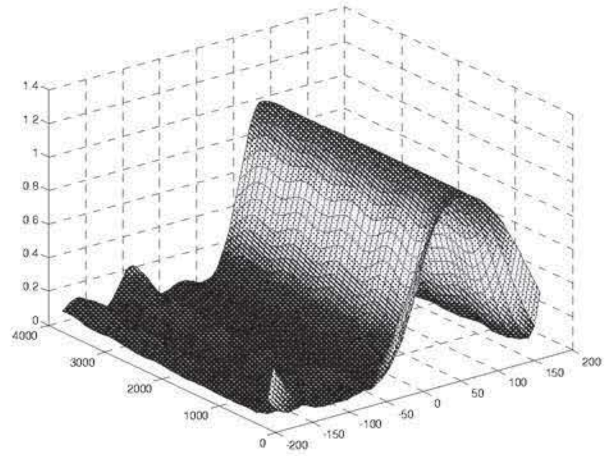


FIG. 17F

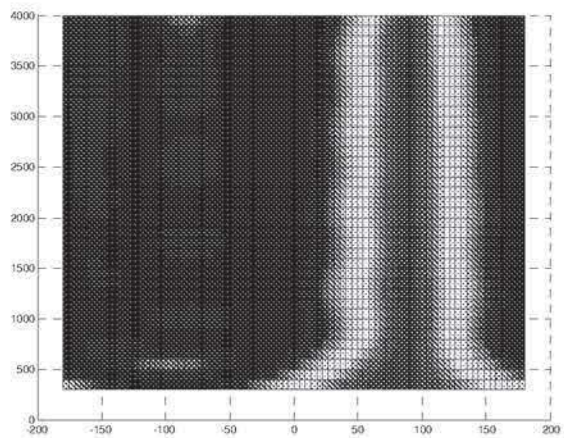


FIG. 17G

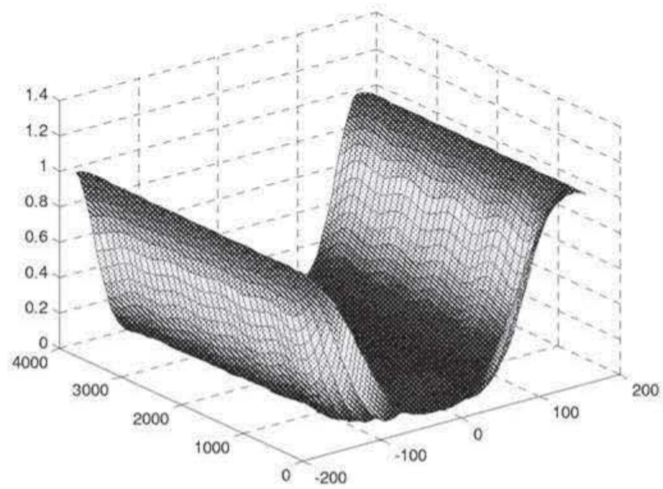


FIG. 17H

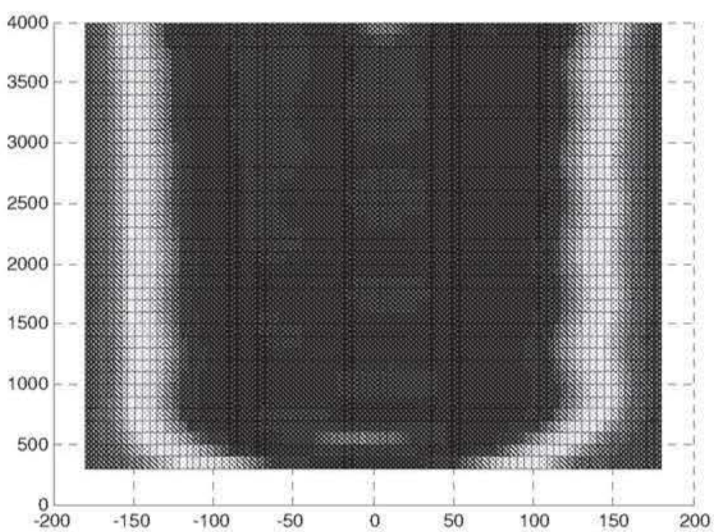


FIG. 17I

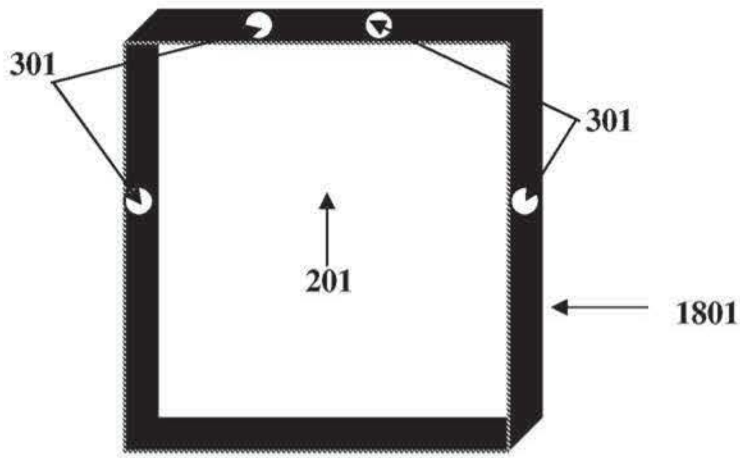


FIG. 18A

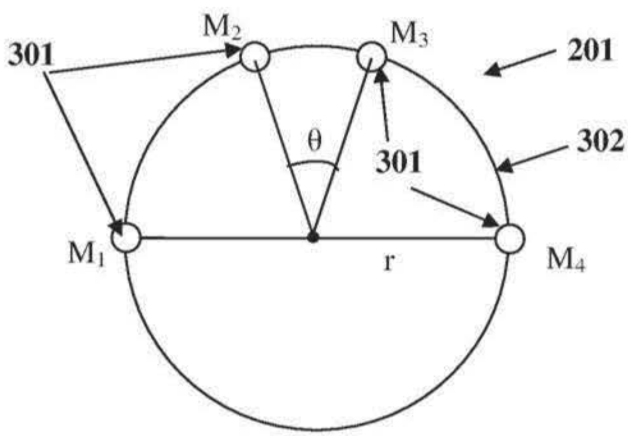


FIG. 18B

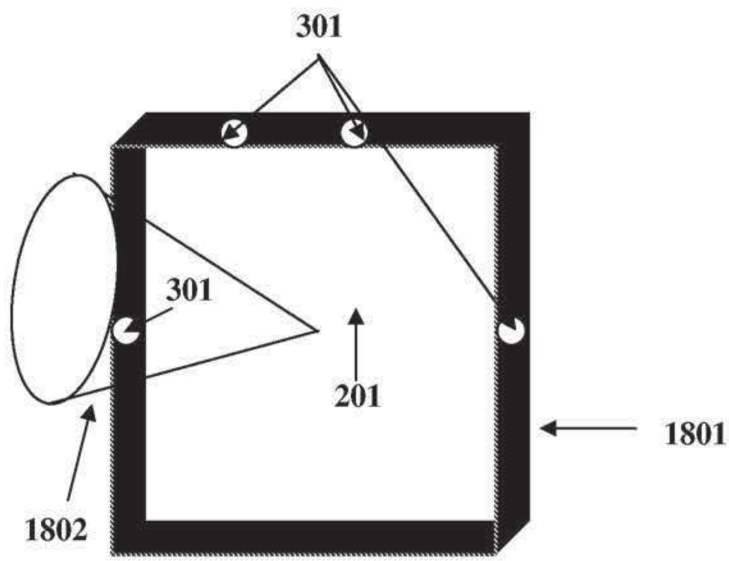


FIG. 18C

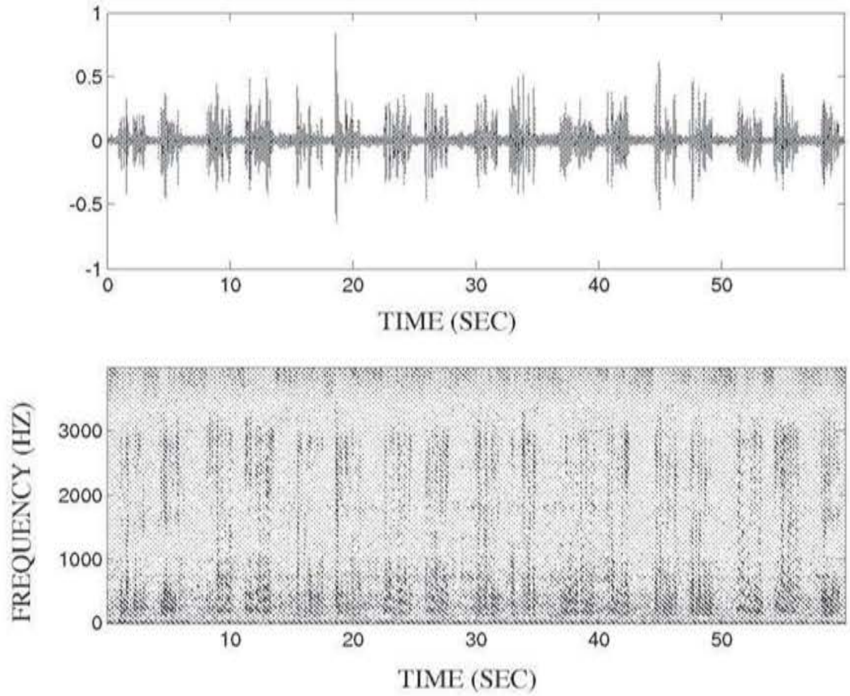


FIG. 18D

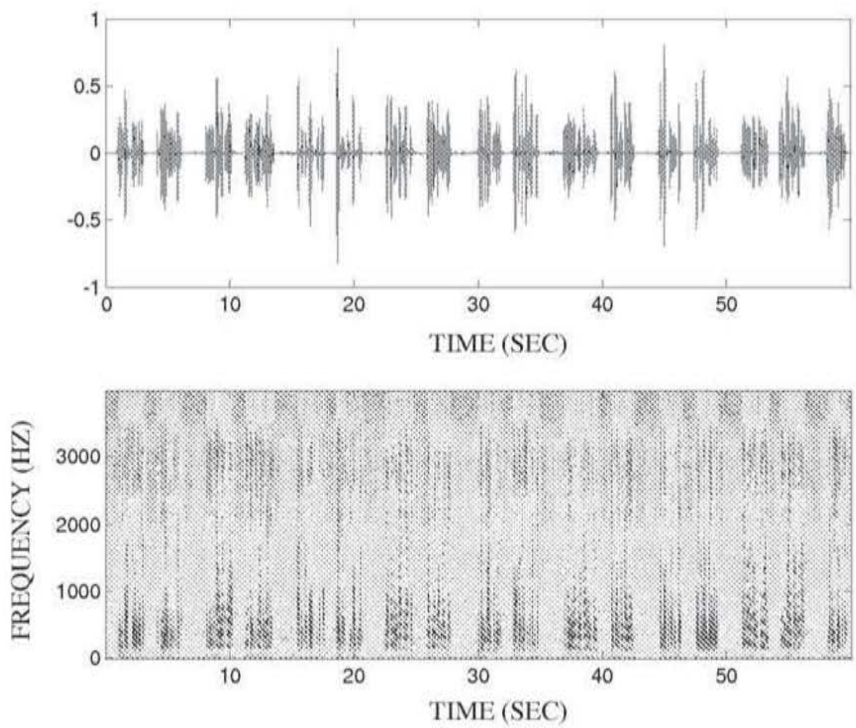


FIG. 18E

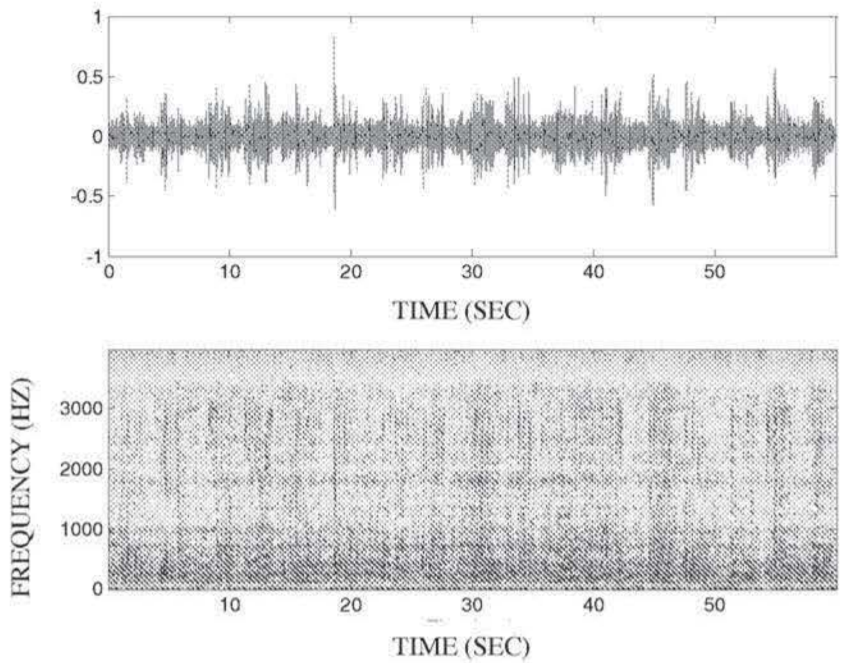


FIG. 18F

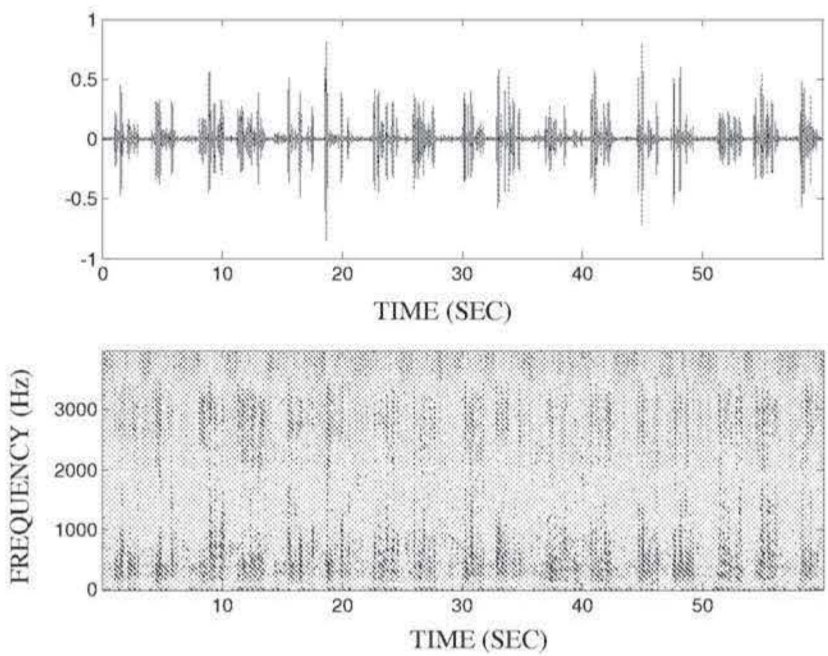
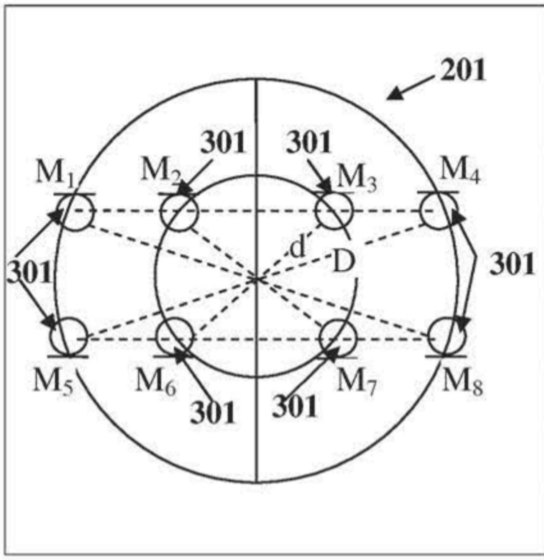
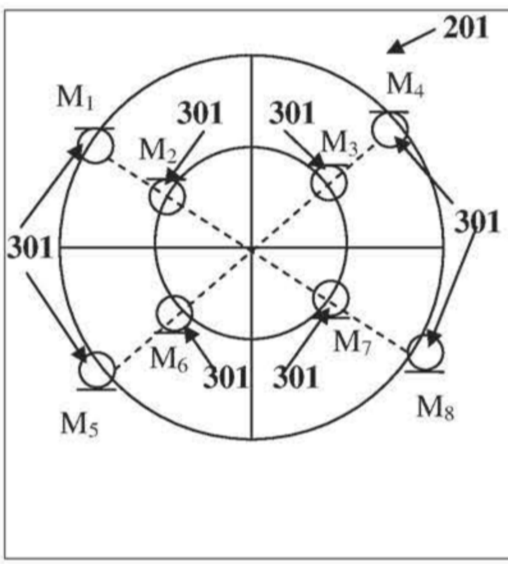


FIG. 18G



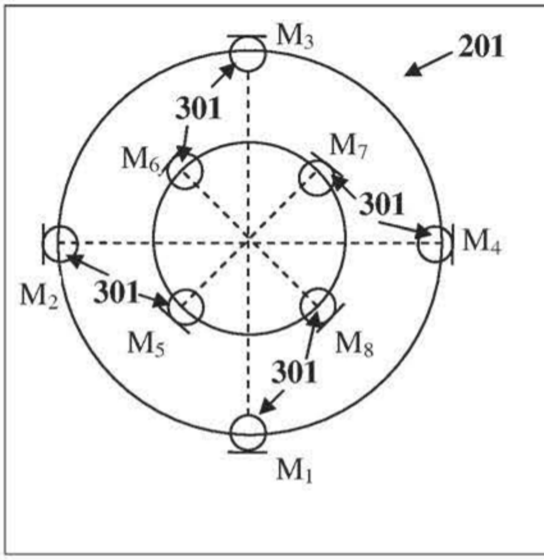
$\tau_1=D*\cos(\theta+\Phi_1)$
$\tau_2=d*\cos(\theta+\Phi_2)$
$\tau_3=d*\cos(\theta-\Phi_3)$
$\tau_4=D*\cos(\theta-\Phi_4)$
$\tau_5=-D*\cos(\theta-\Phi_5)$
$\tau_6=-d*\cos(\theta-\Phi_6)$
$\tau_7=-d*\cos(\theta+\Phi_7)$
$\tau_8=-D*\cos(\theta+\Phi_8)$

FIG. 19A



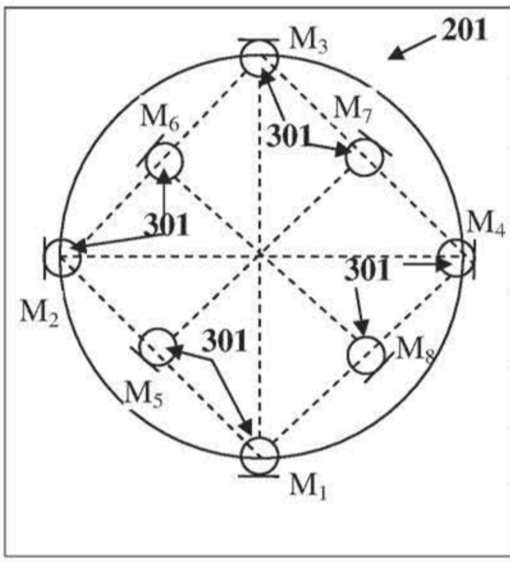
$\tau_1=D*\cos(\theta+\Phi_{12})$
$\tau_2=d*\cos(\theta+\Phi_{12})$
$\tau_3=d*\cos(\theta-\Phi_{34})$
$\tau_4=D*\cos(\theta-\Phi_{34})$
$\tau_5=-D*\cos(\theta-\Phi_{56})$
$\tau_6=-d*\cos(\theta-\Phi_{56})$
$\tau_7=-d*\cos(\theta+\Phi_{78})$
$\tau_8=-D*\cos(\theta+\Phi_{78})$

FIG. 19B



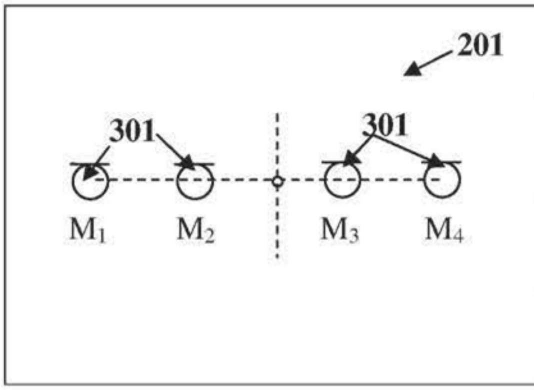
τ_1=-D*cos(θ)
τ_2=-D*sin(θ)
τ_3=D*cos(θ)
τ_4=D*sin(θ)
τ_5=-d*cos(θ-Φ_5)
τ_6=d*cos(θ+Φ_6)
τ_7=d*cos(θ-Φ_7)
τ_8=-d*cos(θ+Φ_8)

FIG. 19C



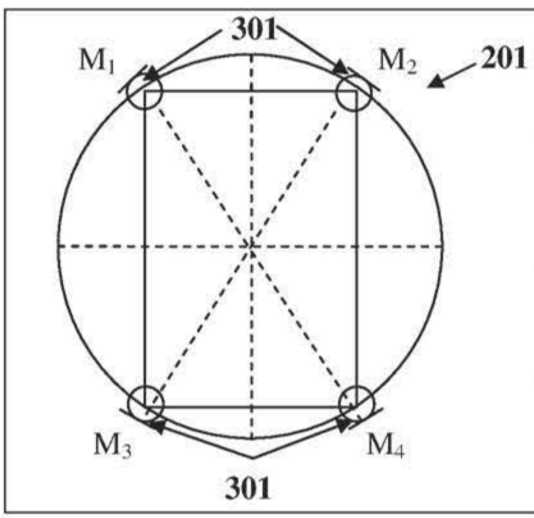
τ_1=-D*cos(θ)
τ_2=-D*sin(θ)
τ_3=D*cos(θ)
τ_4=D*sin(θ)
τ_5=-d*cos(θ-Φ_5)
τ_6=d*cos(θ+Φ_6)
τ_7=d*cos(θ-Φ_7)
τ_8=-d*cos(θ+Φ_8)

FIG. 19D



$\tau_1=-d1*\sin(\theta)$
$\tau_2=-d2*\sin(\theta)$
$\tau_3=d3*\cos(\theta)$
$\tau_4=d4*\cos(\theta)$

FIG. 19E



$\tau_1=d*\cos(\theta+\Phi_1)$
$\tau_2=d*\cos(\theta-\Phi_2)$
$\tau_3=-d*\cos(\theta-\Phi_3)$
$\tau_4=-d*\cos(\theta+\Phi_4)$

FIG. 19F