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NOTICE OF ALLOWANCE AND FEE(S) DUE

64188 7590 05/02/2018
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

EXAMINER	
ESCALANTE, OVIDIO	
ART UNIT	PAPER NUMBER
3992	

DATE MAILED: 05/02/2018

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
15/293,626	10/14/2016	Manli Zhu	CreativeTech_01RE_US	4199

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	$500	$0	$0	$500	08/02/2018

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: Maintenance fees are due in utility patents issuing on applications filed on or after Dec. 12, 1980. It is patentee's responsibility to ensure timely payment of maintenance fees when due. More information is available at www.uspto.gov/PatentMaintenanceFees.

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
or Fax (571)-273-2885

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)

64188 7590 05/02/2018
ASHOK TANKHA
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	$500	$0	$0	$500	08/02/2018

EXAMINER	ART UNIT	CLASS-SUBCLASS
ESCALANTE, OVIDIO	3992	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 Extension or Adjustment under 35 U.S.C. 154 (b)

A reissue patent is for "the unexpired part of the term of the original patent." See 35 U.S.C. 251. Accordingly, the above-identified reissue application is not eligible for Patent Term Extension or Adjustment under 35 U.S.C. 154(b).

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. 15/293,626	Applicant(s) ZHU ET AL.	
	Examiner OVIDIO ESCALANTE	Art Unit 3992	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 *4/18/18*.
   ☐ 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 *1-35*. 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 _____.

/Ovidio Escalante/
Primary Examiner
Art Unit: 3992

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

DETAILED ACTION

Reissue Applications

2. For reissue applications filed before September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the law and rules in effect on September 15, 2012. Where specifically designated, these are "pre-AIA" provisions.

For reissue applications filed on or after September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the current provisions.

3. Applicant is reminded of the continuing obligation under 37 CFR 1.178(b), to timely apprise the Office of any prior or concurrent proceeding in which Patent No. 8,861,756 is or was involved. These proceedings would include interferences, reissues, reexaminations, and litigation.

Applicant is further reminded of the continuing obligation under 37 CFR 1.56, to timely apprise the Office of any information which is material to patentability of the claims under consideration in this reissue application.

These obligations rest with each individual associated with the filing and prosecution of this application for reissue. See also MPEP §§ 1404, 1442.01 and 1442.04.

Response to Arguments

The examiner acknowledges the applicant's corrected reissue declaration. The corrected declaration addresses the issues set forth in the previous office action. Therefore, the rejection to the claims under 35 U.S.C. 251 will be withdrawn.

The examiner also acknowledges applicant's corrected claim formatting. The revised claim formatting is acceptable.

Allowable Subject Matter

4. Claims 1-35 are allowed.

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

Tashev in view of Florencio or Zhan

The Applicant argues Tashev does not teach or suggest a method for determining the delay (τ) 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. Furthermore, Tashev does not teach calculation of time delay as shown in FIG. 5 and Tables 6A, 6B and 7B of applicant's original application. Furthermore, Tashev does not teach or suggest that the distance between each of the sound sensors in the microphone array and the origin corresponds to the time taken for the target sound signal to travel the distance between each of the sound sensors and the origin and is measured by the number of samples within that time period.

In addition, the applicant argues claim 1 recites that the delay is determined in a 2D plane. In contrast, Tashev uses distance in his calculations only in the 3D plane. In the 2D plane, Tashev only uses direction and elevation in his calculations; see Tashev paragraph [0005], which discloses: "This location can be defined in terms of one angle (localization in one dimension), two angles (direction and elevation—localization in 2D) or a full 3D localization (i.e., direction, elevation and distance).").

The examiner acknowledges that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. The examiner notes that Teshav does disclose that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

As set forth in the previous office action, Zhan is directed to a method for controlling sound focusing (see the abstract). With reference to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point). . Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

The applicant states Zhen does not teach or suggest that the delay (τ) between each of the sound sensors and the origin of the microphone array is determined as a function of distance (d) between each of the sound sensors and the origin, a predefined angle (ϕ) between each of the sound sensors and a first reference axis (Y), an elevation angle ('T) 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. Furthermore, Zhan does not teach or suggest a method to determine the delay (i) using 26 the azimuth angle (0), where the delay (r) enables beamforming for multiple number of sound sensors distributed not only in linear but also circular or other layout configurations.

Upon further review the examiner agrees that Zhen does not teach or suggest that the delay (τ) between each of the sound sensors and the origin of the microphone array is determined as a function of distance (d) between each of the sound sensors and the origin, a predefined angle (ϕ) between each of the sound sensors and a first reference axis (Y), an elevation angle ('T) 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 examiner notes that Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

The applicant states Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the uniformly-distributed circular microphone array are used to obtain functions comprising distance, azimuth and elevation data. Florencio does not teach or suggest a method to determine the delay (τ) using the obtained functions comprising distance, azimuth and elevation data, where the delay (τ) enables beamforming for multiple numbers of sound sensors distributed not only in linear but also circular or other layout configurations.

The applicant states that no reference or combination of references teach or suggest the integration of the sound source localization unit 202, the adaptive beamforming unit 203, and the noise reduction unit 207 into a digital signal processor (DSP 1403); see FIG. 14 and paragraph [0090] of applicant's original application, which teaches: "The DSP 1403 implements the sound source localization unit 202, the adaptive beamforming unit 203, and the noise reduction unit 207."

In view of the above comments, the examiner finds the applicant’s arguments persuasive and will withdraw the rejections to the 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

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Ovidio Escalante whose telephone number is (571)272-7537. The examiner can normally be reached on Monday to Friday - 6:00 AM to 2:30 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Michael Fuelling can be reached on (571) 270-1367. The fax phone number for the organization where this application or proceeding is assigned is 571-273-9000. 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.

/Ovidio Escalante/
Ovidio Escalante
Reexamination Specialist
Central Reexamination Unit - Art Unit 3992
(571) 272-7537
Conferees:

/Majid Banankhah/

/M. F./

Supervisory Patent Examiner, Art Unit 3992

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re. application of:
Manli Zhu
Application No.: 15/293,626
Filed: 10/14/2016
Applicant: Li Creative Technologies, Inc.
Title: Microphone Array System

Examiner: Escalante, Ovidio
Art Unit: 3992
Atty. Docket No: CreativeTech_01RE_US

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

Response under "After Final Consideration Pilot Program 2.0"

Examiner Escalante:

In response to the final office action mailed 27 February 2018, please amend the above-referenced application as follows:

Amendments to the Claims begin on page 2 of this response.

Remarks begin on page 21 of this response.

Amendments to the Claims

Claim 1 (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] a linear, circular, or other 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 integrated in a digital signal processor, and 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 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 array of sound sensors [and] in 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 2 (original): 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.

Claim 3 (original): 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 4 (original): The method of claim 3, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 5 (original): The method of claim 3, 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 6 (original): The method of claim 3, 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 7 (original): 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 8 (original): 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 9 (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] <u>a linear, circular, or other</u> 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;

<u>a digital signal processor, said digital signal processor comprising:</u>

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by 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 <u>array of</u> sound sensors [and] <u>in</u> 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.

Claim 10 (original): The system of claim 9, 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 11 (original): The system of claim 9, 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 12 (original): The system of claim 11, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 13 (original): The system of claim 11, 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 14 (original): The system of claim 9, 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 15 (original): The system of claim 9, 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 16 (original): The system of claim 9, 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 17 (original): The system of claim 9, 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 18 (original): The system of claim 9, 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 19 (Amended): The method of claim 1, wherein said delay (τ) is [determined by a formula $\tau = fs*t$, wherein fs is a sampling frequency and t is a time delay] calculated based on said number of samples within a time period and a time delay for said target sound signal to travel said distance between each of said sound sensors in said microphone array and said origin of said array of said sound sensors, and wherein said distance between said each of said sound sensors in the microphone array and said origin of said array of said sound sensors can be same or different.

Claim 20 (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] a linear, circular, or other 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 integrated in a digital signal processor, and 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 array of sound sensors [and] in 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 21 (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] a linear, circular, or other 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by 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 array of sound sensors [and] in 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.

Claim 22 (New, 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, a sound source localization unit, a beamforming unit, and a noise reduction unit, wherein said sound source localization unit, said beamforming unit, and said noise reduction unit are integrated in a digital signal processor, and wherein said sound source localization unit, said 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more 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 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 beamforming unit, wherein said 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, amended): The method of claim 22, wherein said beamforming comprises:

providing a fixed beamformer, a blocking matrix, and an adaptive filter in said 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 24 (New, amended): The method of claim 23, wherein said beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said 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 25 (New, amended): The method of claim 22, 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 beamforming unit for sub-band adaptive beamforming.

Claim 26 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a 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 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 27 (New, amended): The system of claim 26, wherein said 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 28 (New, amended): The system of claim 26, wherein said noise reduction unit performs noise reduction in a plurality of frequency sub-bands employed by an analysis filter bank of said beamforming unit for sub-band adaptive beamforming.

Claim 29 (New, amended): The system of claim 26, wherein said array of said sound sensors is one of a linear array of said sound sensors, and a circular array of said sound sensors, and other types of array of said sound sensors.

Claim 30 (New, 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, a sound source localization unit, a beamforming unit, and a noise reduction unit, wherein said sound source localization unit, said beamforming unit, and said noise reduction unit are integrated in a digital signal processor, and wherein said sound source localization unit, said 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more 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 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 beamforming unit, wherein said 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 31 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a 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 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 32 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis;

a beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals.

Claim 33 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals.

Claim 34 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals.

Claim 35 (New, amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors positioned in a non-circular 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis, wherein said distance between each of said sound sensors and said reference point varies from a minimum value to a maximum value, and wherein said minimum value corresponds to zero and said maximum value is defined based on a limitation associated with size of said system;

a beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reduction unit that suppresses said ambient noise signals.

Remarks

The Pending Claims

Claims 1-35 are currently pending. Reconsideration and allowance of the pending claims is respectfully requested.

Summary of the office action

The reissue oath/declaration filed with this application is defective (see 37 CFR 1.175 and MPEP § 1414).

Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C. 251 as set forth above. See 37 CFR 1.175.

Claim Amendments

Claims 1, 9 and 19-21are amended; claims 2-8 and 10-18 remain as originally presented; claims 22-35 are new, amended.

Response to the rejections

The office action states: "**The reissue oath/declaration filed with this application is defective (see 37 CFR 1.175 and MPEP § 1414).**"

In response to the above rejection, application has filed a new reissue declaration with an attached sheet. The attached sheet identifies the location of the errors, how the errors are rectified and the reason for the claim amendment. Applicant therefore respectfully requests reconsideration and acceptance of the new reissue declaration.

The office action further states: "**Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C. 251 as set forth above. See 37 CFR 1.175.**"

In response to the above rejection, application has filed a new reissue declaration. Furthermore, applicant has amended the claims as per the requirements of a reissue application. Applicant therefore respectfully requests that the rejection of claims 1-35 under 35 U.S.C 251 be reconsidered and withdrawn.

Conclusion

Applicant appreciates examiner's finding that applicant's response overcomes the cited prior arts. Applicant also appreciates examiner's suggestion for overcoming the defects in the reissue declaration and formatting of the amended claims. Applicant has followed the suggestions in the office action for overcoming the defects in the reissue declaration. Applicant has further amended the claims with proper formatting as suggested in the office action. Applicant therefore respectfully requests that a timely notice of allowance be issued in this case. In the interest of compact prosecution, if a claim may be made potentially allowable by an Examiner's amendment, Examiner Escalante is requested to call the undersigned with the proposed amendment.

Respectfully submitted,

Date: April 18, 2018

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
15/293,626	10/14/2016	Manli Zhu	CreativeTech_01RE_US	4199

64188 7590 02/27/2018

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EXAMINER
ESCALANTE, OVIDIO

ART UNIT	PAPER NUMBER
3992	

NOTIFICATION DATE	DELIVERY MODE
02/27/2018	ELECTRONIC

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.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

ASH@IPPROCUREMENT.COM
prosecution@ipprocurement.com

Office Action Summary	Application No. 15/293,626	Applicant(s) ZHU ET AL.	
	Examiner OVIDIO ESCALANTE	Art Unit 3992	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 *1/29/18*.
☐ 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) *1-35* 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) *1-35* 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 *10/14/16* 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: _____.

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

DETAILED ACTION

Reissue Applications

2. For reissue applications filed before September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the law and rules in effect on September 15, 2012. Where specifically designated, these are "pre-AIA" provisions.

For reissue applications filed on or after September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the current provisions.

3. Applicant is reminded of the continuing obligation under 37 CFR 1.178(b), to timely apprise the Office of any prior or concurrent proceeding in which Patent No. 8,861,756 is or was involved. These proceedings would include interferences, reissues, reexaminations, and litigation.

Applicant is further reminded of the continuing obligation under 37 CFR 1.56, to timely apprise the Office of any information which is material to patentability of the claims under consideration in this reissue application.

These obligations rest with each individual associated with the filing and prosecution of this application for reissue. See also MPEP §§ 1404, 1442.01 and 1442.04.

Reissue Declaration

4. The reissue oath/declaration filed with this application is defective (see 37 CFR 1.175 and MPEP § 1414) because of the following:

The examiner notes that in accordance with MPEP 1414, any error in the claims must be identified by reference to the specific claim(s) and the specific claim language wherein lies the error.

The examiner notes that the reissue declaration recites the following reasons:

> Independent claims 1, 9, 20-22, 26 and 30-35 have been amended for more clarity.
>
> 1. Changing “arbitrary configuration” to “a linear, circular, or other configuration”
>
> 2. Integration of sound source localization unit, adaptive beamforming unit and noise reduction unit within a digital signal processor is recited in the independent claims.

The examiner notes this statement does not identify an error. The statement states that the claims have been amended "for more clarity". As a suggestion, the examiner notes that since the application has been identified as a broadening reissue, applicant can point out which limitation in the original claims they seek to broaden in relation to the new claims.

In addition, for a broadening reissue, the declaration must be by made by the inventors. The corrected declaration was only signed by the assignee.

In addition, the applicant has stated that the original patent was filed under 37 CFR 1.46 by the assignee. It is noted that this does not apply since the original patent application was not filed on or after September 16, 2012.

5. Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C. 251 as set forth above. See 37 CFR 1.175.

The nature of the defect(s) in the declaration is set forth in the discussion above in this Office action.

Reissue Amendment

The amendment filed January 29, 2018 proposes amendments to the claims that do not comply with 37 CFR 1.173(b), which sets forth the manner of making amendments in reissue applications.

As set forth therein, any changes relative to the patent being reissued must included the following markings:

(1) The matter to be omitted by reissue must be enclosed in brackets; and

(2) The matter to be added by reissue must be underlined.

In this case, new claims 22-35 are not entirely underlined and the original claims (claims 1-21) do not show matter to be omitted by brackets (the omissions are noted by strikethroughs).

Response to Arguments

Tashev in view of Florencio or Zhan

The Applicant argues Tashev does not teach or suggest a method for determining the delay (τ) 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. Furthermore, Tashev does not teach calculation of time delay as shown in FIG. 5 and Tables 6A, 6B and 7B of applicant's original application. Furthermore, Tashev does not teach or suggest that the distance between each of the sound sensors in the microphone array and the origin corresponds to the time taken for the target sound signal to travel the distance between each of the sound sensors and the origin and is measured by the number of samples within that time period.

In addition, the applicant argues claim 1 recites that the delay is determined in a 2D plane. In contrast, Tashev uses distance in his calculations only in the 3D plane. In the 2D plane, Tashev only uses direction and elevation in his calculations; see Tashev paragraph [0005], which discloses: "This location can be defined in terms of one angle (localization in one dimension), two angles (direction and elevation—localization in 2D) or a full 3D localization (i.e., direction, elevation and distance).").

The examiner acknowledges that Teshav does not specially disclose that the delay is based on a function of distance <u>between each of the sound sensors and a reference point</u>. The examiner notes that Teshav does disclose that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

As set forth in the previous office action, Zhan is directed to a method for controlling sound focusing (see the abstract). With reference to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point). . Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

The applicant states Zhen does not teach or suggest that the delay (τ) between each of the sound sensors and the origin of the microphone array is determined as a function of distance (d) between each of the sound sensors and the origin, a predefined angle (ϕ) between each of the sound sensors and a first reference axis (Y), an elevation angle ('T) 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. Furthermore, Zhan does not teach or suggest a method to determine the delay (i) using 26 the azimuth angle (0), where the delay (r) enables beamforming for multiple number of sound sensors distributed not only in linear but also circular or other layout configurations.

Upon further review the examiner agrees that Zhen does not teach or suggest that the delay (τ) between each of the sound sensors and the origin of the microphone array is determined as a function of distance (d) between each of the sound sensors and the origin, a predefined angle (ϕ) between each of the sound sensors and a first reference axis (Y), an elevation angle ('T) 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 examiner notes that Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

The applicant states Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the uniformly-distributed circular microphone array are used to obtain functions comprising distance, azimuth and elevation data.

Florencio does not teach or suggest a method to determine the delay (τ) using the obtained functions comprising distance, azimuth and elevation data, where the delay (τ) enables beamforming for multiple numbers of sound sensors distributed not only in linear but also circular or other layout configurations.

The applicant states that no reference or combination of references teach or suggest the integration of the sound source localization unit 202, the adaptive beamforming unit 203, and the noise reduction unit 207 into a digital signal processor (DSP 1403); see FIG. 14 and paragraph [0090] of applicant's original application, which teaches: "The DSP 1403 implements the sound source localization unit 202, the adaptive beamforming unit 203, and the noise reduction unit 207."

In view of the above comments, the examiner finds the patent owner's arguments persuasive and will withdraw the rejections to the claims.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Ovidio Escalante whose telephone number is (571)272-7537. The examiner can normally be reached on Monday to Friday - 6:00 AM to 2:30 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Michael Fuelling can be reached on (571) 270-1367. The fax phone number for the organization where this application or proceeding is assigned is 571-273-9000.

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.

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re. application of:
Manli Zhu
Application No.: 15/293,626
Filed: 10/14/2016
Applicant: Li Creative Technologies, Inc.
Title: Microphone Array System

Examiner: Escalante, Ovidio
Art Unit: 3992
Atty. Docket No: CreativeTech_01RE_US

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

Response to non-final office action

Examiner Escalante:

In response to the non-final office action mailed 05 October 2017, please amend the above-referenced application as follows:

Amendments to the Claims begin on page 2 of this response.

Remarks begin on page 21 of this response.

Amendments 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~~ a linear, circular, or other 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 integrated in a digital signal processor, and 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 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 array of sound sensors [[and]] in 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 2 (original): 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.

Claim 3 (original): 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 4 (original): The method of claim 3, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 5 (original): The method of claim 3, 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 6 (original): The method of claim 3, 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 7 (original): 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 8 (original): 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 9 (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~~ <u>a linear, circular, or other</u> 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;

<u>a digital signal processor, said digital signal processor comprising:</u>

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by 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 array of sound sensors [[and]] in 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.

Claim 10 (original): The system of claim 9, 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 11 (original): The system of claim 9, 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 12 (original): The system of claim 11, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

Claim 13 (original): The system of claim 11, 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 14 (original): The system of claim 9, 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 15 (original): The system of claim 9, 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 16 (original): The system of claim 9, 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 17 (original): The system of claim 9, 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 18 (original): The system of claim 9, 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 19 (currently amended): The method of claim 1, wherein said delay (τ) is ~~determined by a formula τ =fs*t, wherein fs is a sampling frequency and t is a time delay~~ <u>calculated based on said number of samples within a time period and a time delay for said target sound signal to travel said distance between each of said sound sensors in said microphone array and said origin of said array of said sound sensors, and wherein said distance between said each of said sound sensors in the microphone array and said origin of said array of said sound sensors can be same or different</u>.

Claim 20 (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~~ <u>a linear, circular, or other</u> 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 integrated in a digital signal processor, and 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 array of sound sensors [[and]] in 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 21 (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~~ a linear, circular, or other 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 digital signal processor, said digital signal processor comprising:

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by 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 array of sound sensors [[and]] in 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.

Claim 22 (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, a sound source localization unit ~~localizer~~, a ~~beamformer~~ beamforming unit, and a noise ~~reducer~~ reduction unit, wherein said sound source localization unit, said beamforming unit, and said noise reduction unit are integrated in a digital signal processor, and wherein said sound source localization unit ~~localizer~~, said ~~beamformer~~ beamforming unit, and said noise ~~reducer~~ 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

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

performing 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 ~~beamformer~~ beamforming unit, wherein said ~~beamformer~~ beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

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

Claim 23 (currently amended): The method of claim 22, wherein said beamforming comprises:

providing a fixed beamformer, a blocking matrix, and an adaptive filter in said ~~beamformer~~ 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 24 (currently amended): The method of claim 23, wherein said beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said ~~beamformer~~ 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 25 (currently amended): The method of claim 22, wherein said noise reducer 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 ~~beamformer~~ beamforming unit for sub-band adaptive beamforming.

Claim 26 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a ~~beamformer~~ 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 ~~beamformer~~ beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

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

Claim 27 (currently amended): The system of claim 26, wherein said ~~beamformer~~ 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 28 (currently amended): The system of claim 26, wherein said noise ~~reducer~~ reduction unit performs noise reduction in a plurality of frequency sub-bands employed by an analysis filter bank of said ~~beamformer~~ beamforming unit for sub-band adaptive beamforming.

Claim 29 (currently amended): The system of claim 26, wherein said array of said sound sensors is one of a linear array of said sound sensors, [[and]] a circular array of said sound sensors, and other types of array of said sound sensors.

Claim 30 (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, a sound source localization unit ~~localizer~~, a ~~beamformer~~ beamforming unit, and a noise ~~reducer~~ reduction unit, wherein said sound source localization unit, said beamforming unit, and said noise reduction unit are integrated in a digital signal processor, and wherein said sound source localization unit ~~localizer~~, said ~~beamformer~~ beamforming unit, and said noise ~~reducer~~ 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

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

performing 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 ~~beamformer~~ beamforming unit, wherein said ~~beamformer~~ beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

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

Claim 31 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a ~~beamformer~~ 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 ~~beamformer~~ beamforming unit enhances said target sound signal and partially suppresses said ambient noise signals; and

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

Claim 32 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis;

a ~~beamformer~~ beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise ~~reducer~~ reduction unit that suppresses said ambient noise signals.

Claim 33 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a ~~beamformer~~ beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise ~~reducer~~ reduction unit that suppresses said ambient noise signals.

Claim 34 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors, 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a ~~beamformer~~ beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise ~~reducer~~ reduction unit that suppresses said ambient noise signals.

Claim 35 (currently amended): A system for enhancing a target sound signal from a plurality of sound signals, comprising:

an array of sound sensors positioned in a non-circular 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 digital signal processor, said digital signal processor comprising:

a sound source ~~localizer~~ localization unit that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis, wherein said distance between each of said sound sensors and said reference point varies from a minimum value to a maximum value, and wherein said minimum value corresponds to zero and said maximum value is defined based on a limitation associated with size of said system;

a ~~beamformer~~ beamforming unit that enhances said target sound signal and suppresses said ambient noise signals; and

a noise ~~reducer~~ reduction unit that suppresses said ambient noise signals.

Remarks

The Pending Claims

Claims 1-35 are currently pending. Reconsideration and allowance of the pending claims is respectfully requested.

Summary of the office action

Defective Reissue Declaration

Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C 251.

35 USC § 112 Sixth Paragraph

The office action states that claims 9, 21, 23, 31-35 recite phrases that invoke 35 U.S.C. § 112, 6th paragraph.

Claim Rejections - 35 USC § 103

Claims 1, 2, 4, 7, 9, 10, 12, 15, 18, 20-22, 24, 29-35 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010)1. (U.S. Patent Publication 2011/0135125).

Claim 16 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Marash U.S. Patent 6,198,693.

Claims 8, 17, 25 and 28 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125) and further in view of Nemer U.S. Patent Pub. 2011/0096915.

Claims 3, 6, 11, 14, 23, 24 and 27 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Tashev et al. U.S. Patent Pub. 2008/0232607 (hereinafter Tashev '607).

Claim Amendments

Claims 1, 9, 19-29 and 30-35 are currently amended; claims 2-8 and 10-18 remain as originally presented; claim 29 remains as previously presented.

Support for the amendment: "*providing a microphone array system comprising an array of sound sensors positioned in a linear, circular, or other configuration*" in claim 1 is found in paragraph [0061] of applicant's original application.

Support for the amendment: "*wherein said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are integrated in a digital signal processor*" in claim 1 is found in **FIG. 14** and paragraph [0090] of applicant's original application.

Support for amended claim 19 is found in paragraph [0063] of applicant's original application.

Support for amended claim 29 is found in paragraph [0061] of applicant's original application.

Applicant submits that the claim amendments do not add any new subject matter.

Response to the rejections

The office action states: "Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C 251."

In response to the above rejection, application has filed a new reissue declaration, and requests that the rejection of claims 1-35 under 35 U.S.C 251 be reconsidered and withdrawn.

The office action further states: "**The office action states that claims 9, 21, 23, 31-35 recite phrases that invoke 35 U.S.C. § 112, 6th paragraph.**"

In response to the above rejection, applicant submits that amended claims 9, 21 and 31-35 clearly recite the hardware structure of the invention. Furthermore, **FIG. 14** exemplarily illustrates a hardware implementation of the microphone array system **200** recited in claims 9, 21 and 31-35. Furthermore, the microphone array system **200** is disclosed in the detailed description of **FIG. 2**. The hardware implementation comprises the microphone array **201** having a number of sound sensors **301** positioned in a linear, circular, or other 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 audio codec **1402** receives the amplified output from the microphone amplifiers **1401**. 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** (see **FIG. 14** and paragraphs [0088]-[0095] of applicant's original application). Furthermore, 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**. Furthermore, the DSP **1403** is programmed for beamforming, noise reduction, echo cancellation, and USB interfacing according to the method recited in the claims, and fine tuned for optimal performance. Therefore, the drawings and the specification clearly disclose the structure of the hardware elements forming the system recited in the claims.

Applicant therefore respectfully requests that the rejection of claims 9, 21 and 31-35 under 35 U.S.C. § 112, 6th paragraph be reconsidered and withdrawn.

Claim 23 is dependent on claim 21. Applicant therefore requests that the rejection of claim 23 under 35 U.S.C. § 112, 6th paragraph be reconsidered and withdrawn.

The office action further states: **"Claims 1, 2, 4, 7, 9, 10, 12, 15, 18, 20-22, 24, 29-35 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010). (U.S. Patent Publication 2011/0135125)."**

In response to the above rejection, applicant submits that Tashev, in view Florencio or Zhan, does not teach or suggest all the limitations in applicant's claim 1.

Claim 1 recites 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 said array of sound sensors in a plurality of configurations;"

In paragraph [0062] of applicant's original application, applicant teaches that 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 distance between each of the sound sensors in the microphone array and the origin can be same (see FIGS. 16A, 16B and 18 B), or different (see FIGS. 19A and 19B). The claimed method is applicable for both cases. The determined delay (τ) is represented in terms of number of samples; see paragraph [0063], which discloses: "*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 in the microphone array and the origin 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.

In contrast, Tashev discloses, *inter alia*, a system and process for sound source localization, by calculating the energy of each frame set of the microphone signal in the sequence they were captured. This energy value is used for both noise floor tracking and frame classification. Thus, the frame set passing the minimum energy threshold test is subjected to the beamsteering procedure. This involves computing the full spectrum energy for each of a prescribed number of directions. After finding the energy as a function of the direction angle, the direction exhibiting the maximum energy and a prescribed number of its neighboring (i.e., adjacent) search directions are interpolated. The result of the interpolation process is then designated as the direction identifying the location of the sound source; see Tashev paragraphs [0072]-[0074].

Tashev does not teach or suggest a method for determining the delay (τ) 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. Furthermore, Tashev does not teach calculation of time delay as shown in **FIG. 5** and **Tables 6A, 6B and 7B** of applicant's original application. Furthermore, Tashev does not teach or suggest that the distance between each of the sound sensors in the microphone array and the origin corresponds to the time taken for the target sound signal to travel the distance between each of the sound sensors and the origin and is measured by the number of samples within that time period.

Furthermore, claim 1 recites that the delay is determined in a 2D plane. In contrast, Tashev uses distance in his calculations only in the 3D plane. In the 2D plane, Tashev only uses direction and elevation in his calculations; see Tashev paragraph [0005], which discloses: "*This location can be defined in terms of one angle (localization in one dimension), two angles (direction and elevation—localization in 2D) or a full 3D localization (i.e., direction, elevation and distance).*").

Zhen discloses a method for controlling sound focusing, where a sound source locating module computes the position information of a sound source relative to a reference microphone, that is, how to compute the distance and the azimuth θ from the sound source to the reference microphone. The position of a sound source relative to the microphone array computed by the sound source locating module is the position of the sound source relative to the reference microphone, where the reference microphone is in the center of a linearly configured microphone array. Zhen does not teach or suggest that the delay (τ) between each of the sound sensors and the origin of the microphone array is determined as a function of distance (d) between each of the sound sensors and the origin, a predefined angle (Φ) between each of the sound sensors 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. Furthermore, Zhan does not teach or suggest a method to determine the delay (τ) using the azimuth angle (θ), where the delay (τ) enables beamforming for multiple number of sound sensors distributed not only in linear but also circular or other layout configurations.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the uniformly-distributed circular microphone array are used to obtain functions comprising distance, azimuth and elevation data. Florencio does not teach or suggest a method to determine the delay (τ) using the obtained functions comprising distance, azimuth and elevation data, where the delay (τ) enables beamforming for multiple numbers of sound sensors distributed not only in linear but also circular or other layout configurations.

Furthermore, no reference or combination of references teach or suggest the integration of the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207** into a digital signal processor (DSP **1403**); see **FIG. 14** and paragraph [0090] of applicant's original application, which teaches: "*The DSP **1403** implements the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207**.*"

Therefore, even if teaching of Tashev and Zhan or Florencio are combined as suggested in the office action, the combination that results will be unsuccessful in arriving at 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 said array of sound sensors in a plurality of configurations;"

For the reasons presented above, applicant submits that claim 1 is not anticipated by Tashev in view of Florencio or Zhan. Applicant therefore respectfully requests that the rejection of claim 1 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Claims 2, 4 and claim 7 are dependent on claim 1. Since claim 1 is non-obvious over a combination of Tashev and Zhan or Florencio, dependent claims 2, 4 and claim 7 are also non-obvious over the combination of Tashev and Zhan or Florencio. Applicant therefore respectfully requests that the rejection of claims 2, 4 and claim 7 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Claims 9, 20-22, 26 and 30-35 are analogous to claim 1. Since claim 1 is non-obvious over a combination of Tashev and Zhan or Florencio, applicant submits that claims 9, 20-22, 26 and 30-35 are also non-obvious over the combination of Tashev and Zhan or Florencio. Applicant therefore respectfully requests that the rejection of claims 9, 20-22, 26 and 30-35 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Claims 10, 12, 15 and 18 are dependent on claim 9. Claim 24 is dependent on claim 22. Claims 29 is dependent on claim 26. Since the claim 9, 22 and 26 are non-obvious over a combination of Tashev and Zhan or Florencio, dependent claims 10,12, 15, 18, 24 and 29 are also non-obvious over the combination of Tashev and Zhan or Florencio. Applicant therefore respectfully requests that the rejection of claims 10, 12, 15 18, 24 and 29 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

The office action further states: "**Claim 16 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Marash U.S. Patent 6,198,693.**"

In response to the above rejection, applicant submits that Tashev, in view of Florencio or Zhan and further in view of Marash does not teach or suggest all the limitations in claim 1.

In an earlier part of this response, applicant submitted arguments to show that Tashev, in view of Florencio or Zhan does not teach or suggest the following limitation in 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 said array of sound sensors in a plurality of configurations;"

Applicant further presented an argument to show that claim 9, which is analogous to claim 1 is also non-obvious over Tashev, in view of Florencio or Zhan. Marash does not remedy the deficiencies in the combination of Tashev and Florencio or Zhan. Therefore, Tashev, in view of Florencio or Zhan and further in view of Marash does not teach or suggest the following limitation in claim 9:

> "a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by 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 said array of sound sensors in a plurality of configurations;"

Therefore, claim 9 is non-obvious over Tashev, in view of Florencio or Zhan and further in view of Marash. Claim 16 is dependent on claim 9. Since claim 9 is non-obvious over Tashev, in view of Florencio or Zhan and further in view of Marash, dependent claim 16 is also non-obvious over Tashev, in view of Florencio or Zhan and further in view of Marash. Applicant therefore respectfully requests that the rejection of claim 16 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

The office action further states: "**Claims 8, 17, 25 and 28 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125) and further in view of Nemer U.S. Patent Pub. 2011/0096915.**"

In response to the above rejection, applicant submits that Tashev, in view of Florencio or Zhan and further in view of Nemer does not teach or suggest all the limitations in claim 1.

In an earlier part of this response, applicant submitted arguments to show that Tashev, in view of Florencio or Zhan does not teach or suggest the following limitation in 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 said array of sound sensors in a plurality of configurations;"

It has been further shown that claims 9, 20-21 and 26, which are analogous to claim 1 are also non-obvious over Tashev, in view of Florencio or Zhan. Nemer does not remedy the deficiencies in the combination of Tashev and Florencio or Zhan. Nemer does not teach or suggest the following limitations:

In claim 9:

"a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by 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 said array of sound sensors in a plurality of configurations;"

In claim 20:

"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 said array of sound sensors in a plurality of configurations;"

In claim 21:

"a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by 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 said array of sound sensors in a plurality of configurations;"

In claim 26:

"a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;"

Therefore, claims 9, 20-21 and 26 are non-obvious over Tashev, in view of Florencio or Zhan and further in view of Nemer.

Claim 8 is dependent on claim 1. Claim 17 is dependent on claim 9. Claim 25 is dependent on claim 22. Claim 28 is dependent on claim 26. Since claims 9, 20-22, and 26 are non-obvious over Tashev, in view of Florencio or Zhan and further in view of Nemer, dependent claims 8, 17, 25 and 28 are also non-obvious over Tashev, in view of Florencio or Zhan and further in view of Nemer. Applicant therefore respectfully requests that the rejection of claims 8, 17, 25 and 28 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

The office action further states: "**Claims 3, 6, 11, 14, 23, 24 and 27 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/02528450 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Tashev et al. U.S. Patent Pub. 2008/0232607 (hereinafter Tashev '607).**"

In response to the above rejection, applicant submits that Tashev, in view of Florencio or Zhan and further in view of Tashev '607 does not teach or suggest all the limitations in claim 1.

In an earlier part of this response, applicant submitted arguments to show that Tashev, in view of Florencio or Zhan does not teach or suggest the following limitation in 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 said array of sound sensors in a plurality of configurations;"

It has been further shown that claims 9, 20-21 and 26, which are analogous to claim 1 is also non-obvious over Tashev, in view of Florencio or Zhan. Tashev '607 does not remedy the deficiencies in the combination of Tashev and Florencio or Zhan. Therefore, claims 9, 20-21 and 26 are non-obvious over Tashev, in view of Florencio or Zhan and further in view of Tashev '607.

Claim 3 and claim 6 is dependent on claim 1; Claim 11and claim 14 is dependent on claim 9. Claims 23-24 are dependent on claim 22. Claim 27 is dependent on claim 26. Since claims 9, 20-22, and 26 are non-obvious over Tashev, in view of Florencio or Zhan and further in view of Tashev '607, dependent claims 3, 6, 11, 14, 23-24 and 27 are also non-obvious over Tashev, in view of Florencio or Zhan and further in view of Tashev '607. Applicant therefore respectfully requests that the rejection of claims 3, 6, 11, 14, 23-24 and 27 under 35 U.S.C. 103(a) be reconsidered and withdrawn.

Conclusion

Applicant respectfully requests that a timely notice of allowance be issued in this case. In the interest of compact prosecution, if a claim may be made potentially allowable by an Examiner's amendment, Examiner Escalante is requested to call the undersigned with the proposed amendment.

Respectfully submitted,

Date: January 29, 2018

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
15/293,626	10/14/2016	Manli Zhu	CreativeTech_01RE_US	4199

64188 7590 10/05/2017

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EXAMINER	
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ART UNIT	PAPER NUMBER
3992	

NOTIFICATION DATE	DELIVERY MODE
10/05/2017	ELECTRONIC

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.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

ASH@IPPROCUREMENT.COM
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Office Action Summary	Application No. 15/293,626	Applicant(s) ZHU ET AL.	
	Examiner OVIDIO ESCALANTE	Art Unit 3992	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 *10/14/2016*.

☐ 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) *1-35* 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) *1-35* 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 *10/14/16* 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: _____.

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

DETAILED ACTION

Reissue Applications

2. For reissue applications filed before September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the law and rules in effect on September 15, 2012. Where specifically designated, these are "pre-AIA" provisions.

For reissue applications filed on or after September 16, 2012, all references to 35 U.S.C. 251 and 37 CFR 1.172, 1.175, and 3.73 are to the current provisions.

3. Applicant is reminded of the continuing obligation under 37 CFR 1.178(b), to timely apprise the Office of any prior or concurrent proceeding in which Patent No. 8,861,756 is or was involved. These proceedings would include interferences, reissues, reexaminations, and litigation.

Applicant is further reminded of the continuing obligation under 37 CFR 1.56, to timely apprise the Office of any information which is material to patentability of the claims under consideration in this reissue application.

These obligations rest with each individual associated with the filing and prosecution of this application for reissue. See also MPEP §§ 1404, 1442.01 and 1442.04.

Reissue Declaration

4. The reissue oath/declaration filed with this application is defective (see 37 CFR 1.175 and MPEP § 1414) because of the following:

The examiner notes that in accordance with MPEP 1414, any error in the claims must be identified by reference to the specific claim(s) and the specific claim language wherein lies the error.

The examiner notes that the reissue declaration recites the following reasons:

> The reissue is a broadening reissue.

The examiner determines that the statement does not identify the specific claim(s) and the specific claim language wherein lies the error.

5. Claims 1-35 are rejected as being based upon a defective reissue declaration under 35 U.S.C. 251 as set forth above. See 37 CFR 1.175.

The nature of the defect(s) in the declaration is set forth in the discussion above in this Office action.

35 U.S.C. §112 Sixth Paragraph

The Examiner finds that claims 9, 21, 23, 31-35 recite phrases that invoke 35 U.S.C. §112, 6th paragraph. For support of the Examiners position the Examiner notes the following appropriate 3-prong analysis. See MPEP §2181 I. *See Williamson v. Citrix Online, L.L.C.,* 115 USPQ2d 1105, 1112 (Fed. Cir. 2015).

"Functional Phrase #1 "sound source localization unit" as in claims 9 and 21 and "sound source localizer" as in claims 26, 31, 32, 33, 34 and 35.

The Examiner concludes the phrase: "sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals," as in claims 9 and 21 and "sound source localizer" as in claims 26, 31, 32, 33, 34 and 35; ("Functional Phrase #1"), invokes 35 U.S.C § 112 6th paragraph. To support the Examiner's conclusion, the Examiner notes the following 3-Prong analysis:

a) ***3-Prong Analysis Prong (A)***:

In accordance with the MPEP, Prong (A) requires:

> (A) the claim limitation uses the term "means" or "step" or a term used as a substitute for "means" that is a generic placeholder (also called a nonce term or a non-structural term having no specific structural meaning) for performing the claimed function ....

MPEP § 2181 I. — Prong (A).

As an initial matter, the Examiner finds that Functional Phrase #1 does not use the phrase "means for." The issue arising under Prong (A) then becomes whether or not the claimed "sound source localization unit, or "sound source localizer" is a generic placeholder for the phrase 'means for,' i.e., being applied as a generic means for performing the function. See MPEP 2181 I (C) ¶4.

The Examiner has reviewed the specification of '756 Patent and finds: (1) the '756 Patent **does not** indicate that the phrase "sound source localization unit" or "sound source localizer" is lexicographically defined as a particular structure that performs the recited function; (2) the '756 Patent does not indicate that the phrase "sound source localization unit" or "sound source localizer" refers to a particular structure in the art, that a person having ordinary skill in the art (PHOSITA) would recognize as performing or possessing the recited function. In addition the

Examiner has reviewed the prior art of record and finds that the phrase "sound source localization unit" or "sound source localizer" **does not** infer or require any particular structure as related to FP#1.

Accordingly based upon a review of the '756 Patent itself and the prior art, the Examiner concludes that the phrase "sound source localization unit" as set forth in Functional Phrase #1 is being used as a generic term for structure performing the function, and therefore a place holder for the phrase "means for" performing the recited function. Because "sound source localization unit" or "sound source localizer" is merely a generic placeholder having no specific structure associated therewith, the Examiner concludes that Functional Phrase #1 (FP #1) meets invocation Prong (A).

b) 3-Prong Analysis Prong (B):

In accordance with the MPEP prong (B) requires:

> (B) the term "means" or "step" or the generic placeholder is modified by functional language, typically, but not always linked by the transition word "for" (e.g., "means for") or another linking word or phrase, such as "configured to" or "so that" ....

MPEP § 2181 I. — Prong (B).

Based upon a review of claims 9 and 21, the Examiner finds that the function associated with Functional Phrase #1 is: *estimates a spatial location of said target sound signals from said received sound signals.*

Because Functional Phrase #1 includes the function expressly noted above, the Examiner concludes that Functional Phrase #1 meets invocation Prong (B). Additionally, the Examiner notes that because nothing in the written description contradicts the plain language describing this function, the function within Functional Phrase #1 will have its ordinary and accustomed meaning.

c) 3-Prong Analysis: Prong (C)

In accordance with the MPEP, Prong (C) requires:

> (C) the term "means" or "step" or the generic placeholder is not modified by sufficient structure, material, or acts for performing the claimed function.

MPEP § 2181 (I) — Prong (C)

Based upon a review of the entire Functional Phrase #1, the Examiner finds that Functional Phrase #1 does not contain sufficient structure for performing the entire claimed function that is set forth within Functional Phrase #1. In fact, the Examiner finds that Functional Phrase #1 recites very little structure (if any) for performing the claimed function.

Because Functional Phrase #1 does not contain sufficient structure for performing the entire claimed function, the Examiner concludes that Functional Phrase #1 meets invocation Prong (C).

d) Corresponding Structure for Functional Phrase #1

With reference to figure 2 and its related text, the examiner notes that the patent specification does not describe the specific structural requirements of the sound localization unit. That is, while the related text describes the function of the sound localization unit, the related text of figure 2 does not disclose the structure of the sound localization unit.

With reference to Figure 14, the patent specification states that the function of the localization unit is performed by a Digital Signal Processor. See col. 15, lines 4-6 which discloses that the DSP 1403 implements the sound source localization unit.

Therefore, the examiner considers the structure of the sound localization unit or sound localizer to be a digital signal processor or equivalents thereof.

Claim Rejections - 35 USC § 103

6. In the event the determination of the status of the application as subject to AIA 35 U.S.C. 102 and 103 (or as subject to pre-AIA 35 U.S.C. 102 and 103) is incorrect, any correction of the statutory basis for the rejection will not be considered a new ground of rejection if the prior art relied upon, and the rationale supporting the rejection, would be the same under either status.

7. The following is a quotation of pre-AIA 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, 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.

8. Claims 1, 2, 4, 7, 9, 10, 12, 15, 18, 20-22, 24, 29-35 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010)[1]. (U.S. Patent Publication 2011/0135125).

Regarding claim 1:

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[1] The examiner notes that WO 2010/0120162 was published on February 25, 2010 and therefore qualifies as prior art. Although the description is not in English, the examiner is relying upon U.S. Patent Publication 2011/0135125 for providing citations. The U.S. Publication is a continuation and therefore, the subject matter in the U.S. Publication is fully supported by the WO 2010/0120162 publication.

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other directions. See paragraph [0011]. See also paragraph [0050] which discloses that the spatial sound distribution model includes ambient noise in addition to the target sound (plurality of sound signals).

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors) position in an arbitrary configuration (Tashev discloses of planar array and linear arrays as examples--see paragraph [0059]), a sound source localizer (As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined), a beamformer (paragraph [0071]), and a noise reducer (See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.)

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source can be a human voice.

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,

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the origin point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance <u>between each of the sound sensors and a reference point</u>. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Nonetheless, Zhan is directed to a method for controlling sound focusing (see the abstract). With reference to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point).

Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

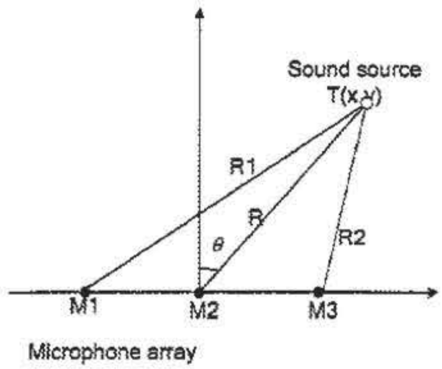

FIG. 2

Alternatively, Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation).

Likewise, the use of distance, angle and delay as evidence by Zhan shows that it was well known to sue this information in order to locate a sound source. As set forth above, both Teshav and Zhan are directed to locating a sound source relative to a reference microphone in a microphone array.

Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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;

As disclosed in paragraph [0059], Teshav discloses the “delay and sum” computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the "delay and sum" process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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

See paragraph [0045] which describes the designation of the maximum energy direction as the direction to the sound source. In addition, see steps 204-208 of figure 2 and its related text.

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

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 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.

See paragraphs [0050-0058] of Tashev

Regarding claims 4:

The method of claim 3, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059]

Regarding claim 7:

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.

See paragraph [0052] of Florencio which discloses the use of a noise model algorithm. As set forth above, the examiner notes that Tashev discloses a noise reduction unit. Therefore, it would have been obvious to use a noise reduction model as disclosed by Florencio. In addition, Tashev discloses in paragraph [0072] than any conventional noise suppression procedure may be used.

Regarding claim 9:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors) position in an arbitrary configuration (Tashev discloses of planar array and linear arrays as examples--see paragraph [0059]). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and an origin of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

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,

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Zhan is directed to a method for controlling sound focusing (see the abstract). With refer3ence to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point). .

Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

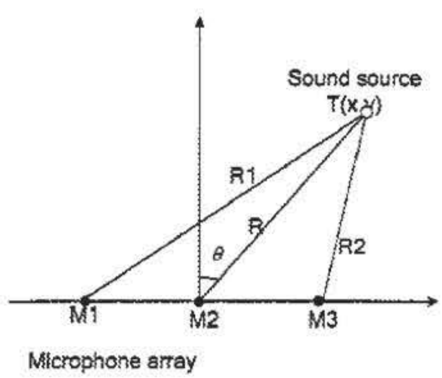

FIG. 2

Alternatively, Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation).

Likewise, the use of distance, angle and delay as evidence by Zhan shows that it was well known to sue this information in order to locate a sound source. As set forth above, both Teshav and Zhan are directed to locating a sound source relative to a reference microphone in a microphone array.

Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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;

As disclosed in paragraph [0059], Teshav discloses the “delay and sum” computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the “delay and sum” process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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;

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 10:

The system of claim 9, 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.

See paragraphs [0050-0058] of Tashev

Regarding claim 12:

The system of claim 11, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

Regarding claim 15:

The system of claim 9, 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.

See paragraph [0052] of Florencio which discloses the use of a noise model algorithm. As set forth above, the examiner notes that Tashev discloses a noise reduction unit. Therefore, it would have been obvious to use a noise reduction model as disclosed by Florencio. In addition, Tashev discloses in paragraph [0072] than any conventional noise suppression procedure may be used.

Regarding claim 18:

The system of claim 9, 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.

See paragraph [0059] where Teshav discloses a linear array of sound sensors.

Regarding claim 20:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other directions. See paragraph [0011]. See also paragraph [0050] which discloses that the spatial sound distribution model includes ambient noise in addition to the target sound (plurality of sound signals).

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors) position in an arbitrary configuration (Tashev discloses of planar array and linear arrays as examples--see paragraph [0059]), a sound source localizer (As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined), a beamformer (paragraph [0071]), and a noise reducer (See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.)

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

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,

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Zhan is directed to a method for controlling sound focusing (see the abstract). With refer3ence to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point).

Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

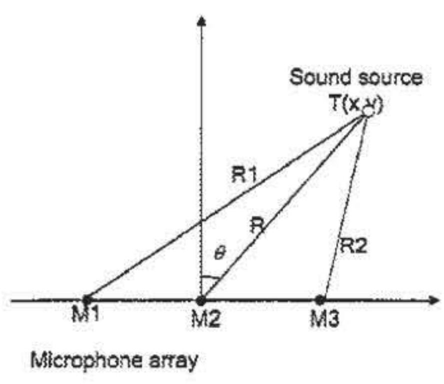

FIG. 2

Alternatively, Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation).

Likewise, the use of distance, angle and delay as evidence by Zhan shows that it was well known to sue this information in order to locate a sound source. As set forth above, both Teshav and Zhan are directed to locating a sound source relative to a reference microphone in a microphone array.

Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the "delay and sum" process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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

See paragraph [0045] which describes the designation of the maximum energy direction as the direction to the sound source.

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

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 21:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other directions. See paragraph [0011]. See also paragraph [0050] which discloses that the spatial sound distribution model includes ambient noise in addition to the target sound (plurality of sound signals).

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors) position in an arbitrary configuration (Tashev discloses of planar array and linear arrays as examples--see paragraph [0059]). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals <u>by determining a delay between each of said sound sensors and an origin of said array of said sound</u> sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the “earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

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,

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Zhan is directed to a method for controlling sound focusing (see the abstract). With refer3ence to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point). .

Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

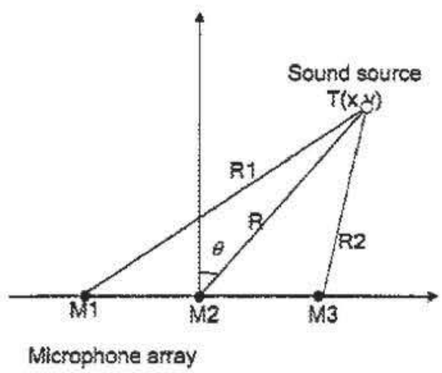

FIG. 2

Alternatively, Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation).

Likewise, the use of distance, angle and delay as evidence by Zhan shows that it was well known to sue this information in order to locate a sound source. As set forth above, both Teshav and Zhan are directed to locating a sound source relative to a reference microphone in a microphone array.

Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can

likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the “delay and sum” process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 22:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other directions. See paragraph [0011]. See also paragraph [0050] which discloses that the spatial sound distribution model includes ambient noise in addition to the target sound (plurality of sound signals).

providing a microphone array system comprising an array of sound sensors, a sound source localizer, a beamformer, and a noise reducer, wherein said sound source localizer, said beamformer, and said noise reducer are in operative communication with said array of said sound sensors;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors), a sound source localizer (As discussed in paragraph [0072], Tashev discloses that the frames representing the “earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined), a beamformer (paragraph [0071]), and a noise reducer (See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.)

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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,

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Zhan is directed to a method for controlling sound focusing (see the abstract). With refer3ence to Figure 2 and paragraph [0033], Zhan discloses determining a delay between each sound sensor and a reference point of a microphone array (Zhan discloses M1, M2 and M3 are omnidirectional microphones at intervals of d. Zhan explains that the time delay between M1 and M2 and between M2 and M3 are determined). Zhan discloses that the reference microphone is M2 in one embodiment (see paragraph [0032]). Zhan discloses that the distance between each sound sensor is referenced by d. That is distance between M1 and M2 is d_{12} and the distance between M2 and M3 is d_{23} (with M2 being the reference point). .

Zhan explains that figure 2 shows how to obtain the position information of a sound source relative to a reference microphone by computing the distance and the azimuth from the sound source to the reference microphone (M2), where the azimuth is an angle between the rectilinear direction from the sound source to the reference microphone and the vertical direction (See paragraphs [0032] – [0035]).

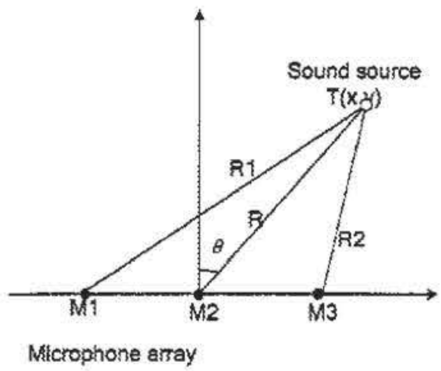

FIG. 2

Alternatively, Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation).

Likewise, the use of distance, angle and delay as evidence by Zhan shows that it was well known to sue this information in order to locate a sound source. As set forth above, both Teshav and Zhan are directed to locating a sound source relative to a reference microphone in a microphone array.

Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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 two or more of said sound sensors;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the "delay and sum" process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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

See paragraph [0045] which describes the designation of the maximum energy direction as the direction to the sound source. In addition, see steps 204-208 of figure 2 and its related text.

performing 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 beamformer, wherein said beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 26:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

an array of sound sensors, 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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point, 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,

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

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 two or more of said sound sensors;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can

likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the "delay and sum" process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

a beamformer 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 beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 29:

The system of claim 26, wherein said array of said sound sensors is one of a linear array of said sound sensors and a circular array of said sound sensors.

Tashev in paragraph [0059] discloses a linear array. In addition, see figure 2 of Florencio which discloses a circular array and figure 2 of Zhan which discloses a linear array of sound sensors.

Regarding claim 30:

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

providing a microphone array system comprising an array of sound sensors, a sound source localizer, a beamformer, and a noise reducer, wherein said sound source localizer, said beamformer, and said noise reducer are in operative communication with said array of said sound sensors;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors), a sound source localizer (As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined), a beamformer (paragraph [0071]), and a noise reducer (See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.)

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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,

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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 two or more of said sound sensors;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the "delay and sum" process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

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

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

performing 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 beamformer, wherein said beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 31:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph

[0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point, 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,

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

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 two or more of said sound sensors;

As disclosed in paragraph [0059], Teshav discloses the "delay and sum" computation determines the number of beams required and the search area around the microphone array can

likewise be determined. Teshav also discloses that the target source can emit in 2D or in 3D planes as discussed in paragraph [0059].

As described in paragraph [0072], for frames are selected and processed. The selected frames are then subjected to the disclosed process. Teshav discloses a number of frames/samples that is used in the "delay and sum" process.

As discussed above, based on the “delay and sum” process, Tashev discloses that the frames are subjected to the beamsteering procedure and the energy direction of the microphone arrays is changed as to the direction of the sound source (step 208 of fig. 2 and related text).

a beamformer 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 beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

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

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 32:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

an array of sound sensors, 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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis;

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

a noise reducer that suppresses said ambient noise signals.

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 33:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

an array of sound sensors, 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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point, 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;

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

a noise reducer that suppresses said ambient noise signals.

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

Regarding claim 34:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

an array of sound sensors, 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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point, 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;

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

a noise reducer that suppresses said ambient noise signals.

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

Regarding claim 35:

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

Tashev is directed to a system and process for finding the direction of a sound source from a microphone array including doing spatial filtering which suppresses noises coming from other direction. See paragraph [0011].

an array of sound sensors positioned in a non-circular 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;

See paragraphs [0010-0011]. Tashev discloses a microphone array (array of sound sensors). As explained in paragraph [0011], noise is detected as well as frames which contain a significant sound source. In paragraph [0013], Tashev discloses the sound source to be a human voice.

a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors

As discussed in the background of Tashev, a process usually finds the direction of the sound source in two phases. During the first phase, the delays are calculated for each microphone pair based on correlation function estimation. In the second phase, all time delay estimates are combined to compute the final direction of the sound source. See paragraph [0006]

As discussed in paragraph [0072], Tashev discloses that the frames representing the "earliest captured frame of each microphone signal" are selected. As discussed in paragraph [0073], each of the energy values computed for each directed is weighted and the energy as a function of the direction angle is determined. The examiner notes that the reference point of said array of said sound sensors is the center of the microphone array (see paragraphs [0071-0072])

as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis,

The examiner notes that in the background section, Teshav discloses that sound localization systems define location in two angles (direction and elevation) or full 3D localization (i.e. direction, elevation and distance). See paragraph [0005]. In addition, Teshav considers the distance that speakers are away from the microphones (see paragraph [0071] where it is determined that a person is 1 meter away from the microphone and another person is 2.5 meters away).

The examiner notes that Teshav does not specially disclose that the delay is based on a function of distance between each of the sound sensors and a reference point. However, it would have been obvious, in not inherent, that distance is considered since Teshav discloses that the earliest captured frame of each microphone signal is selected (and thus closer sound sources will arrive first and hence represents a shorter distance). In addition, Teshav discloses that distance is considered for full 3D localization (see paragraph [0059] which discloses considering the elevation angle of the sound source in a teleconferencing situation.

Florencio discloses a method for sound source localization. Florencio discloses in paragraph [0007] that the signals detected at the microphone array are used to obtain functions comprising distance, azimuth and elevation data. With reference to figure 5 and paragraphs [0058-0059], Florencio discloses of two sources that have the same azimuth and elevation angles. As further shown in paragraph [0041], Florencio discloses using time delay from the source along with distance to the source as part of its function.

Therefore, it would have been obvious to one of ordinary skill in the art to consider delay as a function of distance. As explained by Teshav distance is considered when 3D localization is needed and in addition, Florencio likewise discloses a 3D localization type of system which considers, time delay, elevation, distance and azimuth for determining the location of the sound source.

The examiner notes that the use of distance, azimuth and elevation would have been predictable to one of ordinary skill in the art in determine the location of the sound source since both Teshav and Florencio both disclose the use of 3D localization as well as 2D localization (which doesn't rely on elevation). Thus, based on the well-known teachings of considering distance, delay and angle, the examiner maintains that the using this as part of the function would have been obvious to one of ordinary skill in the art.

wherein said distance between each of said sound sensors and said reference point varies from a minimum value to a maximum value, and wherein said minimum value corresponds to zero and said maximum value is defined based on a limitation associated with size of said system;

As shown in figure 2 of Zhan, the minimum value corresponds to zero since the reference point is located at M2 which is the midpoint of the array. The maximum value is at M1 or M2 which is located at the two extremities of the linear array.

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise. In addition, see steps 204-208 of figure 2 and its related text. See also paragraph [0048] which discloses of changing the beam of the microphone array.

a noise reducer that suppresses said ambient noise signals.

See paragraph [0011] and paragraph [0072] of Tashev which discusses the use of noise suppression on the ambient noise signals.

9. Claim 16 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub.

2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Marash U.S. Patent 6,198,693.

Regarding claim 16:

The system of claim 9, 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.

Teshav, as set forth above, does not specifically disclose of an audio codec that converts sound signals to digital sound signals.

Marash discloses a system and method for finding the direction of a wave source using an array of sensors. As disclosed in col. 5, lines 57-65, analog signals representing the sound sensed or measured by the microphones are converted o digital signals by the A-to-D converter 2 which samples the analog signals at an appropriate sampling frequency.

Therefore, it would have been obvious to one of ordinary skill in the art to use an A-to-D converter so that the signals can be filtered for specific frequency optimal for detecting or determining the direction of the signal. See col. 5, lines 27-47.

10. Claims 8, 17, 25 and 28 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S. Patent Publication 2011/0135125 and further in view of Nemer U.S. Patent Pub. 2011/0096915.

Regarding claims 8 and 25:

The method of claims 1 and 22, wherein said noise reduction unit or noise reducer 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 or said beamformer for sub-band adaptive beamforming.

Regarding claims 17 and 28:

The system of claims 9 and 26, wherein said noise reduction unit or said noise reducer 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.

Tashev as disclosed above, teaches a noise reducer which performs noise reduction. Tashev does not specifically disclose of using an analysis filter bank.

Nonetheless, Nemer discloses at paragraphs [0064-0068] which discloses the use of an analysis filter bank to generate sub-band signals.

Therefore, it would have been obvious to one of ordinary skill in the art to use an analysis filter bank for the creation of frequency sub-bands so that each energy band can be analyzed. The examiner notes that Tashev already discloses of using a noise reducer within a beamformer and therefore it would have been obvious to include a filterbank so that specific frequency sub-bands can employed for the noise reduction.

11. Claims 3, 6, 11, 14, 23, 24 and 27 is/are rejected under pre-AIA 35 U.S.C. 103(a) as being unpatentable over Tashev U.S. Patent Pub. 2004/0252845 in view of Florencio et al. U.S. Patent Pub. 2011/0317522 or Zhan et al. WO 2010/0120162 (published February 25, 2010) (U.S.

Patent Publication 2011/0135125 and further in view of Tashev et al. U.S. Patent Pub. 2008/0232607 (hereinafter Tashev '607).

Regarding claims 3, 11 and 23:

The method of claims 1, 9 and 22, wherein said beamforming or adaptive beamforming comprises:

The examiner notes that Tashev as described above, discloses a beamformer. However, Tashev does not specifically disclose of a blocking matrix and an adaptive filter as claimed.

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

With reference to figure 2 of Tashev '607, a fixed beamformer, blocking matrix and adaptive filter are disclosed. See also the abstract.

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;

See paragraphs [0059 & 0071], which discuss the shape of the beam of the beamformer as well as paragraph [0072] which discloses the suppression of ambient noise.

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

See paragraphs [0028-0030] which describes the input of ambient noise signals to the adaptive filter.

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.

See paragraph [0030] which discloses that the noise signals are filtered since the output only consists of the target signal.

Therefore, it would have been obvious to include a fixed beamformer with adaptive blocking matrix and adaptive filtering in order to provide enhanced noise suppression capability.

The examiner notes that both Tashev and Tashev '607 discloses of a noise suppression technique for a beamformer. Therefore, one of ordinary skill the in the art would have found it predictable to use the technique described by Tashev '607 in order to increase the noise suppression capability of

Regarding claims 6 and 24:

The method of claims 3 and 23, wherein said beamforming or said adaptive beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said beamformer 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.

Regarding claims 14 and 27:

The system of claims 9 and 26, wherein said beamformer or 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.

As discussed above with respect to claim 23, the examiner notes that Tashev '607 discloses the use of adaptive filtering. As further described by Tashev '607 as part of its adaptive interference canceller paragraphs [0044-007], discloses removing the signals that are correlated to the interference signals. Tashev '607 discloses of updated the filter coefficients by adjusting the step size.

The examiner notes that the teachings of Tashev '607 shows that it was well known in the art to adjust the step size for the filter coefficients in response to detecting the absence of the target sound signal.

Therefore, it would have been obvious to one of ordinary skill in the art to adjust a step size since this method was well known in the art to be used when updated filter coefficients during an adaptive filtering process. The use of this method would have been predictable to one of ordinary skill in the art.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Ovidio Escalante whose telephone number is (571)272-7537. The examiner can normally be reached on Monday to Friday - 6:00 AM to 2:30 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Michael Fuelling can be reached on (571) 270-1367. The fax phone number for the organization where this application or proceeding is assigned is 571-273-9000.

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.

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re. application of:
Zhu, Manli

Application No.:	Reissue of US Pat no. 8861756
Art Unit:	2653
Filed:	03/16/2011
Examiner:	Not assigned
Applicant:	Li Creative Technologies, Inc.
Atty. Docket No.:	CreativeTech_01RE_US
Title:	Microphone Array System

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

Preliminary amendment

With reference to the above-identified patent application, please amend the application as shown below.

Amendments to the Specification begin on page 2 of this document.

Amendments to the Claims begin on page 3 of this document.

Remarks begin on page 14 of this document.

Amendments to the Specification

Please amend the first paragraph of the specification as shown below:

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is a reissue application of U.S. Patent application No. 13/049,877, filed March 16, 2011 (now U.S. Patent No. 8861756), which 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.

Amendments to the Claims:

The listing of claims provided below will replace all prior versions.

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;

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;

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, 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.

4. The method of claim 3, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

5. The method of claim 3, 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.

6. The method of claim 3, 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.

7. 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.

8. 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.

9. 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 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;

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.

10. The system of claim 9, 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.

11. The system of claim 9, 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.

12. The system of claim 11, wherein said fixed beamformer performs fixed beamforming by filtering and summing output sound signals from said sound sensors.

13. The system of claim 11, 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.

14. The system of claim 9, 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.

15. The system of claim 9, 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.

16. The system of claim 9, 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.

17. The system of claim 9, 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.

18. The system of claim 9, 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.

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

20. 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.

21. 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 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;

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.

22. 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, a sound source localizer, a beamformer, and a noise reducer, wherein said sound source localizer, said beamformer, and said noise reducer 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

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

performing 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 beamformer, wherein said beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

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

23. The method of claim 22, wherein said beamforming comprises:

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

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.

24. The method of claim 23, wherein said beamforming further comprises detecting said presence of said target sound signal by an adaptation control unit provided in said beamformer 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.

25. The method of claim 22, wherein said noise reducer performs noise reduction in a plurality of frequency sub-bands, wherein said frequency sub-bands are employed by an analysis filter bank of said beamformer for sub-band adaptive beamforming.

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

an array of sound sensors, 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 localizer that estimates a spatial location of said target sound signal from said received sound signals, by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a beamformer 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 beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

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

27. The system of claim 26, wherein said beamformer 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.

28. The system of claim 26, wherein said noise reducer performs noise reduction in a plurality of frequency sub-bands employed by an analysis filter bank of said beamformer for sub-band adaptive beamforming.

29. The system of claim 26, wherein said array of said sound sensors is one of a linear array of said sound sensors and a circular array of said sound sensors.

30. 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, a sound source localizer, a beamformer, and a noise reducer, wherein said sound source localizer, said beamformer, and said noise reducer 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 a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

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

performing 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 beamformer, wherein said beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

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

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

an array of sound sensors, 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 localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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 two or more of said sound sensors;

a beamformer 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 beamformer enhances said target sound signal and partially suppresses said ambient noise signals; and

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

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

an array of sound sensors, 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 localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis;

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reducer that suppresses said ambient noise signals.

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

an array of sound sensors, 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 localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reducer that suppresses said ambient noise signals.

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

an array of sound sensors, 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 localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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;

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reducer that suppresses said ambient noise signals.

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

an array of sound sensors positioned in a non-circular 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 localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis, wherein said distance between each of said sound sensors and said reference point varies from a minimum value to a maximum value, and wherein said minimum value corresponds to zero and said maximum value is defined based on a limitation associated with size of said system;

a beamformer that enhances said target sound signal and suppresses said ambient noise signals; and

a noise reducer that suppresses said ambient noise signals.

Remarks

Amendments to specification

Applicant respectfully submits that in the specification, the 'cross-reference to related applications' paragraph has been amended on the first page of the original application after the title.

Amendments to claims

Support for the following amendment in claim 21 "a sound source localization unit that estimates a spatial location of said target sound signal from said received sound signals by 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" is found in claim 1, 9 and 20 of the applicant's patent US 8,861,756.

Support for the following limitation in new claims 22, 26, 30, 31, 32, 33 and 34 "determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point" is found in column 7, lines 65-66 of the applicant's patent US 8,861,756.

Support for the following limitation in new claim 32 "a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point and an angle of each of said sound sensors biased from a reference axis" is found in column 7, lines 56-67 and column 8, lines 1-21 of the applicant's patent US 8,861,756.

Support for the following limitation in new claim 33 "a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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" is found in claims 1 and 9, and in column 8, lines 22-67 and column 9, lines 1-11 of the applicant's patent US 8,861,756.

Support for the following limitation in new claim 34 "a sound source localizer that estimates a spatial location of said target sound signal from said received sound signals by determining a delay between each of said sound sensors and a reference point of said array of said sound sensors as a function of distance between each of said sound sensors and said reference point, 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" is found in claims 20 and 21, and column 9, lines 12-67 and column 10, lines 1-67 of the applicant's patent US 8,861,756.

Support for the following limitation in new claims 32, 33 and 34 "a beamformer that enhances said target sound signal and suppresses said ambient noise signals" is found in claims 1, 9, 20 and 21 of the applicant's patent US 8,861,756.

Support for the following limitation in new claims 32, 33 and 34 "a noise reducer that suppresses said ambient noise signals" is found in claims 1, 9, 20 and 21 of the applicant's patent US 8,861,756.

Support for the following limitation in new claim 35 "an array of sound sensors positioned in a non-circular configuration" is found in FIG. 19E.

Support for the following limitation is new claim 35 "wherein said distance between each of said sound sensors and said reference point varies from a minimum value to a maximum value, and wherein said minimum value corresponds to zero and said maximum value is defined based on a limitation associated with size of said system" is found in column 7, lines 56-67 and column 8, lines 1-21 of the applicant's patent US 8,861,756.

Applicant respectfully request that the forgoing amendments be made prior to examination of the present application. Claim 21 is amended; claims 1-20 remain as previously presented; and claims 22-34 are newly added.

Applicant submits that no new matter is added to the claims.

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 to call the undersigned with the proposed amendment.

Respectfully submitted,

Date: October 14, 2016

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PTO/AIA/50 (09-14)
Approved for use through 10/31/2016. OMB 0651-0033
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

REISSUE PATENT APPLICATION TRANSMITTAL

Address to:		
Mail Stop Reissue **Commissioner for Patents** **P.O. Box 1450** **Alexandria, VA 22313-1450**	Attorney Docket No.	CreativeTech_01RE_US
	First Named Inventor	Manli, Zhu
	Original Patent Number	US8861756
	Original Patent Issue Date (Month/Day/Year)	10/14/2014
	Express Mail Label No.	

APPLICATION FOR REISSUE OF:
(*Check applicable box*) ☑ Utility Patent ☐ Design Patent ☐ Plant Patent

APPLICATION ELEMENTS (37 CFR 1.173)

1. ☐ **Fee Transmittal Form** (PTO/SB/56)
2. ☑ **Applicant asserts small entity status.** See 37 CFR 1.27
3. ☐ **Applicant certifies micro entity status.** See 37 CFR 1.29. Applicant must attach form PTO/SB/15A or B or equivalent.
4. ☑ **Specification and Claims** in double column copy of patent format (*amended, if appropriate*)
5. ☑ **Drawing(s)** (*proposed amendments, if appropriate*)
6. ☑ **Reissue Oath/Declaration or Substitute Statement** (37 CFR 1.175) (PTO/AIA/05, 06, or 07)
7. ☑ **Application Data Sheet** NOTE: Benefit claims under 37 CFR 1.78 and foreign priority claims under 37 CFR 1.55 MUST be set forth in an Application Data Sheet (ADS).
8. ☑ **Original U.S. Patent currently assigned?** ☑ Yes ☐ No
   (*If Yes, check applicable box(es)*)
   - ☑ Written Consent of all Assigness (PTO/AIA/53)
   - ☑ 37 CFR 3.73(c) Statement (PTO/AIA/96)
9. ☐ **CD-ROM or CD-R** in duplicate, Computer Program (*Appendix*) or large table
   - ☐ Landscape Table on CD
10. **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 (2 copies) or CD-R (2 copies); **or**
      - ii. ☐ Paper
    - c. ☐ Statements verifying identity of above copies

ACCOMPANYING APPLICATION PARTS

11. ☑ **Statement of status and support for all changes to the claims.** See 37 CFR 1.173(c).
12. ☑ **Power of Attorney**
13. ☐ **Information Disclosure Statement (IDS)** PTOSB/08 or PTO-1449
    - ☐ Copies of citations attached
14. ☐ **English translation of Reissue Oath/Declaration** (*if applicable*)
15. ☐ **Return Receipt Postcard** (MPEP § 503) (Should be specifically itemized)
16. ☑ **Preliminary Amendment** (37 CFR 1.173; MPEP § 1453)
17. ☐ **Other:** ______

☐ This is a continuation reissue or divisional reissue application (*i.e.*, a second or subsequent reissue application for the same issued patent). (*Check box if applicable.*)

18. CORRESPONDENCE ADDRESS

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