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.

<u>Amendments to the Claims:</u>

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 τ=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

/a tankha/
Ashok Tankha
Attorney For Applicant
Reg. No. 33,802

Correspondence Address
Lipton, Weinberger & Husick
36 Greenleigh Drive
Sewell, NJ 08080
Phone: 856-266-5145
Fax: 856-374-0246
Email: ash@ipprocure.com

Electronic Patent Application Fee Transmittal

Application Number:	
Filing Date:	
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Filer:	Ashok Tankha
Attorney Docket Number:	CreativeTech_01RE_US

Filed as Small Entity

Filing Fees for **Reissue (Utility)**

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
UTILITY REISSUE BASIC	2014	1	140	140
DESIGN AND UTILITY REISSUE BASIC	2114	1	300	300
DESIGN AND UTILITY REISSUE BASIC	2314	1	1080	1080
Pages:				
Claims:				
REISSUE CLAIMS IN EXCESS OF 20 FOR SMALL	2205	15	40	600
REISSUE- INDEPENDENT CLAIMS	2204	9	210	1890
Miscellaneous-Filing:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**4010**

Electronic Acknowledgement Receipt

EFS ID:	27212512
Application Number:	15293626
International Application Number:	
Confirmation Number:	4199
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Customer Number:	64188
Filer:	Ashok Tankha
Filer Authorized By:	
Attorney Docket Number:	CreativeTech_01RE_US
Receipt Date:	14-OCT-2016
Filing Date:	
Time Stamp:	13:28:15
Application Type:	Reissue (Utility)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	CreativeTech_01RE_US_Transmittal_aia0050.pdf	304239 5ba4cd9fdcfa111ff8c3e18d5db208428a60fb34	no	2

Warnings:

Information:					
2	Application Data Sheet	CreativeTech_01RE_US_ADS.pdf	113526 bd468fb58ceaaf702a73d782639d54aac74de65d	no	9
Warnings:					
Information:					
This is not an USPTO supplied ADS fillable form					
3	Oath or Declaration filed	CreativeTech_01RE_US_Declaration_aia0006_Qi_Li.pdf	341521 3429885581e6461443df26a926bf16866f83d5b8	no	3
Warnings:					
Information:					
4	Oath or Declaration filed	CreativeTech_01RE_US-Suppl_decl_aia0010_Manli.PDF	457375 ee3b37931f70f1c40862c1fae9ee87af9b13fa5f	no	1
Warnings:					
Information:					
5	Power of Attorney	CreativeTech_01RE_US_POA_aia82B_Qi_Li.pdf	217435 dee35f98003745332b0e46376f1e2f0b465ccf31	no	3
Warnings:					
Information:					
6	Miscellaneous Incoming Letter	CreativeTech_01RE_US_Assignment_extract.pdf	37834 6fd05382d647efad4b69996997b99ddc9da975ec	no	1
Warnings:					
Information:					
7	Consent of Assignee accompanying the declaration	CreativeTech_01RE_US_Consent_of_assignee_aia0053_Qi_Li.pdf	294050 3c5d4715f7a21e46a0be6562c2083b99d1b46512	no	2
Warnings:					
Information:					
8	Assignee showing of ownership per 37 CFR 3.73	CreativeTech_01RE_US_aia0096_Qi_Li.pdf	137021 be49baeb9e29a905f7d545bb7c67241decdf8439	no	3
Warnings:					
Information:					

9		CreativeTech_01RE_US_Pat8861756B2.pdf	976281 9468b7d67cf442c5250654e317ad9988f3119e85	yes	47

Multipart Description/PDF files in .zip description

Document Description	Start	End
Abstract	1	1
Drawings-only black and white line drawings	2	35
Specification	36	45
Claims	46	47

Warnings:

Information:

10	Preliminary Amendment	CreativeTech_01RE_US_Amendment.pdf	9561327 0104a273baa61a9080dd13745118dc25c1b9a2cc	no	20

Warnings:

Information:

11	Fee Worksheet (SB06)	fee-info.pdf	38038 badddd6d2d67fa28de55c8d2f85cb81df4bf6ac4d	no	2

Warnings:

Information:

Total Files Size (in bytes):	12478647

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New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

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

☐ The address associated with Customer Number: ____________________ **OR** ☑ Correspondence address below

Name	Ashok Tankha				
Address	36 Greenleigh drive				
City	Sewell	State	NJ	Zip Code	08080
Country	US	Telephone	856-266-5145		
Email	ash@ipprocurement.com				

Signature	/a tankha/	Date	10/14/2016
Name (Print/Type)	Ashok Tankha	Registration No.	33802

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.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2:

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Inventor Information:

Inventor 1 Remove
Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Manli		Zhu	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	Pearl River	State/Province	NY	Country of Residence	US

Mailing Address of Inventor:

Address 1	46 E Crooked Hill Road		
Address 2			
City	Pearl River	**State/Province**	NY
Postal Code	1095	**Country**	US

Inventor 2 Remove
Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Qi		Li	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	New Providence	State/Province	NJ	Country of Residence	US

Mailing Address of Inventor:

Address 1	225 Runnymede Parkway		
Address 2			
City	New Providence	**State/Province**	NJ
Postal Code	07974	**Country**	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below.
For further information see 37 CFR 1.33(a).

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

☒ **An Address is being provided for the correspondence Information of this application.**

Name 1	Ashok Tankha	**Name 2**	
Address 1	36 Greenleigh drive		
Address 2			
City	Sewell	**State/Province**	NJ
Country	US	**Postal Code**	08080
Phone Number	856-266-5145	**Fax Number**	856-374-0246
Email Address	ash@ipprocurement.com	Add Email	Remove Email

Application Information:

Title of the Invention	Microphone Array System		
Attorney Docket Number	CreativeTech_01RE_US	**Small Entity Status Claimed**	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)	4	**Suggested Figure for Publication (if any)**	

Filing By Reference:

Only complete this section when filing an application by reference under 35 U.S.C. 111(c) and 37 CFR 1.57(a). Do not complete this section if application papers including a specification and any drawings are being filed. Any domestic benefit or foreign priority information must be provided in the appropriate section(s) below (i.e., "Domestic Benefit/National Stage Information" and "Foreign Priority Information").

For the purposes of a filing date under 37 CFR 1.53(b), the description and any drawings of the present application are replaced by this reference to the previously filed application, subject to conditions and requirements of 37 CFR 1.57(a).

Application number of the previously filed application	Filing date (YYYY-MM-DD)	Intellectual Property Authority or Country

Publication Information:

☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)

☐ **Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Either enter Customer Number or complete the Representative Name section below. If both sections are completed the customer Number will be used for the Representative Information during processing.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

Please Select One:	◯ Customer Number	◉ US Patent Practitioner	◯ Limited Recognition (37 CFR 11.9)	
Prefix	**Given Name**	Middle Name	**Family Name**	Suffix
Mr.	Ashok		Tankha	
Registration Number	33802			

Additional Representative Information blocks may be generated within this form by selecting the **Add** button.

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, 365(c), or 386(c) or indicate National Stage entry from a PCT application. Providing benefit claim information in the Application Data Sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78.
When referring to the current application, please leave the "Application Number" field blank.

Prior Application Status			
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
	reissue of	13/049877	2011-03-16

Prior Application Status	Patented				
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
13/049877	Claims benefit of provisio	61/403952	2010-09-24	8861756	2014-10-14

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button.

Foreign Priority Information:

This section allows for the applicant to claim priority to a foreign application. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55. When priority is claimed to a foreign application that is eligible for retrieval under the priority document exchange program (PDX)[i] the information will be used by the Office to automatically attempt retrieval pursuant to 37 CFR 1.55(i)(1) and (2). Under the PDX program, applicant bears the ultimate responsibility for ensuring that a copy of the foreign application is received by the Office from the participating foreign intellectual property office, or a certified copy of the foreign priority application is filed, within the time period specified in 37 CFR 1.55(g)(1).

Application Number	Country[i]	Filing Date (YYYY-MM-DD)	Access Code[i] (if applicable)

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications

☐ This application (1) claims priority to or the benefit of an application filed before March 16, 2013 and (2) also contains, or contained at any time, a claim to a claimed invention that has an effective filing date on or after March 16, 2013.
NOTE: By providing this statement under 37 CFR 1.55 or 1.78, this application, with a filing date on or after March 16, 2013, will be examined under the first inventor to file provisions of the AIA.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

Authorization or Opt-Out of Authorization to Permit Access:

When this Application Data Sheet is properly signed and filed with the application, applicant has provided written authority to permit a participating foreign intellectual property (IP) office access to the instant application-as-filed (see paragraph A in subsection 1 below) and the European Patent Office (EPO) access to any search results from the instant application (see paragraph B in subsection 1 below).

Should applicant choose not to provide an authorization identified in subsection 1 below, applicant **must opt-out** of the authorization by checking the corresponding box A or B or both in subsection 2 below.

NOTE: This section of the Application Data Sheet is **ONLY** reviewed and processed with the **INITIAL** filing of an application. After the initial filing of an application, an Application Data Sheet cannot be used to provide or rescind authorization for access by a foreign IP office(s). Instead, Form PTO/SB/39 or PTO/SB/69 must be used as appropriate.

1. Authorization to Permit Access by a Foreign Intellectual Property Office(s)

A. Priority Document Exchange (PDX) - Unless box A in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the European Patent Office (EPO), the Japan Patent Office (JPO), the Korean Intellectual Property Office (KIPO), the State Intellectual Property Office of the People's Republic of China (SIPO), the World Intellectual Property Organization (WIPO), and any other foreign intellectual property office participating with the USPTO in a bilateral or multilateral priority document exchange agreement in which a foreign application claiming priority to the instant patent application is filed, access to: (1) the instant patent application-as-filed and its related bibliographic data, (2) any foreign or domestic application to which priority or benefit is claimed by the instant application and its related bibliographic data, and (3) the date of filing of this Authorization. See 37 CFR 1.14(h)(1).

B. Search Results from U.S. Application to EPO - Unless box B in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the EPO access to the bibliographic data and search results from the instant patent application when a European patent application claiming priority to the instant patent application is filed. See 37 CFR 1.14(h)(2).

The applicant is reminded that the EPO's Rule 141(1) EPC (European Patent Convention) requires applicants to submit a copy of search results from the instant application without delay in a European patent application that claims priority to the instant application.

2. Opt-Out of Authorizations to Permit Access by a Foreign Intellectual Property Office(s)

☐ A. Applicant **DOES NOT** authorize the USPTO to permit a participating foreign IP office access to the instant application-as-filed. If this box is checked, the USPTO will not be providing a participating foreign IP office with any documents and information identified in subsection 1A above.

☐ B. Applicant **DOES NOT** authorize the USPTO to transmit to the EPO any search results from the instant patent application. If this box is checked, the USPTO will not be providing the EPO with search results from the instant application.

NOTE: Once the application has published or is otherwise publicly available, the USPTO may provide access to the application in accordance with 37 CFR 1.14.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

Applicant Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Applicant 1

If the applicant is the inventor (or the remaining joint inventor or inventors under 37 CFR 1.45), this section should not be completed. The information to be provided in this section is the name and address of the legal representative who is the applicant under 37 CFR 1.43; or the name and address of the assignee, person to whom the inventor is under an obligation to assign the invention, or person who otherwise shows sufficient proprietary interest in the matter who is the applicant under 37 CFR 1.46. If the applicant is an applicant under 37 CFR 1.46 (assignee, person to whom the inventor is obligated to assign, or person who otherwise shows sufficient proprietary interest) together with one or more joint inventors, then the joint inventor or inventors who are also the applicant should be identified in this section.

Clear

(●) Assignee ( ) Legal Representative under 35 U.S.C. 117 ( ) Joint Inventor

( ) Person to whom the inventor is obligated to assign. ( ) Person who shows sufficient proprietary interest

If applicant is the legal representative, indicate the authority to file the patent application, the inventor is:

Name of the Deceased or Legally Incapacitated Inventor:

If the Applicant is an Organization check here. ☒

Organization Name	LI Creative Technologies, Inc.		
Mailing Address Information For Applicant:			
Address 1	25B Hanover Road, Suite 140		
Address 2			
City	Florham Park	**State/Province**	NJ
Country	US	Postal Code	07932
Phone Number		Fax Number	
Email Address			

Additional Applicant Data may be generated within this form by selecting the Add button.

Assignee Information including Non-Applicant Assignee Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	

Title of Invention	Microphone Array System

Assignee 1

Complete this section if assignee information, including non-applicant assignee information, is desired to be included on the patent application publication. An assignee-applicant identified in the "Applicant Information" section will appear on the patent application publication as an applicant. For an assignee-applicant, complete this section only if identification as an assignee is also desired on the patent application publication.

If the Assignee or Non-Applicant Assignee is an Organization check here. ☒

Organization Name	LI Creative Technologies, Inc.

Mailing Address Information For Assignee including Non-Applicant Assignee:

Address 1	25B Hanover Road, Suite 140		
Address 2			
City	Florham Park	**State/Province**	NJ
Country i	US	Postal Code	07932
Phone Number		Fax Number	
Email Address			

Additional Assignee or Non-Applicant Assignee Data may be generated within this form by selecting the Add button.

Signature:

NOTE: This Application Data Sheet must be signed in accordance with 37 CFR 1.33(b). **However, if this Application Data Sheet is submitted with the INITIAL filing of the application and either box A or B is not checked in subsection 2 of the "Authorization or Opt-Out of Authorization to Permit Access" section, then this form must also be signed in accordance with 37 CFR 1.14(c).**

This Application Data Sheet **must** be signed by a patent practitioner if one or more of the applicants is a **juristic entity** (e.g., corporation or association). If the applicant is two or more joint inventors, this form must be signed by a patent practitioner, **all** joint inventors who are the applicant, or one or more joint inventor-applicants who have been given power of attorney (e.g., see USPTO Form PTO/AIA/81) on behalf of **all** joint inventor-applicants.

See 37 CFR 1.4(d) for the manner of making signatures and certifications.

Signature	/a tankha/			Date (YYYY-MM-DD)	2016-10-14
First Name	Ashok	Last Name	Tankha	Registration Number	33802

Additional Signature may be generated within this form by selecting the Add button.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CreativeTech_01RE_US
	Application Number	
Title of Invention	Microphone Array System	

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 the Freedom of Information Act requires disclosure of these records.

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 CooperationTreaty.

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 inspections 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.

REISSUE APPLICATION DECLARATION BY THE ASSIGNEE	Docket Number (optional) CreativeTech_01RE_US

I hereby declare that:

The residence and mailing address of the inventor or joint inventors are stated below.

I am authorized to act on behalf of the following assignee: Li Creative Technologies, Inc.

The entire title to the patent identified below is vested in said assignee.

Inventor Li, Qi(Peter)

Residence: City	State	Country
New Providence	NJ	USA

Mailing Address
225 Runnymede Parkway

City	State	Zip	Country
New Providence	NJ	07974	USA

☑ Additional Inventors are named on separately numbered sheets attached hereto.

Patent Number US8861756	Date of Patent Issued 14 October, 2014

I believe said inventor(s) to be the original inventor or original joint inventors of the subject matter which is described and claimed in said patent, for which a reissue patent is sought on the invention titled:

Microphone array system

the specification of which

☑ is attached hereto.

☐ was filed on ______________ as reissue application number ____________.

The above-identified application was made or authorized to be made by me.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

I believe the original patent to be wholly or partly inoperative or invalid, for the reasons described below. (Check all boxes that apply.)

☐ by reason of a defective specification or drawing.

☑ by reason of the patentee claiming more or less than he had the right to claim in the patent.

☑ by reason of other errors.

REISSUE APPLICATION DECLARATION BY THE ASSIGNEE | Docket Number (Optional)

At least one error upon which reissue is based is described below. If the reissue is a broadening reissue, a claim that the application seeks to broaden must be identified and the box below must be checked:

The reissue is a broadening reissue.

[Attach additional sheets, if needed.]

☑ **The application for the original patent was filed under 37 CFR 1.46 by the assignee of the entire interest.**

I hereby appoint:

☐ Practitioners associated with Customer Number:

OR

☑ Practitioner(s) named below:

Name	Registration Number
Ashok Tankha	33802

as my/our attorney(s) or agent(s) to prosecute the application identified above, and to transact all business in the United States Patent and Trademark Office connected therewith.

Correspondence Address: Direct all communications about the application to:

☐ The address associated with Customer Number:

OR

☑ Firm or Individual Name	Ashok Tankha				
Address	36 Greenleigh drive				
City	Sewell	State	NJ	Zip	08080
Country	US				
Telephone	856-266-5145	Email	ash@ipprocurement.com		

WARNING:

Petitioner/applicant is cautioned to avoid submitting personal information in documents filed in a patent application that may contribute to identity theft. Personal information such as social security numbers, bank account numbers, or credit card numbers (other than a check or credit card authorization form PTO-2038 submitted for payment purposes) is never required by the USPTO to support a petition or an application. If this type of personal information is included in documents submitted to the USPTO, petitioners/applicants should consider redacting such personal information from the documents before submitting them to the USPTO. Petitioner/applicant is advised that the record of a patent application is available to the public after publication of the application (unless a non-publication request in compliance with 37 CFR 1.213(a) is made in the application) or issuance of a patent. Furthermore, the record from an abandoned application may also be available to the public if the application is referenced in a published application or an issued patent (see 37 CFR 1.14). Checks and credit card authorization forms PTO-2038 submitted for payment purposes are not retained in the application file and therefore are not publicly available.

Signature | Date (Optional) October 14, 2016

Full name of person signing (given name, family name) Qi Li

Address of Assignee New Providence, NJ, USA

[Page 2 of 2]

Privacy Act Statement

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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.

SUPPLEMENTAL SHEET FOR DECLARATION	ADDITIONAL INVENTOR(S) Supplemental Sheet (for PTO/AIA/08,09) Page 1 of 1

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)
Manli Zhu

Inventor's Signature Manli Zhu			Date (Optional) Oct 14, 2016
Residence: City Pearl River	State NY	Country USA	
Mailing Address 46 E Crooked Hill Road			
City Pearl River	State NY	Zip 1095	Country USA

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)

Inventor's Signature			Date (Optional)
Residence: City	State	Country	
Mailing Address			
City	State	Zip	Country

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)

Inventor's Signature			Date (Optional)
Residence: City	State	Country	
Mailing Address			
City	State	Zip	Country

POWER OF ATTORNEY BY APPLICANT

I hereby revoke all previous powers of attorney given in the application identified in either the attached transmittal letter or the boxes below.

Application Number	Filing Date

(Note: The boxes above may be left blank if information is provided on form PTO/AIA/82A.)

☐ I hereby appoint the Patent Practitioner(s) associated with the following Customer Number as my/our attorney(s) or agent(s), and to transact all business in the United States Patent and Trademark Office connected therewith for the application referenced in the attached transmittal letter (form PTO/AIA/82A) or identified above:

OR

☑ I hereby appoint Practitioner(s) named in the attached list (form PTO/AIA/82C) as my/our attorney(s) or agent(s), and to transact all business in the United States Patent and Trademark Office connected therewith for the patent application referenced in the attached transmittal letter (form PTO/AIA/82A) or identified above. (Note: Complete form PTO/AIA/82C.)

Please recognize or change the correspondence address for the application identified in the attached transmittal letter or the boxes above to:

☐ The address associated with the above-mentioned Customer Number

OR

☐ The address associated with Customer Number:

OR

Firm or Individual Name	Ashok Tankha				
Address	36 Greenleigh Drive				
City	Sewell	State	NJ	Zip	08080
Country	USA				
Telephone	856-266-5145	Email	ash@ipprocurement.com		

I am the Applicant (if the Applicant is a juristic entity, list the Applicant name in the box):

Li Creative Technologies, Inc.

☐ Inventor or Joint Inventor (title not required below)

☐ Legal Representative of a Deceased or Legally Incapacitated Inventor (title not required below)

☑ Assignee or Person to Whom the Inventor is Under an Obligation to Assign (provide signer's title if applicant is a juristic entity)

☐ Person Who Otherwise Shows Sufficient Proprietary Interest (e.g., a petition under 37 CFR 1.46(b)(2) was granted in the application or is concurrently being filed with this document) (provide signer's title if applicant is a juristic entity)

SIGNATURE of Applicant for Patent

The undersigned (whose title is supplied below) is authorized to act on behalf of the applicant (e.g., where the applicant is a juristic entity).

Signature		Date (Optional)	October 14, 2016
Name	Qi Li		
Title	President & Executive Officer		

NOTE: Signature - This form must be signed by the applicant in accordance with 37 CFR 1.33. See 37 CFR 1.4 for signature requirements and certifications. If more than one applicant, use multiple forms.

☐ Total of forms are submitted.

POWER OF ATTORNEY BY APPLICANT

No more than ten (10) patent practitioners total may be appointed as set forth below by name and registration number. This page need not be submitted if appointing the Patent Practitioner(s) associated with a Customer Number (see form PTO/AIA/82B):

Name	Registration Number
Ashok Tankha	33,802

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UNITED STATES
PATENT AND TRADEMARK OFFICE

Assignment abstract of title for Application 13049877

Invention title/Inventor	Patent	Publication	Application	PCT	International registration
Microphone Array System	8861756	20120076316	13049877		
Manli Zhu, Qi Li	Oct 14, 2014	Mar 29, 2012	Mar 16, 2011		

Assignments (1 of 1 total)

Assignment 1

Reel/frame	Execution date	Date recorded	Properties	Pages
026003/0985	Dec 17, 2010	Mar 21, 2011	1	3

Conveyance
ASSIGNMENT OF ASSIGNORS INTEREST (SEE DOCUMENT FOR DETAILS).

Assignors
ZHU, MANLI
LI, QI

Assignee
LI CREATIVE TECHNOLOGIES, INC.
25B HANOVER ROAD, SUITE 140
FLORHAM PARK, NEW JERSEY 07932

Correspondent
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

REISSUE APPLICATION: CONSENT OF ASSIGNEE; STATEMENT OF NON-ASSIGNMENT	Docket Number (Optional) CreativeTech_01RE_US

This is part of the application for a reissue patent based on the original patent identified below.

Name of Patentee(s)
Li Creative Technologies, Inc.

Patent Number	Date Patent Issued
US8861756	October 14, 2014

Title of Invention
Microphone array system

1. ☑ Filed herein is a statement under 37 CFR 3.73(c). (Form PTO/AIA/96)

2. ☐ Ownership of the patent is in the inventor(s), and no assignment of the patent is in effect.

One of boxes 1 or 2 above must be checked. If multiple assignees, complete this form for each assignee. If box 2 is checked, skip the next entry and go directly to "Name of Assignee."

The written consent of all assignees and inventors owning an undivided interest in the original patent is included in this application for reissue.

The assignee(s) owning an undivided interest in said original patent is/are Li Creative Technologies, Inc., and the assignee(s) consents to the accompanying application for reissue.

Name of assignee/inventor (if not assigned)
Li Creative Technologies, Inc.

Signature	Date
	October 14, 2016

Typed or printed name and title of person signing for assignee (if assigned)

Qi(Peter) Li, President

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STATEMENT UNDER 37 CFR 3.73(c)

Applicant/Patent Owner: Li Creative Technologies, Inc.

Application No./Patent No.: Reissue of Pat no. 8861756 Filed/Issue Date: Oct 14, 2014

Titled: Microphone array system

LI Creative Technologies, Inc. , a corporation

(Name of Assignee) (Type of Assignee, e.g., corporation, partnership, university, government agency, etc.)

states that, for the patent application/patent identified above, it is (choose **one** of options 1, 2, 3 or 4 below):

1. ☑ The assignee of the entire right, title, and interest.

2. ☐ An assignee of less than the entire right, title, and interest (check applicable box):

   ☐ The extent (by percentage) of its ownership interest is ____________%. Additional Statement(s) by the owners holding the balance of the interest must be submitted to account for 100% of the ownership interest.

   ☐ There are unspecified percentages of ownership. The other parties, including inventors, who together own the entire right, title and interest are:

   Additional Statement(s) by the owner(s) holding the balance of the interest must be submitted to account for the entire right, title, and interest.

3. ☐ The assignee of an undivided interest in the entirety (a complete assignment from one of the joint inventors was made). The other parties, including inventors, who together own the entire right, title, and interest are:

   Additional Statement(s) by the owner(s) holding the balance of the interest must be submitted to account for the entire right, title, and interest.

4. ☐ The recipient, via a court proceeding or the like (*e.g.*, bankruptcy, probate), of an undivided interest in the entirety (a complete transfer of ownership interest was made). The certified document(s) showing the transfer is attached.

The interest identified in option 1, 2 or 3 above (not option 4) is evidenced by either (choose **one** of options A or B below):

A. ☑ An assignment from the inventor(s) of the patent application/patent identified above. The assignment was recorded in the United States Patent and Trademark Office at Reel 026003, Frame 0985, or for which a copy thereof is attached.

B. ☐ A chain of title from the inventor(s), of the patent application/patent identified above, to the current assignee as follows:

   1. From: ____________________ To: ____________________

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      Reel ____________, Frame ____________, or for which a copy thereof is attached.

   2. From: ____________________ To: ____________________

      The document was recorded in the United States Patent and Trademark Office at

      Reel ____________, Frame ____________, or for which a copy thereof is attached.

STATEMENT UNDER 37 CFR 3.73(c)

3. From: ____________________ To: ____________________

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The document was recorded in the United States Patent and Trademark Office at

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The document was recorded in the United States Patent and Trademark Office at

Reel __________, Frame __________, or for which a copy thereof is attached.

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The document was recorded in the United States Patent and Trademark Office at

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☐ Additional documents in the chain of title are listed on a supplemental sheet(s).

☐ As required by 37 CFR 3.73(c)(1)(i), the documentary evidence of the chain of title from the original owner to the assignee was, or concurrently is being, submitted for recordation pursuant to 37 CFR 3.11.

[NOTE: A separate copy (i.e., a true copy of the original assignment document(s)) must be submitted to Assignment Division in accordance with 37 CFR Part 3, to record the assignment in the records of the USPTO. See MPEP 302.08]

The undersigned (whose title is supplied below) is authorized to act on behalf of the assignee.

____________________	October 14, 2016
Signature	Date
Li, Qi (Peter)	President
Printed or Typed Name	Title or Registration Number

[Page 2 of 2]

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US008861756B2

(12) **United States Patent**
Zhu et al.

(10) **Patent No.: US 8,861,756 B2**
(45) **Date of Patent: Oct. 14, 2014**

(54) **MICROPHONE ARRAY SYSTEM**

(75) Inventors: **Manli Zhu**, Pearl River, NY (US); **Qi Li**, New Providence, NJ (US)

(73) Assignee: **LI Creative Technologies, Inc.**, Florham Park, NJ (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 794 days.

(21) Appl. No.: **13/049,877**

(22) Filed: **Mar. 16, 2011**

(65) **Prior Publication Data**

US 2012/0076316 A1 Mar. 29, 2012

Related U.S. Application Data

(60) Provisional application No. 61/403,952, filed on Sep. 24, 2010.

(51) **Int. Cl.**

H04R 25/00	(2006.01)
H03G 3/20	(2006.01)
H04R 3/00	(2006.01)
G01S 5/22	(2006.01)
G01S 3/801	(2006.01)
G01S 3/805	(2006.01)
H04R 1/40	(2006.01)
H04M 3/56	(2006.01)

(52) **U.S. Cl.**
CPC ***G01S 3/8055*** (2013.01); *H04M 3/568* (2013.01); ***H04R 3/005*** (2013.01); ***G01S 5/22*** (2013.01); *H04R 2201/403* (2013.01); ***G01S 3/801*** (2013.01); *H04R 2201/401* (2013.01); ***H04R 1/406*** (2013.01)
USPC **381/300**; 381/57

(58) **Field of Classification Search**
CPC G01S 3/80; G01S 3/801; G01S 3/8055; G01S 5/22; H04R 1/406; H04R 3/005; H04R 2201/401; H04R 2201/403
USPC 381/92, 94.1, 93
See application file for complete search history.

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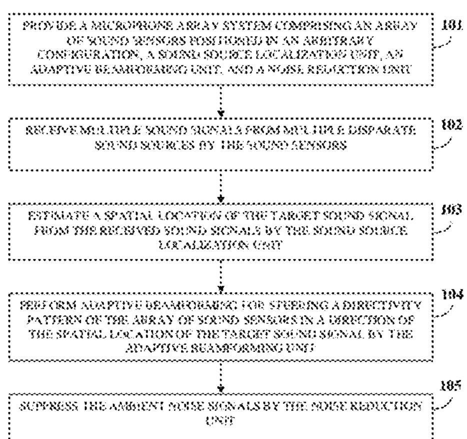
* cited by examiner

Primary Examiner — Fan Tsang
Assistant Examiner — Eugene Zhao
(74) *Attorney, Agent, or Firm* — Ash Tankha; Lipton, Weinberger & Husick

(57) **ABSTRACT**

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

21 Claims, 34 Drawing Sheets



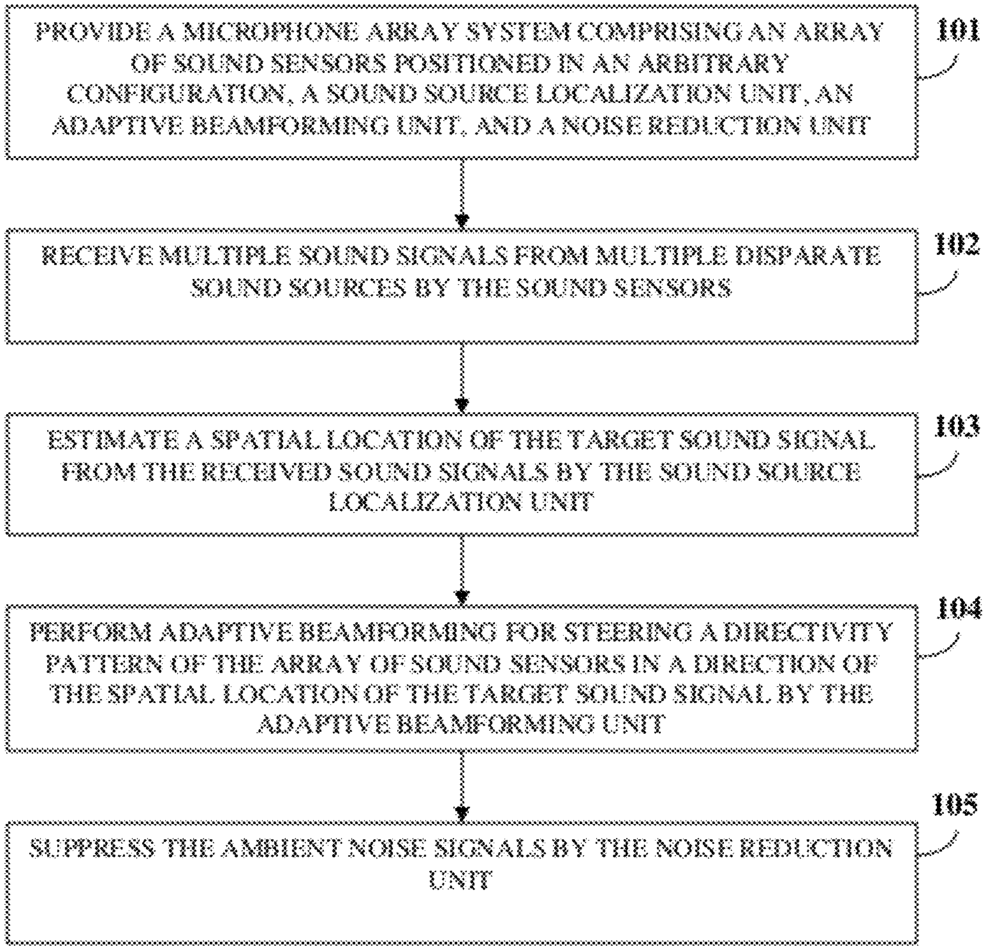

PROVIDE A MICROPHONE ARRAY SYSTEM COMPRISING AN ARRAY OF SOUND SENSORS POSITIONED IN AN ARBITRARY CONFIGURATION, A SOUND SOURCE LOCALIZATION UNIT, AN ADAPTIVE BEAMFORMING UNIT, AND A NOISE REDUCTION UNIT
101
RECEIVE MULTIPLE SOUND SIGNALS FROM MULTIPLE DISPARATE SOUND SOURCES BY THE SOUND SENSORS
102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

FIG. 1

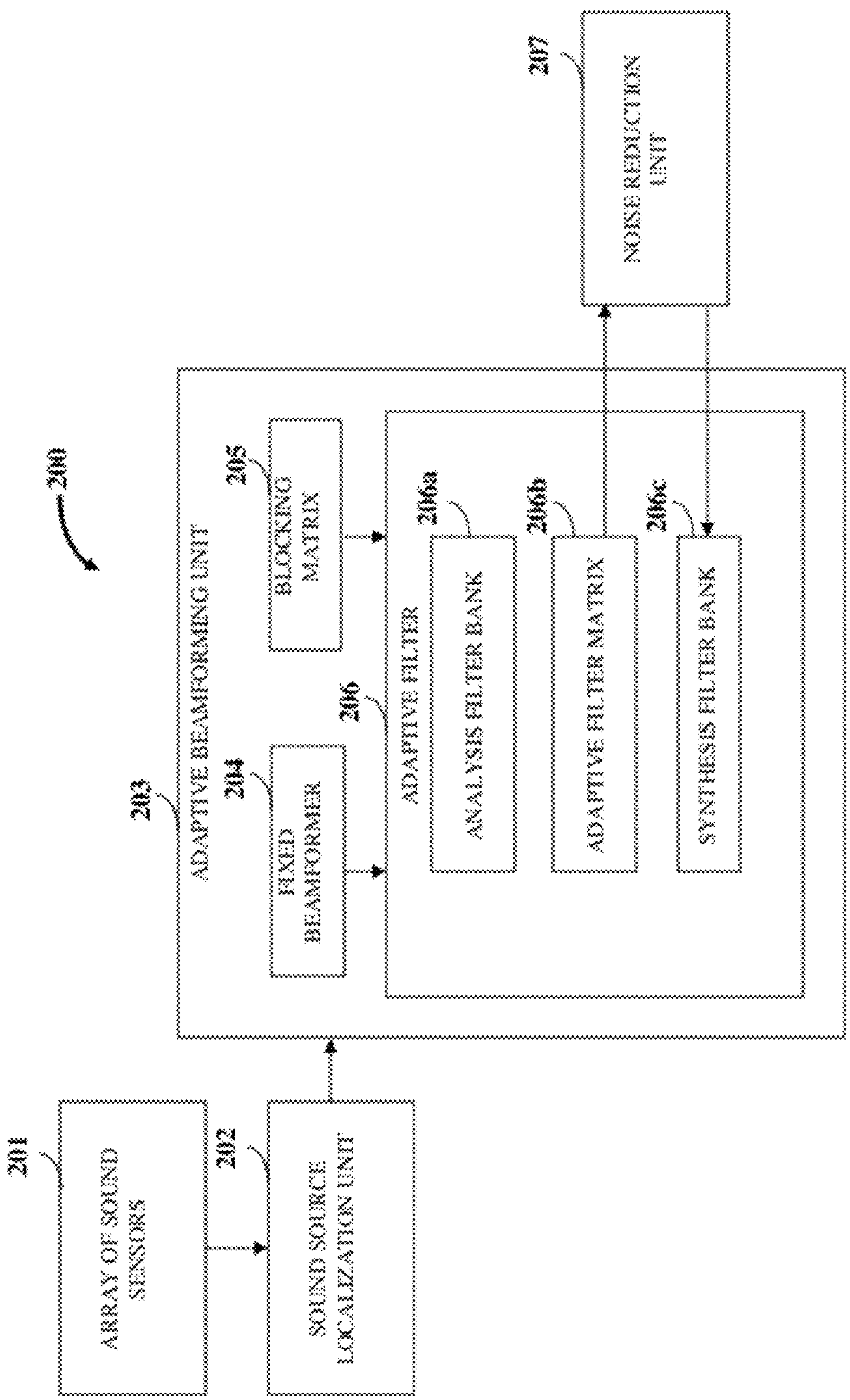

200
201
ARRAY OF SOUND SENSORS
202
SOUND SOURCE LOCALIZATION UNIT
203
ADAPTIVE BEAMFORMING UNIT
204
FIXED BEAMFORMER
205
BLOCKING MATRIX
206
ADAPTIVE FILTER
206a
ANALYSIS FILTER BANK
206b
ADAPTIVE FILTER MATRIX
206c
SYNTHESIS FILTER BANK
207
NOISE REDUCTION UNIT

FIG. 2

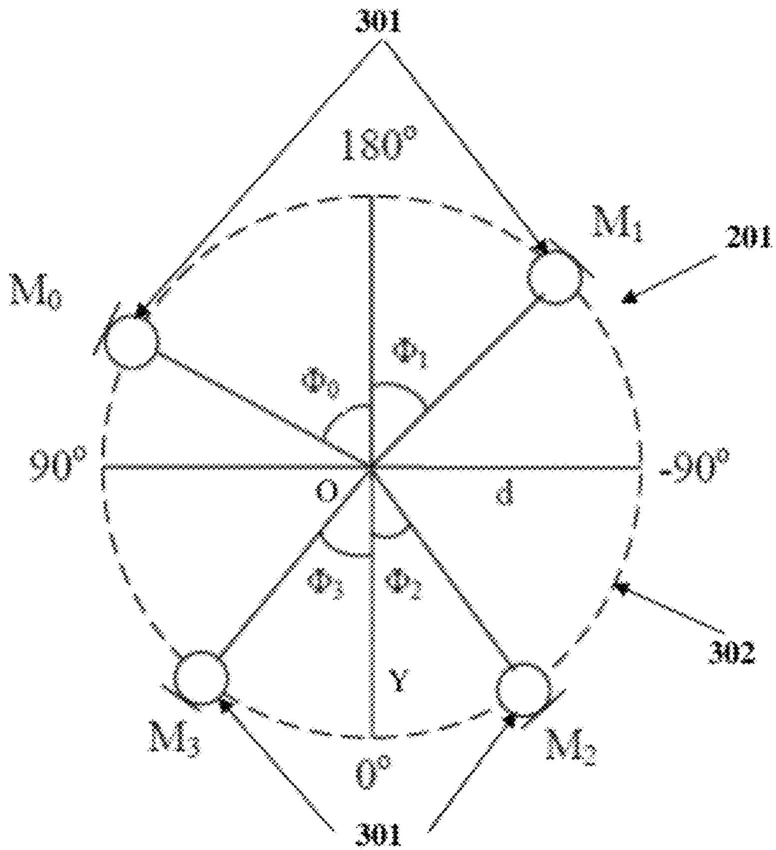


FIG. 3

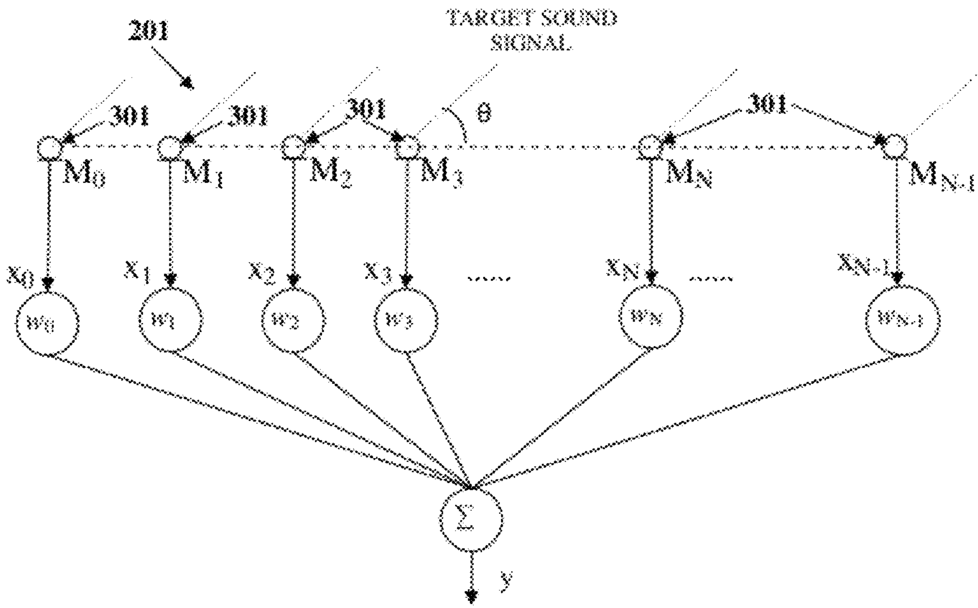

TARGET SOUND SIGNAL
201
301
301
301
301
θ
M_0
M_1
M_2
M_3
M_N
M_{N-1}
x_0
x_1
x_2
x_3
x_N
x_{N-1}
w_0
w_1
w_2
w_3
w_N
w_{N-1}
Σ
y

FIG. 4

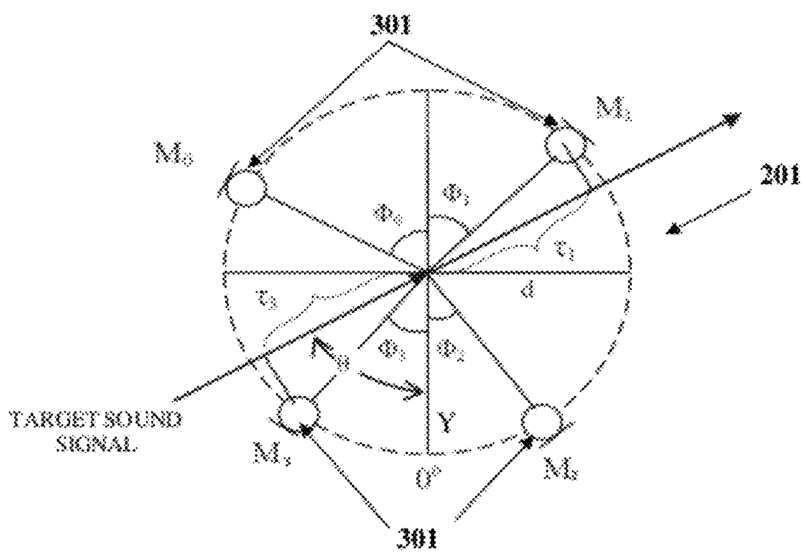

301
M_0
M_1
201
Φ_0
Φ_1
τ_1
τ_3
d
Φ_3
Φ_2
θ
TARGET SOUND SIGNAL
Y
M_3
M_2
0°
301

FIG. 5

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

FIG. 6A

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

FIG. 6B

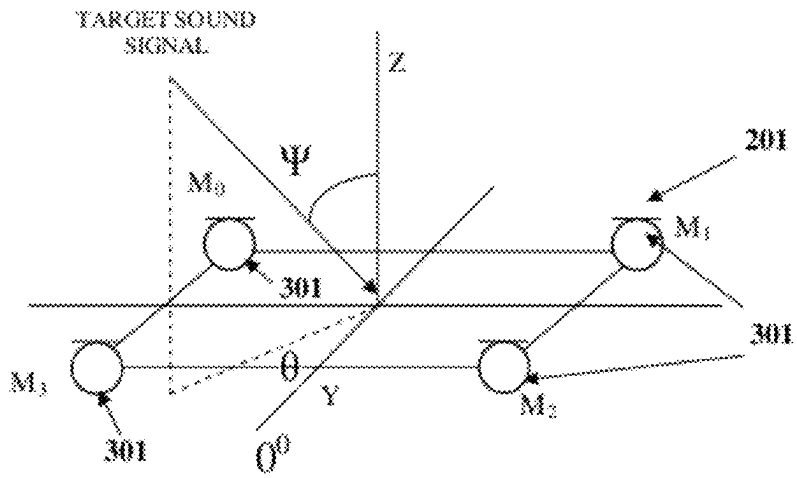


FIG. 7A

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

FIG. 7B

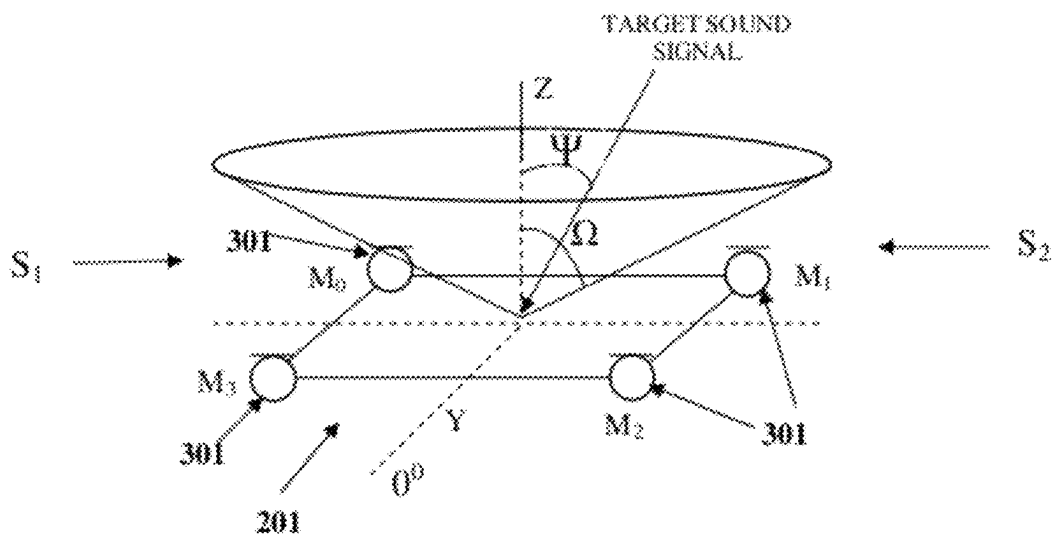


FIG. 7C

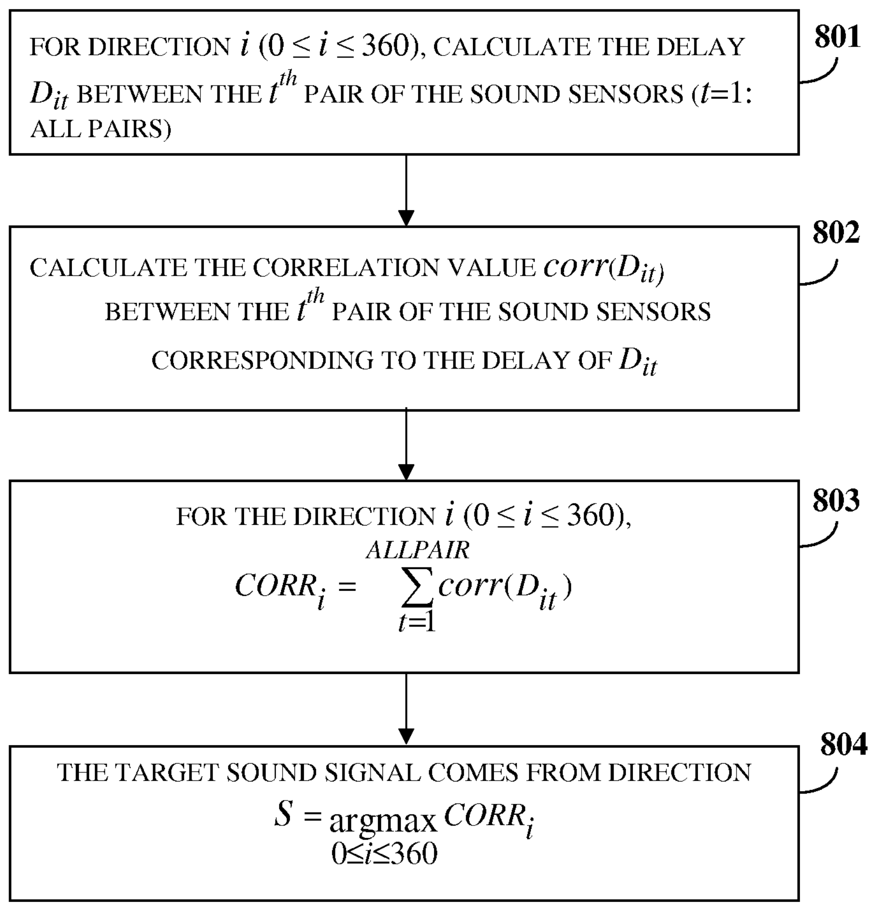

FOR DIRECTION i ($0 \le i \le 360$), CALCULATE THE DELAY D_{it} BETWEEN THE t^{th} PAIR OF THE SOUND SENSORS (t=1: ALL PAIRS)
801
CALCULATE THE CORRELATION VALUE $corr(D_{it})$ BETWEEN THE t^{th} PAIR OF THE SOUND SENSORS CORRESPONDING TO THE DELAY OF D_{it}
802
FOR THE DIRECTION i ($0 \le i \le 360$),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

FIG. 8

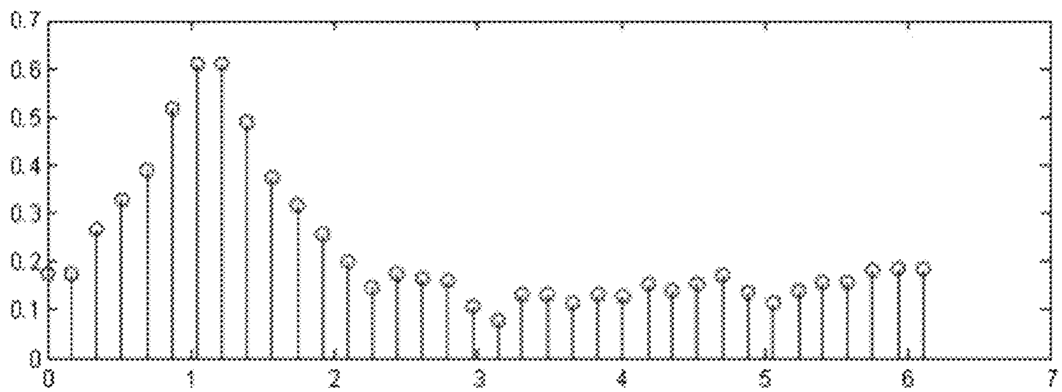

0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
1
2
3
4
5
6
7

FIG. 9A

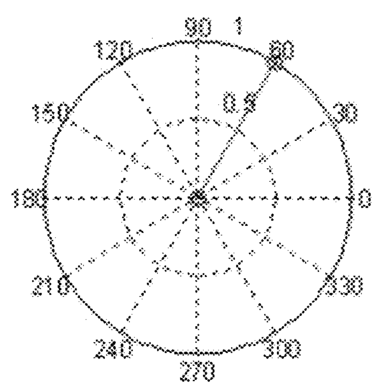

90
1
120
60
0.5
150
30
180
0
210
330
240
300
270

FIG. 9B

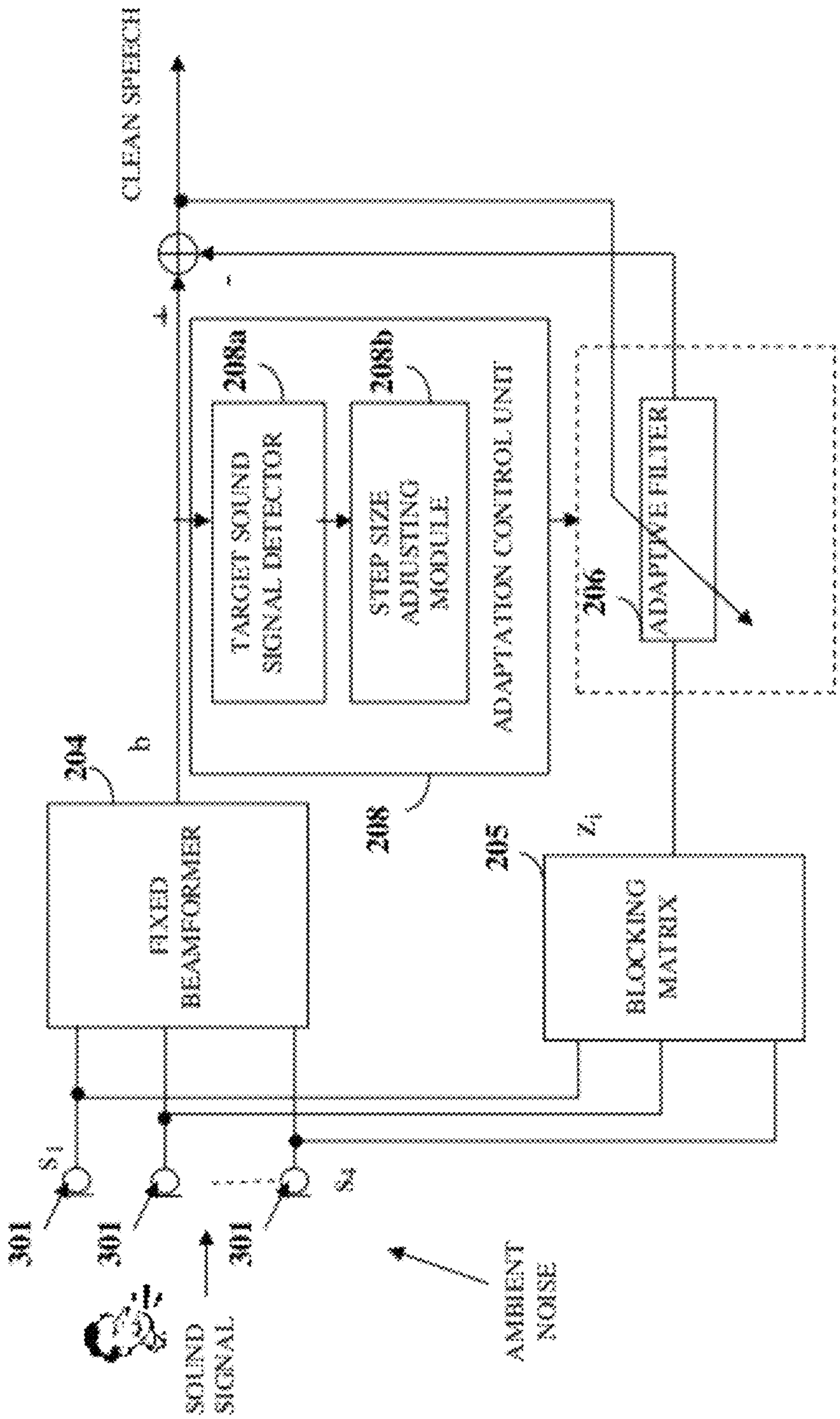

CLEAN SPEECH
+
-
208a
208b
TARGET SOUND
SIGNAL DETECTOR
STEP SIZE
ADJUSTING
MODULE
ADAPTATION CONTROL UNIT
ADAPTIVE FILTER
206
204
b
208
205
z_i
FIXED
BEAMFORMER
BLOCKING
MATRIX
S_1
S_M
301
301
301
SOUND
SIGNAL
AMBIENT
NOISE

FIG. 10

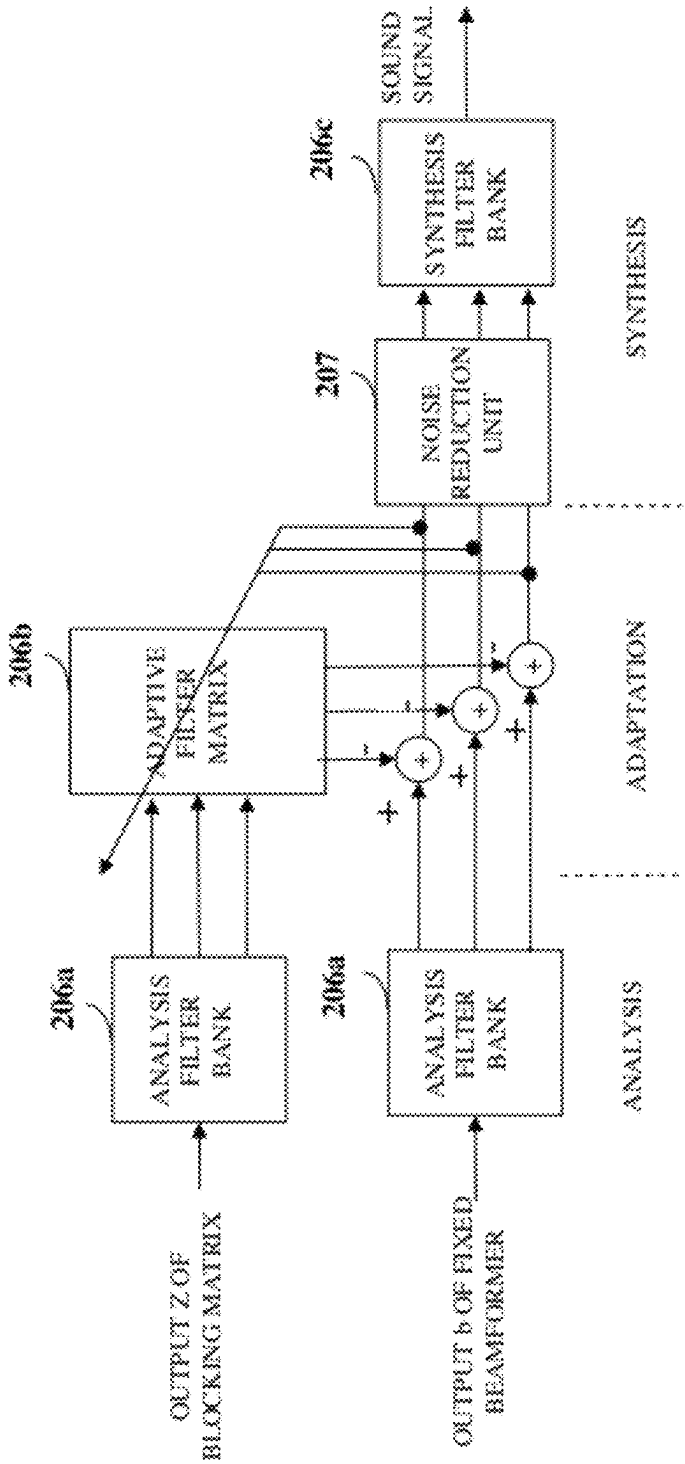

OUTPUT Z OF BLOCKING MATRIX
OUTPUT b OF FIXED BEAMFORMER
206a
ANALYSIS FILTER BANK
206a
ANALYSIS FILTER BANK
206b
ADAPTIVE FILTER MATRIX
207
NOISE REDUCTION UNIT
206c
SYNTHESIS FILTER BANK
SOUND SIGNAL
ANALYSIS
ADAPTATION
SYNTHESIS

FIG. 11

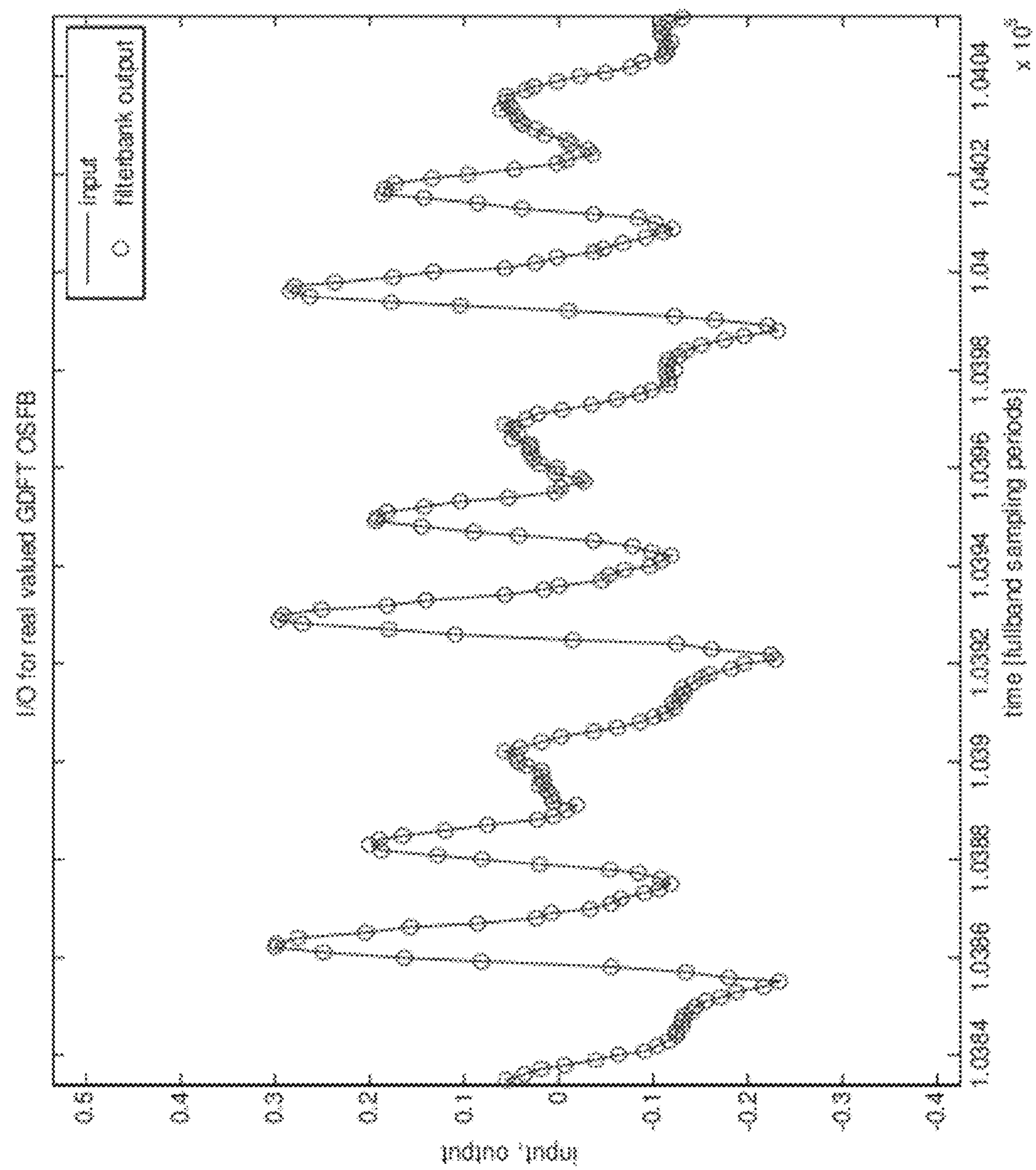

I/O for real valued GDFT OSFB
input
filterbank output
input, output
time [fullband sampling periods]
x 10^4

FIG. 12

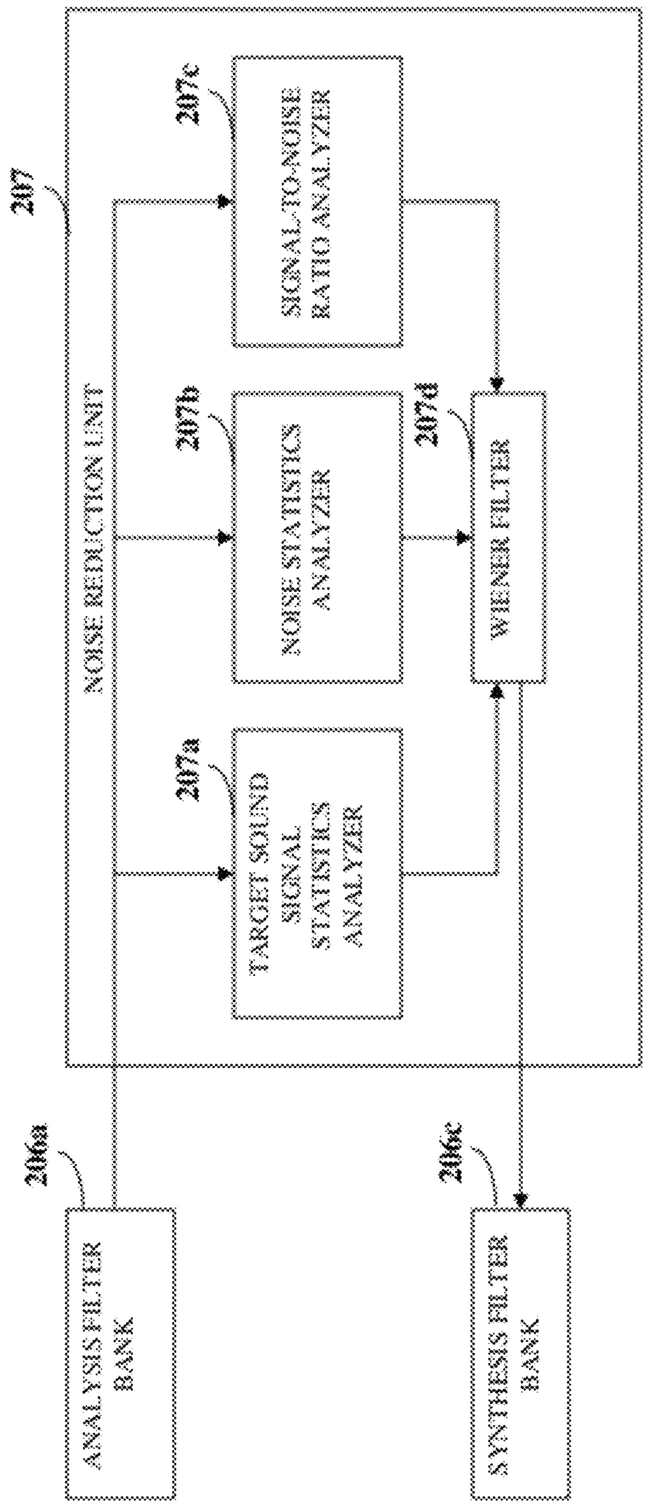

207
NOISE REDUCTION UNIT
207a
TARGET SOUND SIGNAL STATISTICS ANALYZER
207b
NOISE STATISTICS ANALYZER
207c
SIGNAL-TO-NOISE RATIO ANALYZER
207d
WIENER FILTER
206a
ANALYSIS FILTER BANK
206c
SYNTHESIS FILTER BANK

FIG. 13

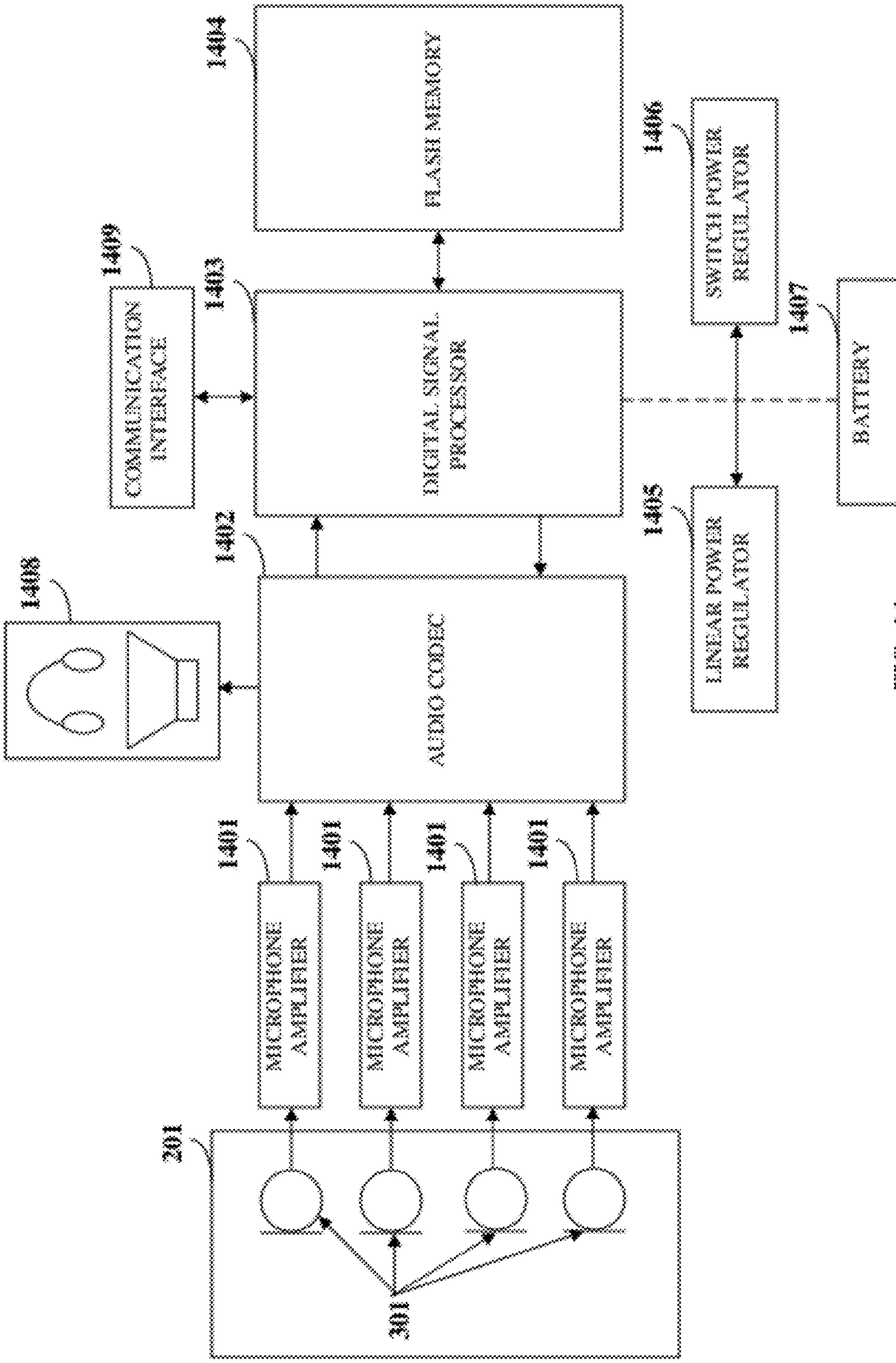

1404
FLASH MEMORY
1409
COMMUNICATION INTERFACE
1403
DIGITAL SIGNAL PROCESSOR
1406
SWITCH POWER REGULATOR
1407
BATTERY
1408
1402
AUDIO CODEC
1405
LINEAR POWER REGULATOR
1401
MICROPHONE AMPLIFIER
1401
MICROPHONE AMPLIFIER
1401
MICROPHONE AMPLIFIER
1401
MICROPHONE AMPLIFIER
201
301

FIG. 14

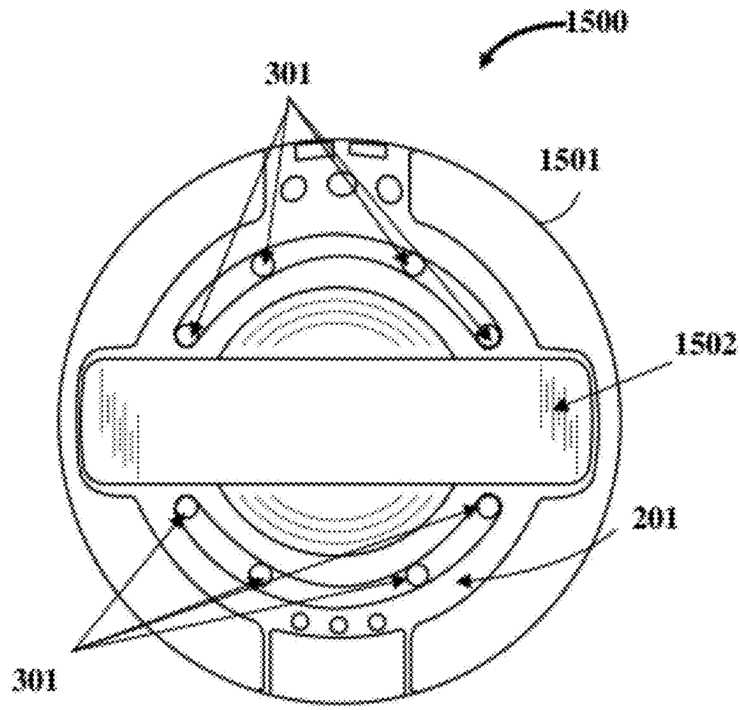


FIG. 15A

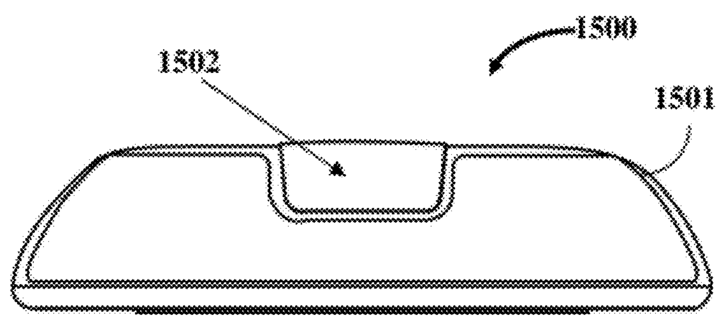


FIG. 15B

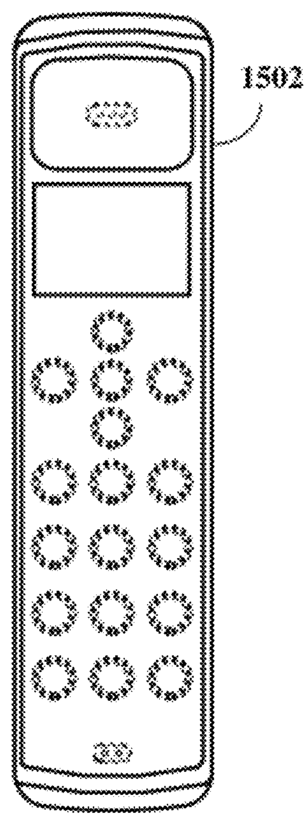


FIG. 15C

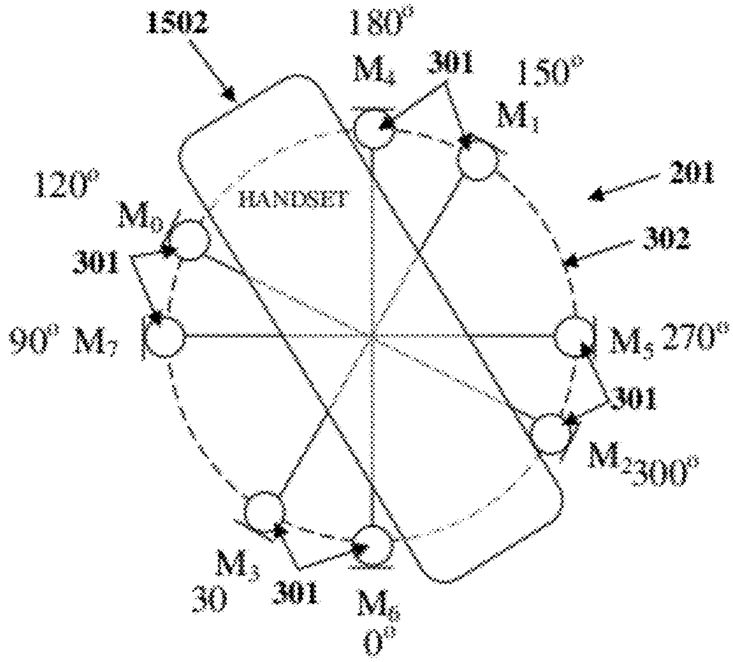


FIG. 16A

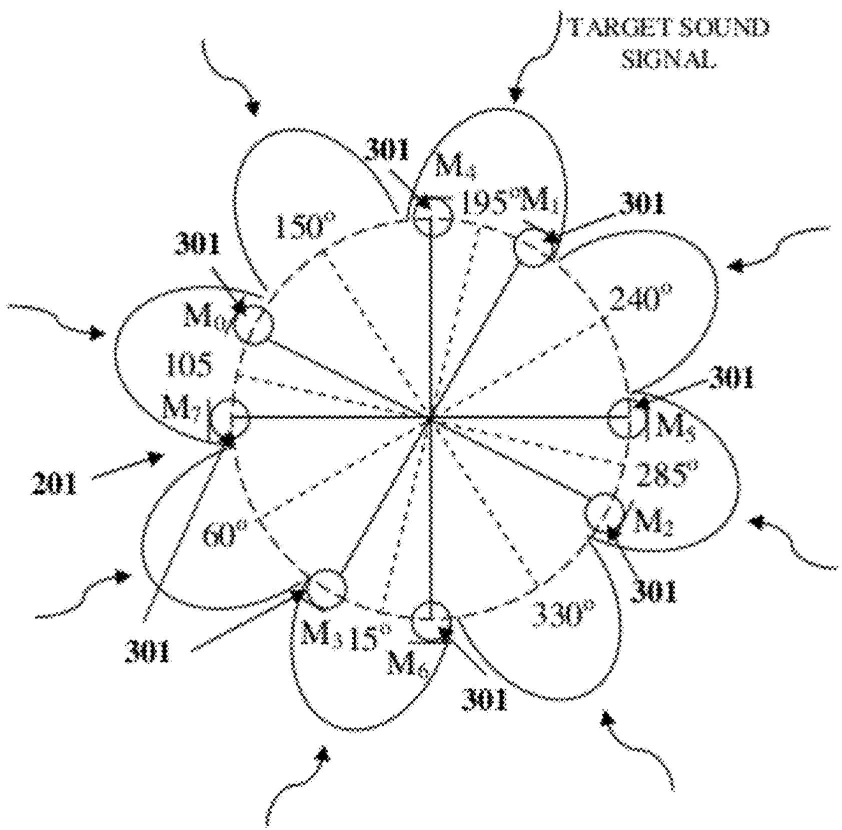


FIG. 16B

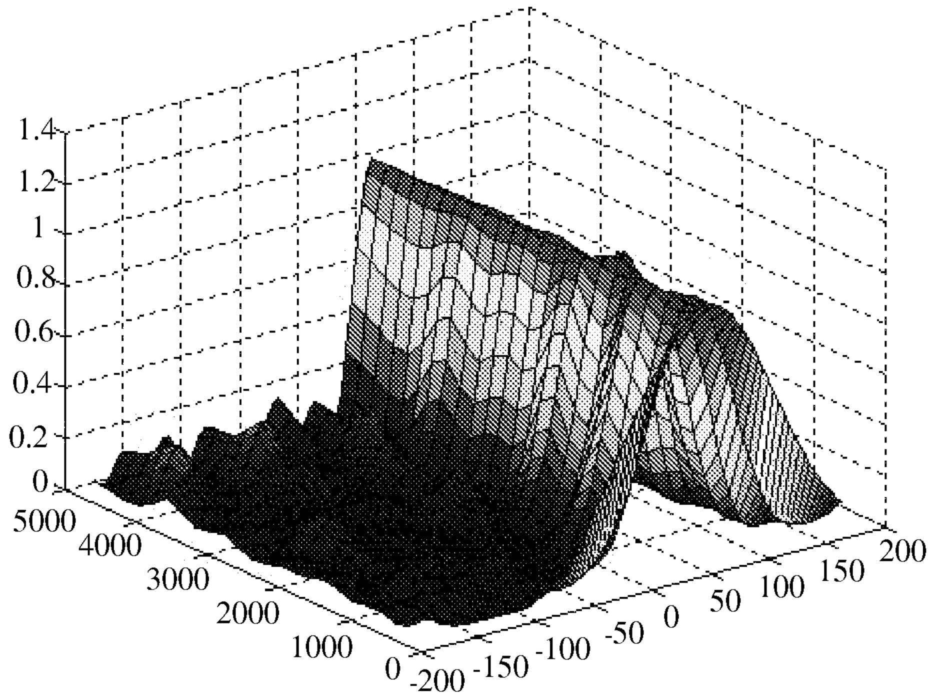


FIG. 16C

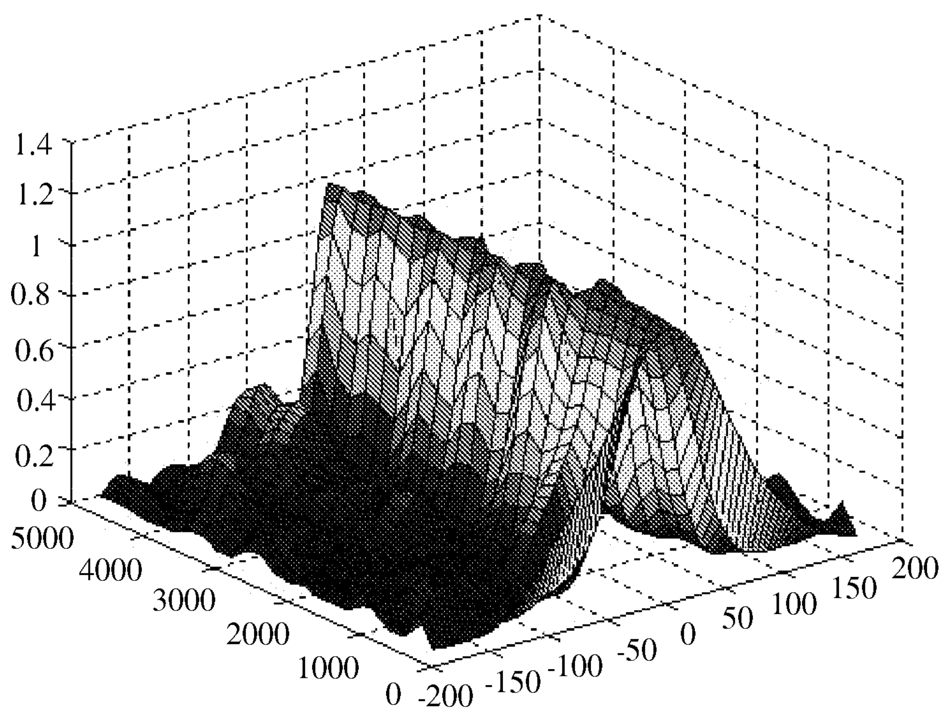


FIG. 16D

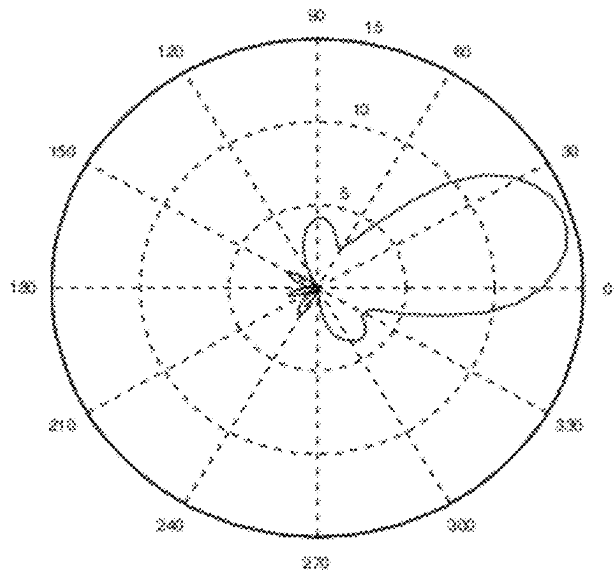


FIG. 16E

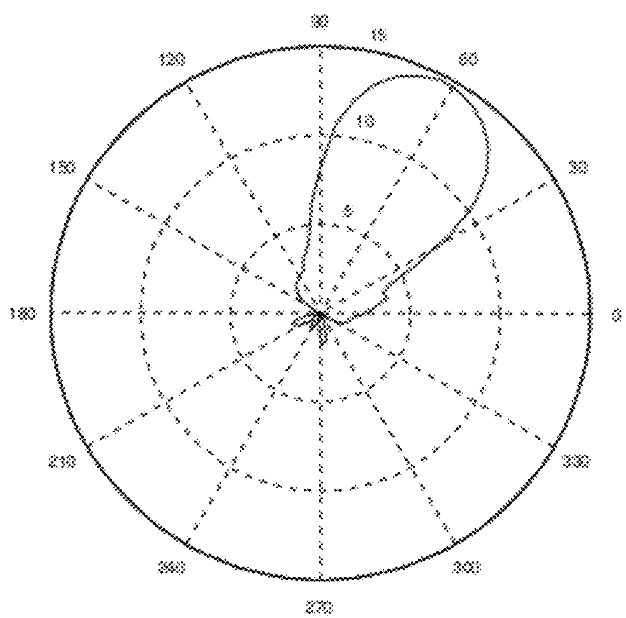


FIG. 16F

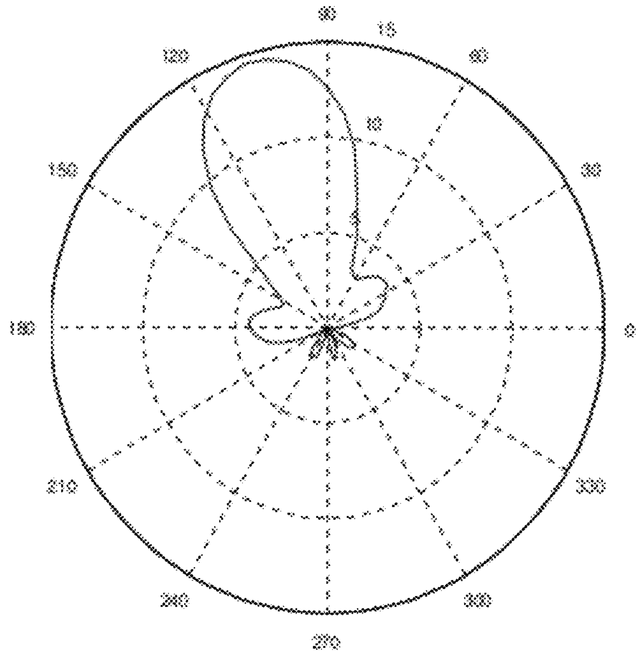


FIG. 16G

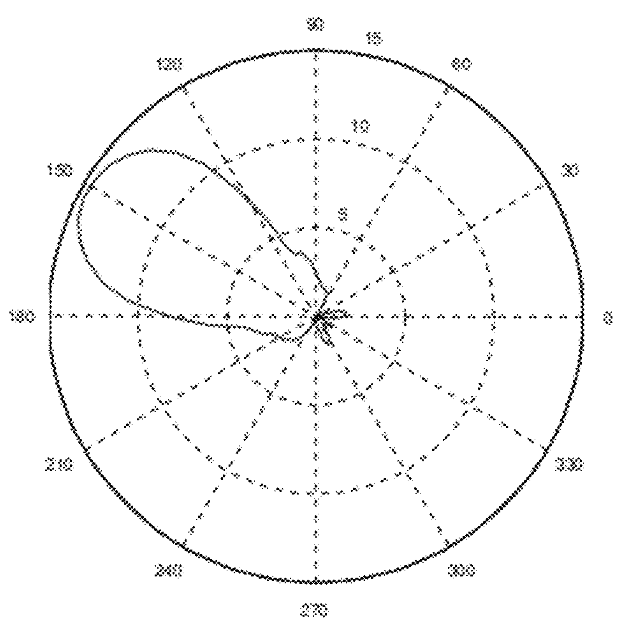


FIG. 16H

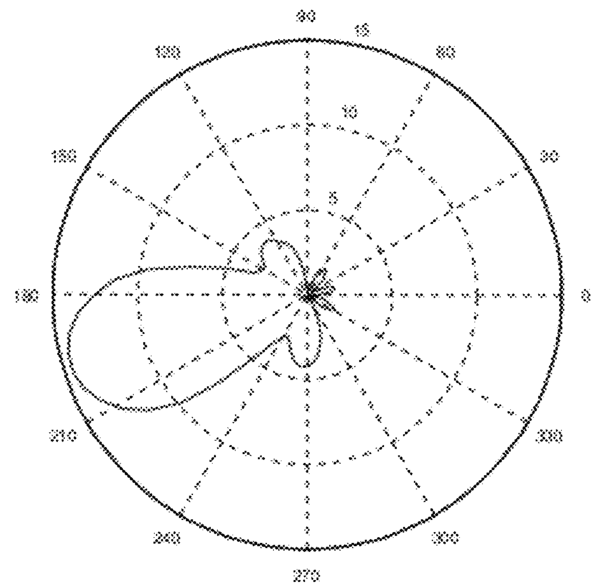


FIG. 16I

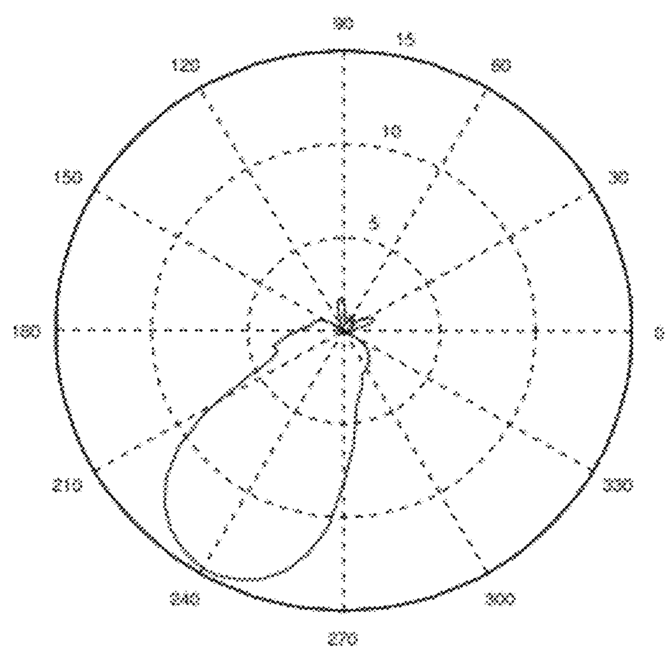


FIG. 16J

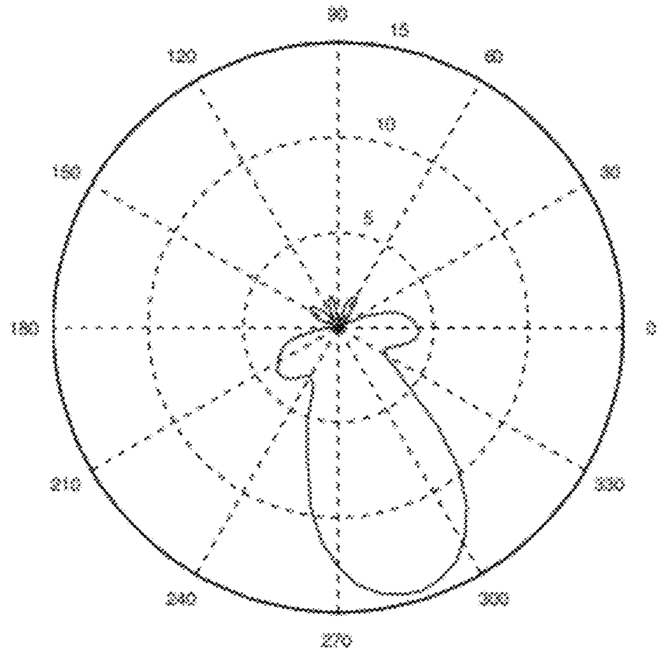


FIG. 16K

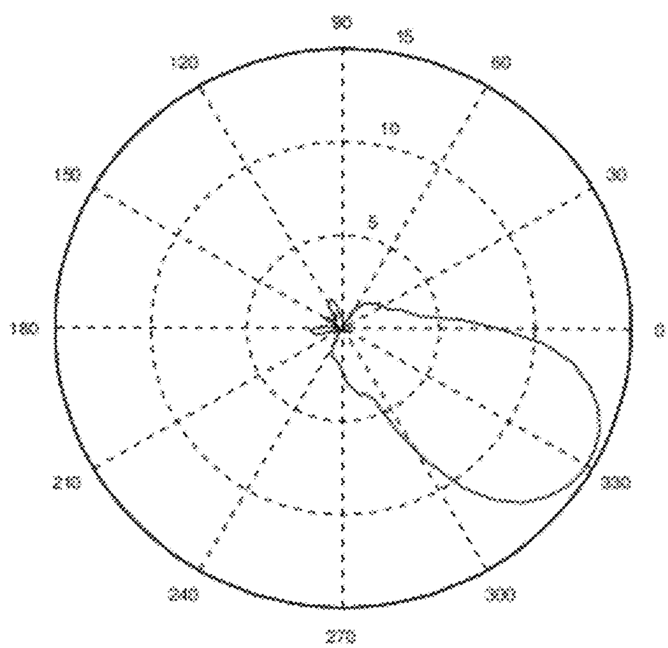


FIG. 16L

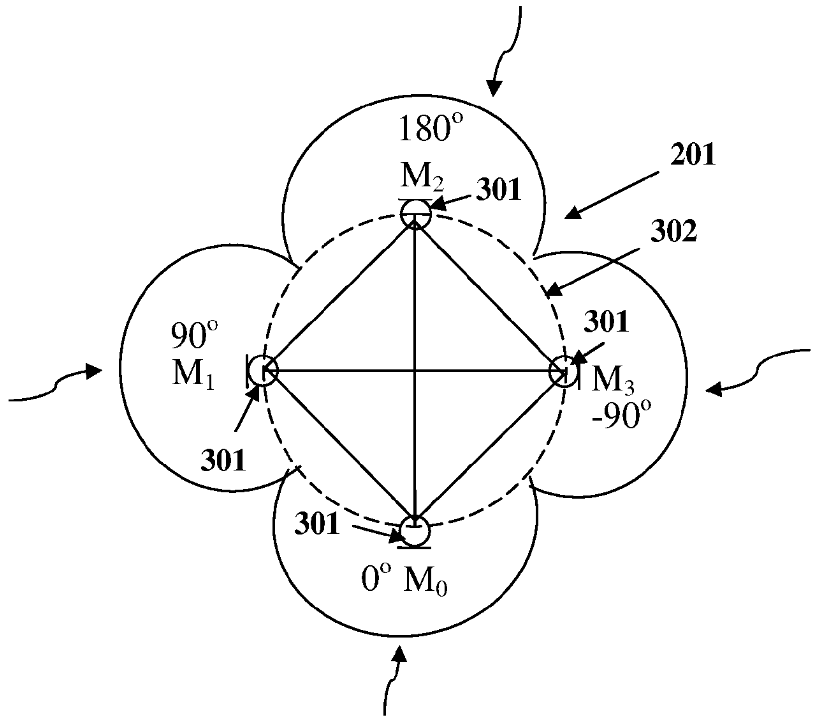


FIG. 17A

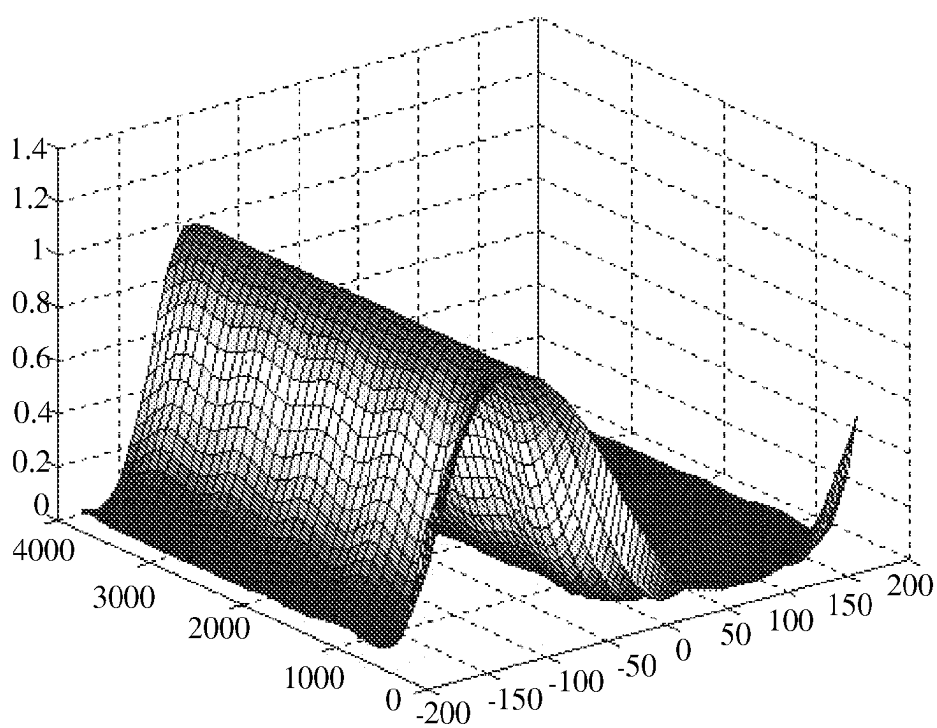


FIG. 17B

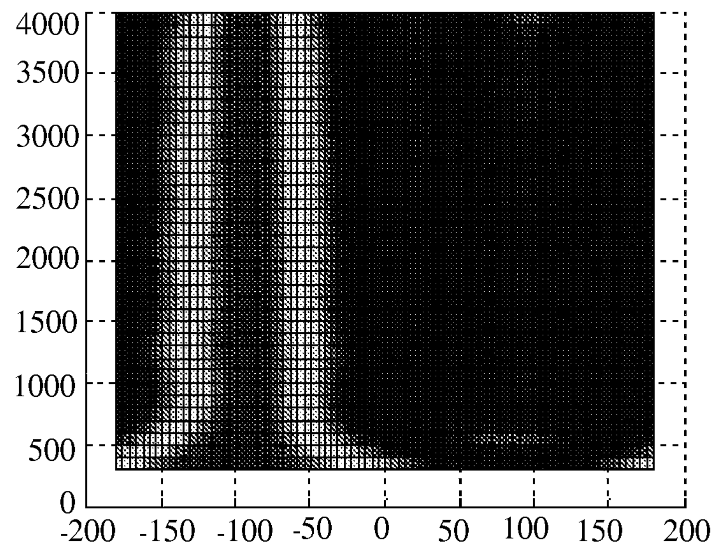


FIG. 17C

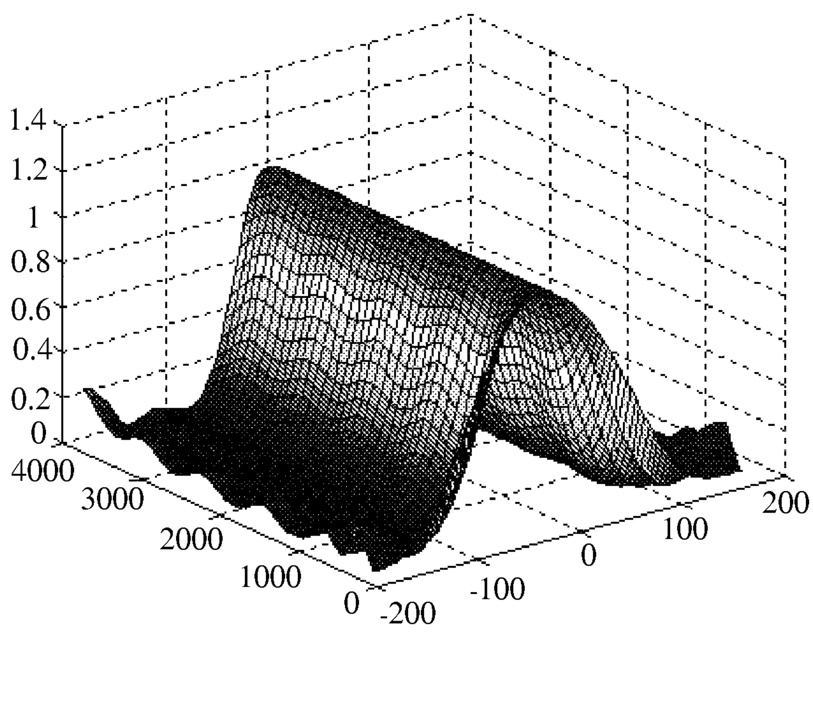


FIG. 17D

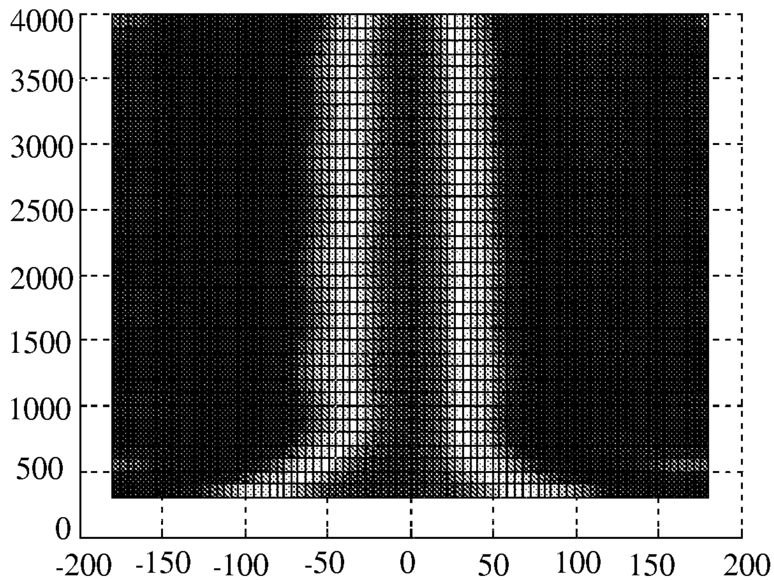

4000
3500
3000
2500
2000
1500
1000
500
0
-200
-150
-100
-50
0
50
100
150
200

FIG. 17E

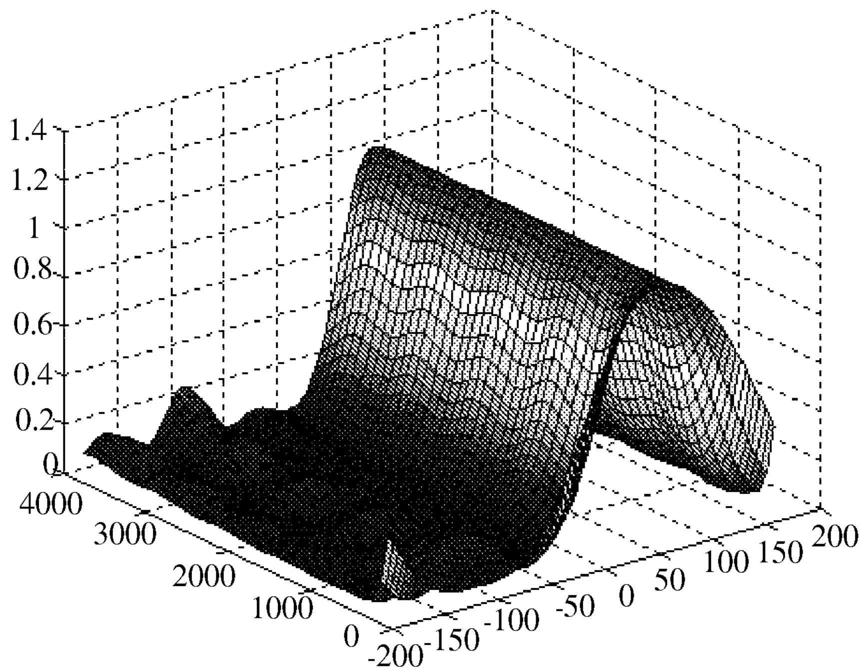


FIG. 17F

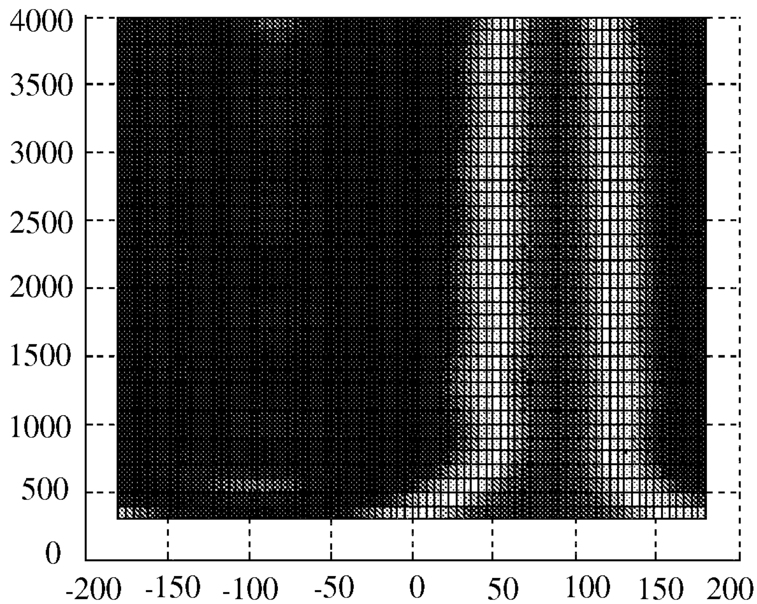


FIG. 17G

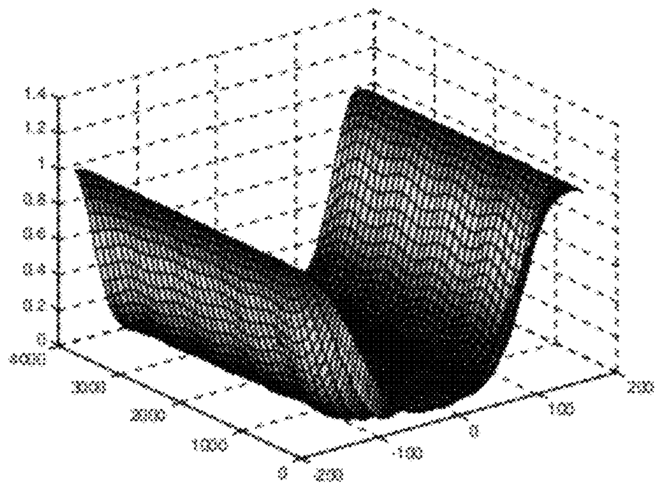


FIG. 17H

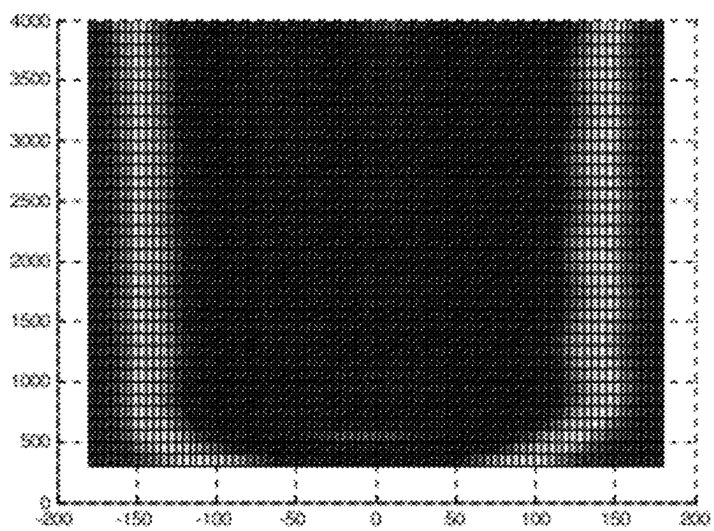


FIG. 17I

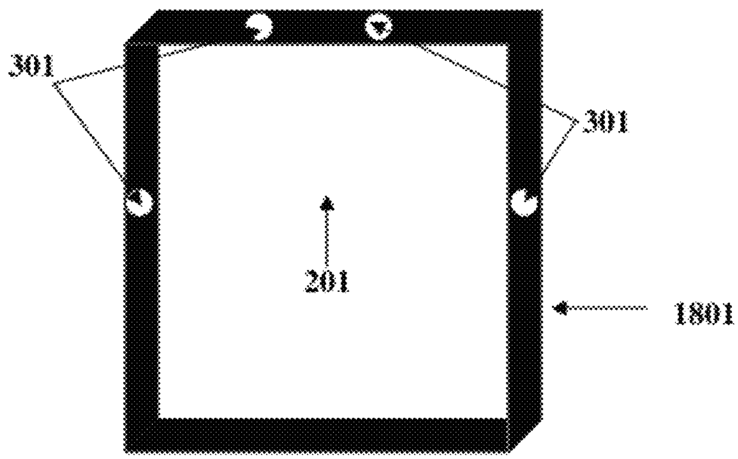


FIG. 18A

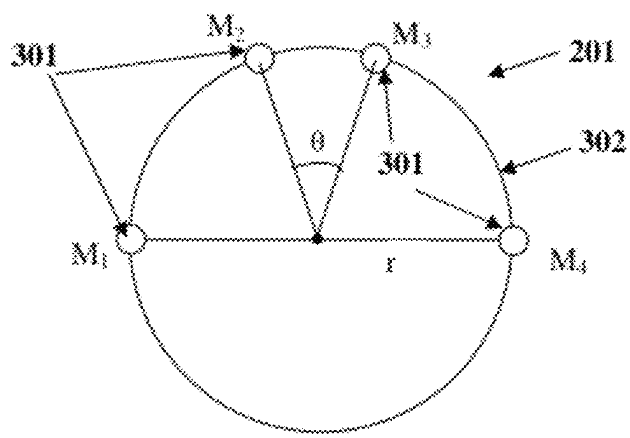


FIG. 18B

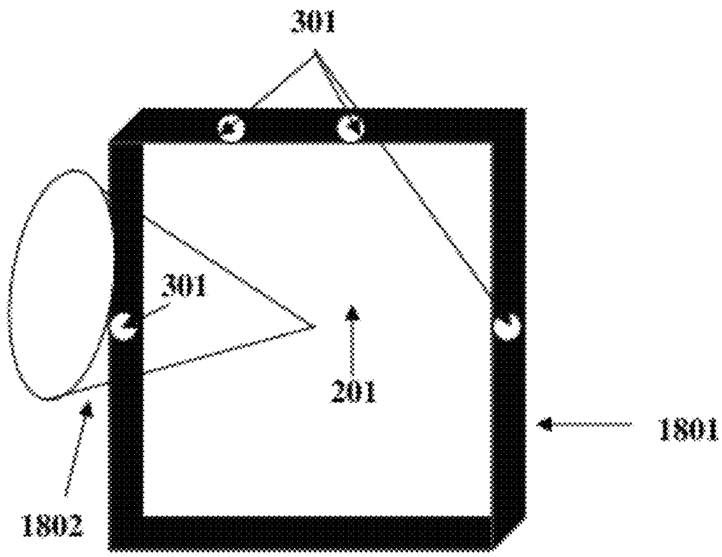


FIG. 18C

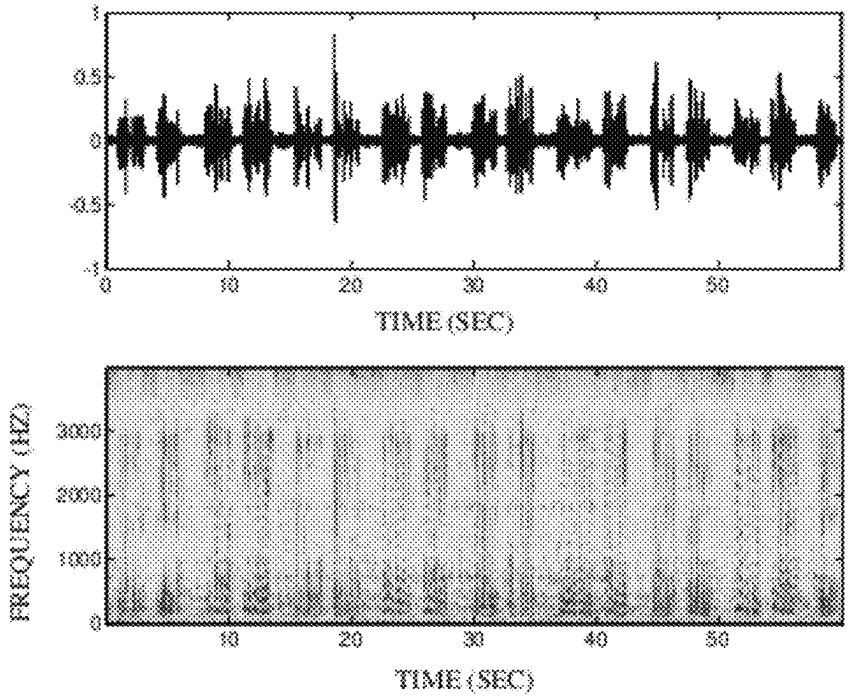


FIG. 18D

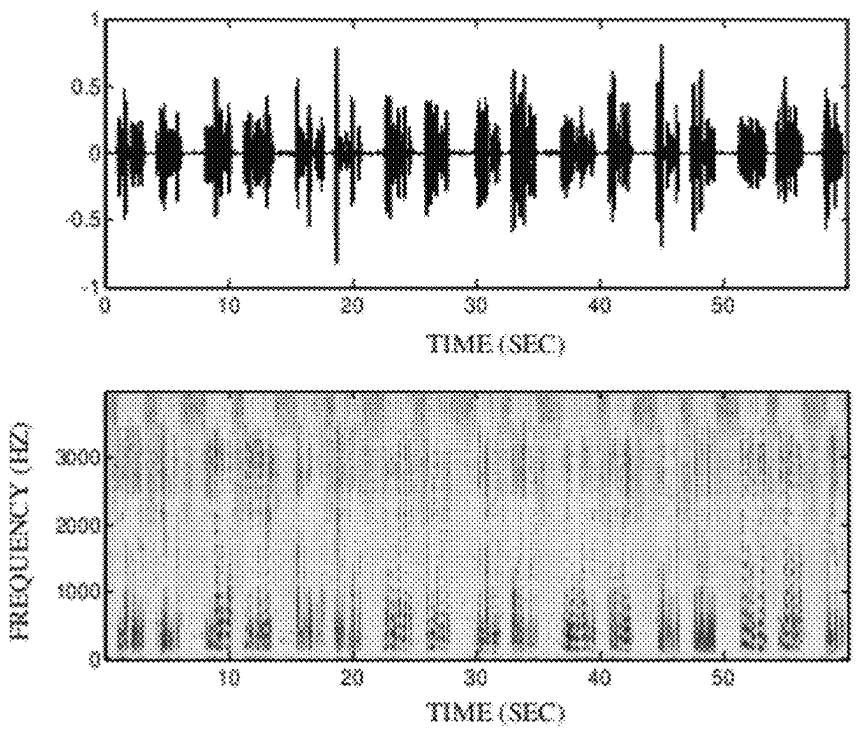


FIG. 18E

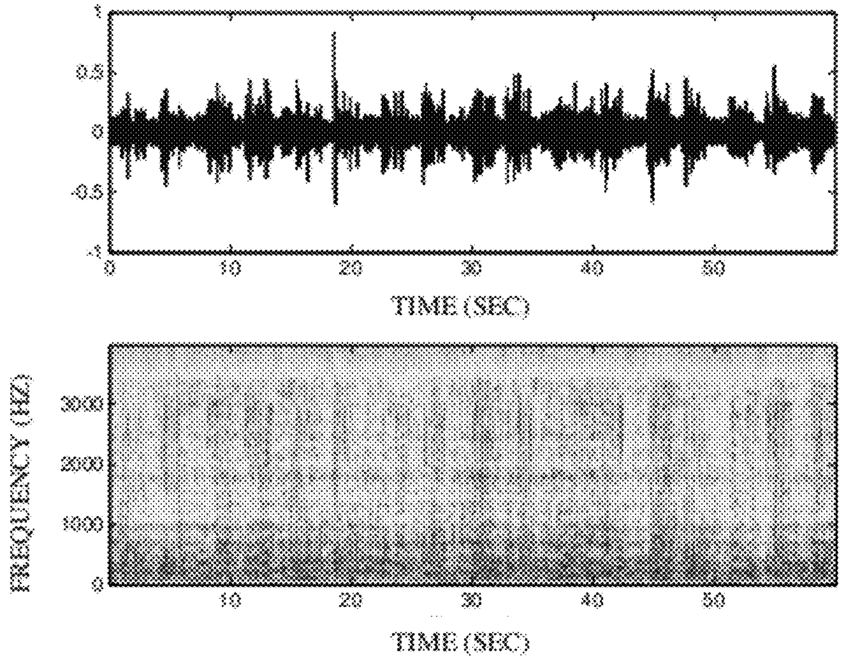


FIG. 18F

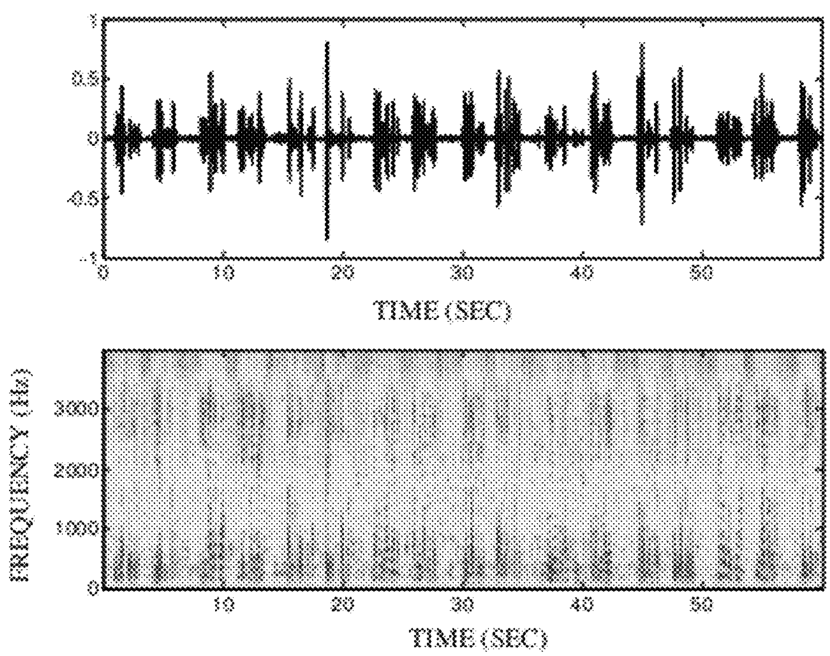
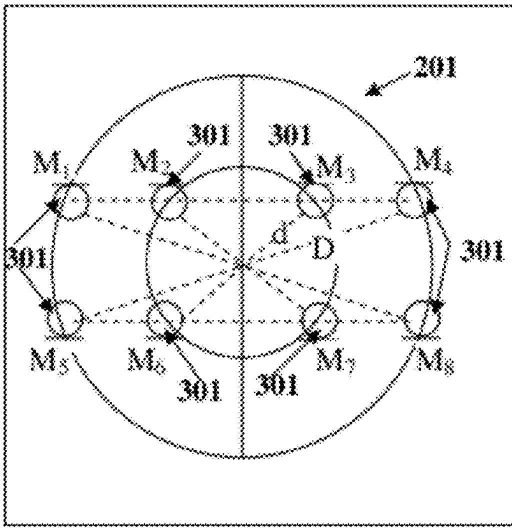
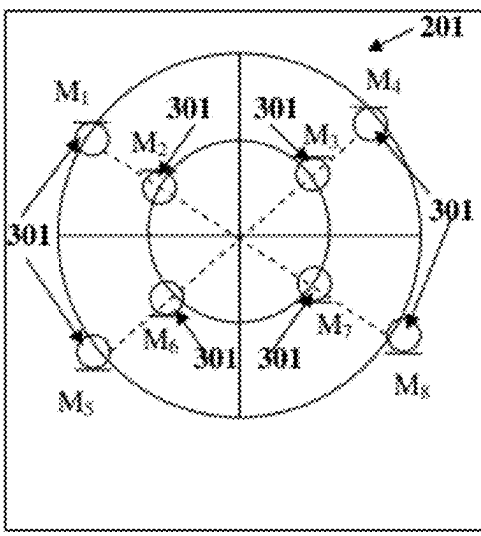


FIG. 18G



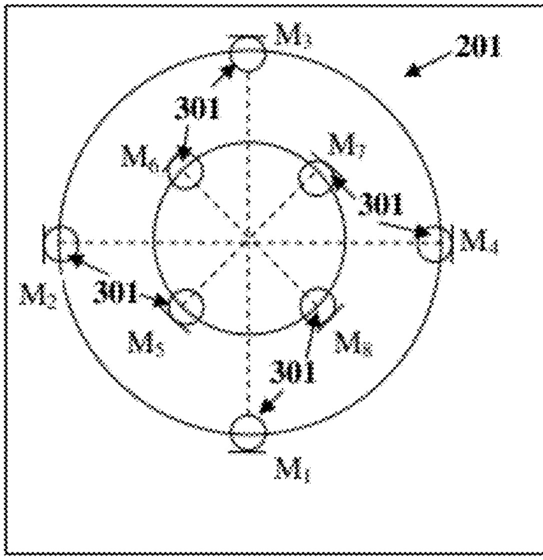
$\tau_1=D*\cos(\theta+\Phi_1)$
$\tau_2=d*\cos(\theta+\Phi_2)$
$\tau_3=d*\cos(\theta-\Phi_3)$
$\tau_4=D*\cos(\theta-\Phi_4)$
$\tau_5=-D*\cos(\theta-\Phi_5)$
$\tau_6=-d*\cos(\theta-\Phi_6)$
$\tau_7=-d*\cos(\theta+\Phi_7)$
$\tau_8=-D*\cos(\theta+\Phi_8)$

FIG. 19A



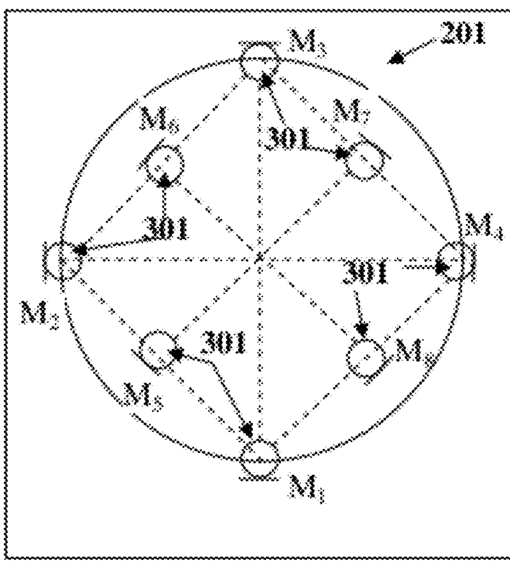
$\tau_1=D*\cos(\theta+\Phi_{12})$
$\tau_2=d*\cos(\theta+\Phi_{12})$
$\tau_3=d*\cos(\theta-\Phi_{34})$
$\tau_4=D*\cos(\theta-\Phi_{34})$
$\tau_5=-D*\cos(\theta-\Phi_{56})$
$\tau_6=-d*\cos(\theta-\Phi_{56})$
$\tau_7=-d*\cos(\theta+\Phi_{78})$
$\tau_8=-D*\cos(\theta+\Phi_{78})$

FIG. 19B



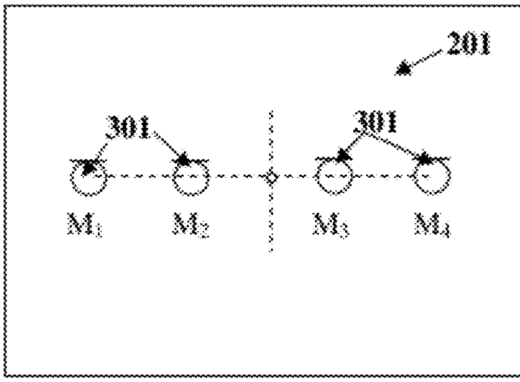
τ_1=-D*cos(θ)
τ_2=-D*sin(θ)
τ_3=D*cos(θ)
τ_4=D*sin(θ)
τ_5=-d*cos(θ-Φ_5)
τ_6=d*cos(θ+Φ_6)
τ_7=d*cos(θ-Φ_7)
τ_8=-d*cos(θ+Φ_8)

FIG. 19C



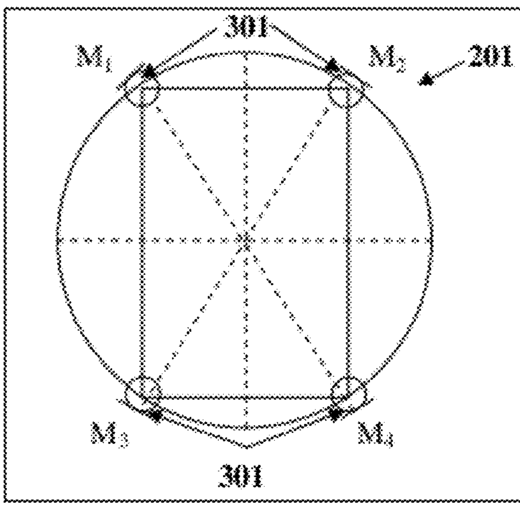
τ_1=-D*cos(θ)
τ_2=-D*sin(θ)
τ_3=D*cos(θ)
τ_4=D*sin(θ)
τ_5=-d*cos(θ-Φ_5)
τ_6=d*cos(θ+Φ_6)
τ_7=d*cos(θ-Φ_7)
τ_8=-d*cos(θ+Φ_8)

FIG. 19D



τ_1=-d1*sin(θ)
τ_2=-d2*sin(θ)
τ_3=d3*cos(θ)
τ_4=d4*cos(θ)

FIG. 19E



τ_1=d*cos(θ+Φ_1)
τ_2=d*cos(θ-Φ_2)
τ_3=-d*cos(θ-Φ_3)
τ_4=-d*cos(θ+Φ_4)

FIG. 19F

MICROPHONE ARRAY SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of provisional patent application No. 61/403,952 titled "Microphone array design and implementation for telecommunications and handheld devices", filed on Sep. 24, 2010 in the United States Patent and Trademark Office.

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

BACKGROUND

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

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

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

SUMMARY OF THE INVENTION

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

The method and system disclosed herein addresses the above stated need for enhancing acoustics of a target sound signal received from a target sound source, while suppressing ambient noise signals. As used herein, the term "target sound signal" refers to a sound signal from a desired or target sound

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

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

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

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

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

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

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

BRIEF DESCRIPTION OF THE DRAWINGS

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

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

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

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

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

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

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

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

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

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

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

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

FIG. **9**A exemplarily illustrates a graph showing the value of the steered response power-phase transform for every 10°.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

DETAILED DESCRIPTION OF THE INVENTION

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

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

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

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

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

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

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

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

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

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

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

FIG. **3** exemplarily illustrates a microphone array configuration showing a microphone array **201** having N sound sensors **301** arbitrarily distributed on a circle **302** with a diameter "d", where "N" refers to the number of sound sensors **301** in the microphone array **201**. Consider an example where N=4, that is, there are four sound sensors **301** M_0, M_1, M_2, and M_3 in the microphone array **201**. Each of the sound sensors **301** is positioned at an acute angle "Φ_n," from a Y-axis, where $\Phi_1 \geq 0$ and n=0, 1, 2, . . . N−1. In an example, the sound sensor **301** M_0 is positioned at an acute angle Φ_0 from the Y-axis; the sound sensor **301** M_1 is positioned at an acute angle Φ_1 from the Y-axis; the sound sensor **301** M_2 is positioned at an acute angle Φ_2 from the Y-axis; and the sound sensor **301** M_3 is positioned at an acute angle Φ_3 from the Y-axis. A filter-and-sum beamforming algorithm determines the output "y" of the microphone array **201** having N sound sensors **301** as disclosed in the detailed description of FIG. **4**.

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

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

The spatial directivity pattern H (ω, θ) for the target sound signal from angle θ with normalized frequency w is defined as:

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

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

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

For example, the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_0 is Φ_0, the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_1 is Φ_1, the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_2 is Φ_2, and the angle between the Y-axis and the line joining the origin and the sound sensor **301** M_3 is Φ_3. The distance between the origin O and the sound sensor **301** M_1, and the origin O and the sound sensor **301** M_3 when the incoming target sound signal from the target sound source is at an angle θ from the Y-axis is denoted as τ_1 and τ_3, respectively.

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

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

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

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

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

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

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

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

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

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

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

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

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

$$\begin{aligned} J(w) &= \int_{\Omega_p}\int_{\Theta_p}|H(\omega,\theta)-1|^2 d\omega d\theta + \\ &\quad \alpha\int_{\Omega_s}\int_{\Theta_s}|H(\omega,\theta)|^2 d\omega d\theta \\ &= \int_{\Omega_p}\int_{\Theta_p}|H(\omega,\theta)|^2 d\omega d\theta + \\ &\quad \alpha\int_{\Omega_S}\int_{\Theta_X}|H(\omega,\theta)|^2 d\omega d\theta - \\ &\quad 2\int_{\Omega_P}\int_{\Theta_P}\mathrm{Re}(H(\omega,\theta))d\omega d\theta + \\ &\quad \int_{\Omega_P}\int_{\Theta_P}1 d\omega d\theta \end{aligned} \quad (3)$$

Replacing

$|H(\omega,\theta)|^2=w^T g(\omega,\theta)g^H(\omega,\theta)w=w^T(G_R(\omega,\theta)+jG_1(\omega,\theta))w=w^T G_R(\omega,\theta)w$ and $Re(H(\omega,\theta))=w^T g_R(\omega,\theta)$, $J(\omega)$ becomes

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

$$Q=\int_{\Omega_p}\int_{73\ p}G_R(\omega,\theta)d\omega d\theta+\alpha\theta_{\Omega_s}\int_{\Theta_S}G_R(\omega,\theta)d\omega d\theta$$

$$\alpha=\int_{\Omega_p}\int_{\Theta_P}g_R(\omega,\theta)d\omega d\theta$$

$$d=\int_{\Omega_p}\int_{\Theta_P}1d\omega d\theta \quad (4)$$

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

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

$$w=Q^{-1}\alpha \quad (5)$$

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

$$\begin{bmatrix} g_R^T(\omega_{start},\theta_f) \\ \ldots \\ g_R^T(\omega_{end},\theta_f) \end{bmatrix} w = \begin{bmatrix} b_{start} \\ \ldots \\ b_{end} \end{bmatrix} \quad (6)$$

Now, the design problem becomes:

$$\min_w w^TQw-2w^Ta+d \text{ subject to } Cw=b \quad (7)$$

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

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

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

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

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

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

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

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

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

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

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

As exemplarily illustrated in FIG. **10**, the adaptive beamforming unit **203** disclosed herein comprises a fixed beamformer **204**, a blocking matrix **205**, an adaptation control unit **208**, and an adaptive filter **206**. The fixed beamformer **204** adaptively steers the directivity pattern of the microphone array **201** in the direction of the spatial location of the target sound signal from the target sound source for enhancing the target sound signal, when the target sound source is in motion. The sound sensors **301** in the microphone array **201** receive the sound signals $S_1, \ldots, S_4$, which comprise both the target sound signal from the target sound source and the ambient noise signals. The received sound signals are fed as input to the fixed beamformer **204** and the blocking matrix **205**. The fixed beamformer **204** outputs a signal "b". In an embodiment, the fixed beamformer **204** performs fixed beamforming by filtering and summing output sound signals from the sound sensors **301**. The blocking matrix **205** outputs a signal "z" which primarily comprises the ambient noise signals. The blocking matrix **205** blocks the target sound signal from the target sound source and feeds the ambient noise signals to the adaptive filter **206** to minimize the effect of the ambient noise signals on the enhanced target sound signal.

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

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

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

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

The adaptive filter matrix **206***b* adaptively filters the ambient noise signals in each of the frequency sub-bands in response to detecting the presence or absence of the target sound signal in the sound signals received from the disparate sound sources. The adaptive filter matrix **206***b* performs an adaptation step, where the adaptive filter **206** is adapted such that the filter output only contains the target sound signal, for example, speech. The synthesis filter bank **206***c* synthesizes a full-band sound signal using the frequency sub-bands of the enhanced target sound signal. The synthesis filter bank **206***c* performs a synthesis step where the sub-band sound signal is synthesized into a full-band sound signal. Since the noise reduction and the beamforming are performed in the same sub-band framework, the noise reduction as disclosed in the detailed description of FIG. **13**, by the noise reduction unit **207** is performed prior to the synthesis step, thereby reducing computation.

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

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

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

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

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

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

The audio codec **1402** receives the amplified output from the microphone amplifiers **1401**. The audio codec **1402** provides an adjustable gain level, for example, from about −74 dB to about 6 dB. The received sound signals are in an analog form. The audio codec **1402** converts the four channels of the sound signals in the analog form into digital sound signals.

The pre-amplifiers may not be required for some applications. The audio codec **1402** then transmits the digital sound signals to the DSP **1403** for processing of the digital sound signals. The DSP **1403** implements the sound source localization unit **202**, the adaptive beamforming unit **203**, and the noise reduction unit **207**.

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

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

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

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

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

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

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

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

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

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

Sampling frequency fs=16 k,

FIR filter taper length L=20

Passband (Θ_p, Ω_p)={300-5000 Hz, −5°-35°}, designed spatial directivity pattern is 1.

Stopband (Θ_s, Ω_s)={300~5000 Hz, −180°~−15°+45°~180°}, the designed spatial directivity pattern is 0.

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

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

Sampling frequency fs=16 k,

FIR filter taper length L=20

Passband (Θ_p, Ω_p)={300-5000 Hz, 40°-80°}, designed spatial directivity pattern is 1.

Stopband (Θ_s, Ω_s)={300~5000 Hz, −180°~30°+90°~180°}, the designed spatial directivity pattern is 0.

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

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

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

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

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

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

For the spatial region centered at 0°:

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

Stopband (Θ, Ω_s)={300~4000 Hz, −180°~−30°+30°~180°}, the designed spatial directivity pattern is 0.

For the spatial region centered at 90°:

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

Stopband (Θ_s, Ω_s)={300~4000 Hz, −180°~60°+120°~180°}, the designed spatial directivity pattern is 0. The directivity patterns for the spatial regions centered at −90° and 180° are similarly obtained.

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

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

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

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

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

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

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

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

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

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

We claim:

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

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

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

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 $\tau=f_s*t$, wherein f_s 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 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.

* * * * *

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
15/293,626	10/14/2016	Manli Zhu	CreativeTech_01RE_US

CONFIRMATION NO. 4199

FORMALITIES LETTER

OC000000087062658

64188
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

Date Mailed: 11/08/2016

NOTICE TO FILE MISSING PARTS OF REISSUE APPLICATION

Filing Date Granted

An application number and filing date have been accorded to this reissue application. The item(s) indicated below, however, are missing. Applicant is given **TWO MONTHS** from the date of this Notice within which to file all required items and pay any fees required below to avoid abandonment. Extensions of time may be obtained by filing a petition accompanied by the extension fee under the provisions of 37 CFR 1.136(a).

- The statutory basic filing fee is missing.
- The application search fee must be submitted.
- The application examination fee must be submitted.
- Additional claim fees of $ **2490** as a small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due.
- Surcharge as set forth in 37 CFR 1.16(f) must be submitted.
  The surcharge is due for any one of:
  - late submission of the basic filing fee, search fee, or examination fee,
  - late submission of inventor's oath or declaration,
  - filing an application that does not contain at least one claim on filing, or
  - submission of an application filed by reference to a previously filed application.

SUMMARY OF FEES DUE:

The fee(s) required within **TWO MONTHS** from the date of this Notice to avoid abandonment is/are itemized below. Small entity discount is in effect. If applicant is qualified for micro entity status, an acceptable Certification of Micro Entity Status must be submitted to establish micro entity status. (See 37 CFR 1.29 and forms PTO/SB/15A and 15B.)

- $ **140** basic filing fee.
- $ **70** surcharge.
- $ **300** search fee.
- $ **1080** examination fee.
- $ **1890** for **9** independent claims over the original patent.
- $ **600** for **15** total claims over the higher of 20, or the amount in the original patent.
- $(**0**) previous unapplied payment amount.
- $ **4080** TOTAL FEE BALANCE DUE.

Replies must be received in the USPTO within the set time period or must include a proper Certificate of Mailing or Transmission under 37 CFR 1.8 with a mailing or transmission date within the set time period. For more information and a suggested format, see Form PTO/SB/92 and MPEP 512.

Replies should be mailed to:

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Alexandria VA 22313-1450

Registered users of EFS-Web may alternatively submit their reply to this notice via EFS-Web, including a copy of this Notice and selecting the document description "Applicant response to Pre-Exam Formalities Notice". https://sportal.uspto.gov/authenticate/AuthenticateUserLocalEPF.html

For more information about EFS-Web please call the USPTO Electronic Business Center at 1-866-217-9197 or visit our website at http://www.uspto.gov/ebc.

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Questions about the contents of this notice and the requirements it sets forth should be directed to the Office of Data Management, Application Assistance Unit, at **(571) 272-4000** or **(571) 272-4200** or **1-888-786-0101**.

/ddfelix/

PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number
15/293,626

APPLICATION AS FILED - PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	140		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	300		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	1080		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	35 minus 20 =	* 15	x 40 =	600	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	12 minus 3 =	* 9	x 210 =	1890			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
			TOTAL	4010		TOTAL	

* If the difference in column 1 is less than zero, enter "0" in column 2.

APPLICATION AS AMENDED - PART II

AMENDMENT A

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE($)	ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	ADDITIONAL FEE($)
Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
Application Size Fee (37 CFR 1.16(s))									
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
					TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

AMENDMENT B

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE($)	ADDITIONAL FEE($)	OR	RATE($)	ADDITIONAL FEE($)
Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
Application Size Fee (37 CFR 1.16(s))									
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
					TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest found in the appropriate box in column 1.

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
15/293,626	10/14/2016	2654	0.00	CreativeTech_01RE_US	35	12

CONFIRMATION NO. 4199

FILING RECEIPT

OC000000087062857

64188
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

Date Mailed: 11/08/2016

Receipt is acknowledged of this reissue patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Inventor(s)

Manli Zhu, Pearl River, NY;
Qi Li, New Providence, NJ;

Applicant(s)

LI Creative Technologies, Inc., Florham Park, NJ, Assignee (with 37 CFR 1.172 Interest);

Assignment For Published Patent Application

LI Creative Technologies, Inc., Florham Park, NJ

Power of Attorney: None

Domestic Priority data as claimed by applicant

This application is a REI of 13/049,877 03/16/2011 PAT 8861756
which claims benefit of 61/403,952 09/24/2010

Foreign Applications for which priority is claimed (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.) - None.
Foreign application information must be provided in an Application Data Sheet in order to constitute a claim to foreign priority. See 37 CFR 1.55 and 1.76.

Permission to Access Application via Priority Document Exchange: Yes

Permission to Access Search Results: Yes

Applicant may provide or rescind an authorization for access using Form PTO/SB/39 or Form PTO/SB/69 as appropriate.

If Required, Foreign Filing License Granted: 11/07/2016

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 15/293,626**

Projected Publication Date: None, application is not eligible for pre-grant publication

Non-Publication Request: No

Early Publication Request: No
**** SMALL ENTITY ****
Title

Microphone Array System

Preliminary Class

381

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications: No

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Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

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Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

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Title 37, Code of Federal Regulations, 5.11 & 5.15

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ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

Courtesy Reminder for
Application Serial No: 15/293,626

Attorney Docket No: CreativeTech_01RE_US
Customer Number: 64188
Date of Electronic Notification: 11/08/2016

This is a courtesy reminder that new correspondence is available for this application. If you have not done so already, please review the correspondence. The official date of notification of the outgoing correspondence will be indicated on the form PTOL-90 accompanying the correspondence.

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To view your correspondence online or update your email addresses, please visit us anytime at **https://sportal.uspto.gov/secure/myportal/privatepair.** If you have any questions, please email the Electronic Business Center (EBC) at EBC@uspto.gov or call 1-866-217-9197.

Electronic Patent Application Fee Transmittal

Application Number:	15293626
Filing Date:	14-Oct-2016
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Filer:	Ashok Tankha
Attorney Docket Number:	CreativeTech_01RE_US

Filed as Small Entity

Filing Fees for **Utility under 35 USC 111(a)**

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
UTILITY REISSUE BASIC	2014	1	140	140
DESIGN AND UTILITY REISSUE BASIC	2114	1	300	300
DESIGN AND UTILITY REISSUE BASIC	2314	1	1080	1080
Pages:				
Claims:				
REISSUE CLAIMS IN EXCESS OF 20 FOR SMALL	2205	15	40	600
REISSUE- INDEPENDENT CLAIMS	2204	9	210	1890
Miscellaneous-Filing:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
LATE FILING FEE FOR OATH OR DECLARATION	2051	1	70	70

Petition:
Patent-Appeals-and-Interference:
Post-Allowance-and-Post-Issuance:
Extension-of-Time:
Miscellaneous:

Total in USD ($)	**4080**

Electronic Acknowledgement Receipt

EFS ID:	27537263
Application Number:	15293626
International Application Number:	
Confirmation Number:	4199
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Customer Number:	64188
Filer:	Ashok Tankha
Filer Authorized By:	
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1	Fee Worksheet (SB06)	fee-info.pdf	40270 f5783dff15b880d430478b387f37c4c4db4505eb	no	2

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New Applications Under 35 U.S.C. 111
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National Stage of an International Application under 35 U.S.C. 371
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New International Application Filed with the USPTO as a Receiving Office
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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
15/293,626	10/14/2016	2654	4080	CreativeTech_01RE_US	35	12

CONFIRMATION NO. 4199

UPDATED FILING RECEIPT

OC000000087478833

64188
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

Date Mailed: 11/29/2016

Receipt is acknowledged of this reissue patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Inventor(s)

Manli Zhu, Pearl River, NY;
Qi Li, New Providence, NJ;

Applicant(s)

LI Creative Technologies, Inc., Florham Park, NJ, Assignee (with 37 CFR 1.172 Interest);

Assignment For Published Patent Application

LI Creative Technologies, Inc., Florham Park, NJ

Power of Attorney: None

Domestic Priority data as claimed by applicant

This application is a REI of 13/049,877 03/16/2011 PAT 8861756
which claims benefit of 61/403,952 09/24/2010

Foreign Applications for which priority is claimed (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.) - None.
Foreign application information must be provided in an Application Data Sheet in order to constitute a claim to foreign priority. See 37 CFR 1.55 and 1.76.

Permission to Access Application via Priority Document Exchange: Yes

Permission to Access Search Results: Yes

Applicant may provide or rescind an authorization for access using Form PTO/SB/39 or Form PTO/SB/69 as appropriate.

If Required, Foreign Filing License Granted: 11/07/2016

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 15/293,626**

Projected Publication Date: None, application is not eligible for pre-grant publication

Non-Publication Request: No

Early Publication Request: No
**** SMALL ENTITY ****
Title

Microphone Array System

Preliminary Class

381

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications: No

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

Applicants also are advised that in the case of inventions made in the United States, the Director of the USPTO must issue a license before applicants can apply for a patent in a foreign country. The filing of a U.S. patent application serves as a request for a foreign filing license. The application's filing receipt contains further information and guidance as to the status of applicant's license for foreign filing.

Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

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Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

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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
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

EXAMINER	
ESCALANTE, OVIDIO	

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
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 _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 <u>does not identify the specific claim(s) and the specific claim language</u> 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]).

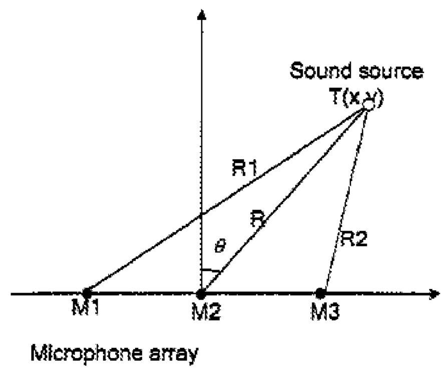

FIG. 2

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]).

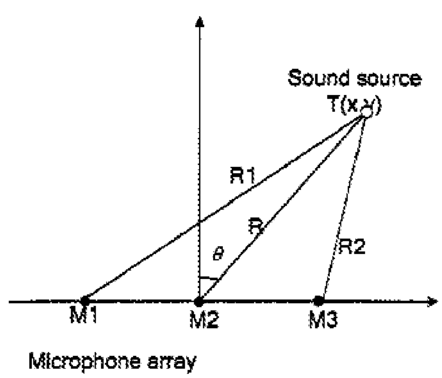

FIG. 2

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]).

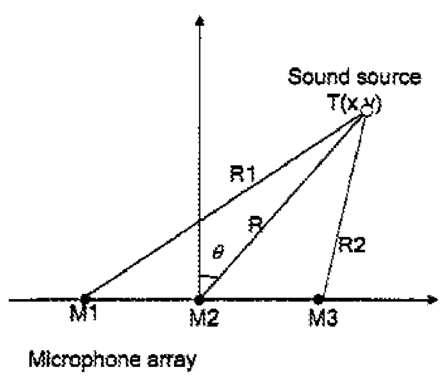

FIG. 2

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]).

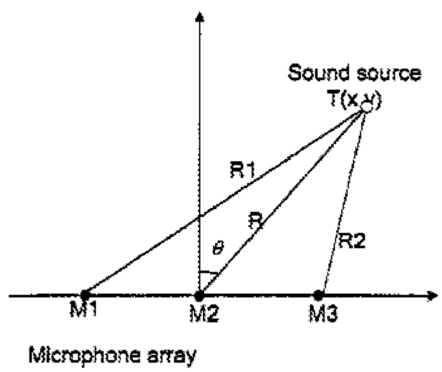

FIG. 2

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]).

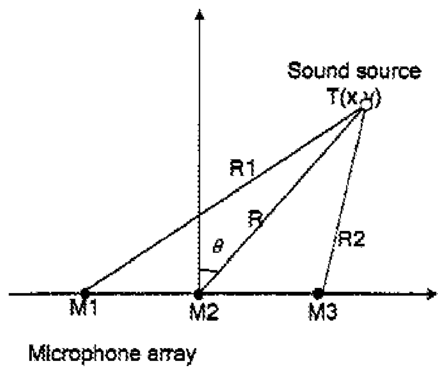

FIG 2

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.

<u>/Ovidio Escalante/</u>
Ovidio Escalante
Reexamination Specialist
Central Reexamination Unit - Art Unit 3992
(571) 272-7537

Conferees:

/Majid Banankhah/

/M. F./

Supervisory Patent Examiner, Art Unit 3992

Notice of References Cited	Application/Control No. 15/293,626	Applicant(s)/Patent Under Reexamination ZHU ET AL.	
	Examiner OVIDIO ESCALANTE	Art Unit 3992	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	CPC Classification	US Classification
*	A	US-2008/0232607 A1	09-2008	Tashev; Ivan	G01S3/86	381/71.11
*	B	US-2010/0241426 A1	09-2010	ZHANG; Chen	G10L21/0208	704/226
*	C	US-2011/0096915 A1	04-2011	Nemer; Elias	H04M3/568	379/158
*	D	US-2011/0135125 A1	06-2011	ZHAN; Wuzhou	H04R1/403	381/303
*	E	US-2011/0317522 A1	12-2011	Florencio; Dinei Afonso Ferreira	G01S3/8006	367/129
	F	US-				
	G	US-				
	H	US-				
	I	US-				
	J	US-				
	K	US-				
	L	US-				
	M	US-				

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	CPC Classification
	N	WO 2010020162 A1	02-2010	CHINA	WANG DONGQI	H04R1/403
	O					
	P					
	Q					
	R					
	S					
	T					

NON-PATENT DOCUMENTS

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

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

Index of Claims	**Application/Control No.** 15293626	**Applicant(s)/Patent Under Reexamination** ZHU ET AL.
	Examiner OVIDIO ESCALANTE	**Art Unit** 3992

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

☐ Claims renumbered in the same order as presented by applicant ☐ CPA ☐ T.D. ☐ R.1.47

CLAIM		DATE								
Final	Original	09/26/2017								
	1	✓								
	2	✓								
	3	✓								
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	33	✓								
	34	✓								
	35	✓								

EAST Search History

EAST Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	8	"8861756"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/26 06:37
L11	0	WO2010020162	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/26 06:50
L12	3	"2010020162"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/26 06:50
L13	1	2010-C11185.NRAN.	DERWENT	OR	OFF	2017/09/26 06:50
S1	3118	delay same sound adj source same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/08/24 10:42
S2	150	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/08/24 11:43
S3	8	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/08/24 11:44
S4	232	delay same sound adj source same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/08/24 14:22
S5	15	delay same sound adj source same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/08/25 08:52
S6	13272	delay same source same distance same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/15 07:54
S7	232	delay same source same distance same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/15 07:54
S8	24	delay same source same distance same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/15 07:55
S9	19	(@rlad<="20100924" or @ad<="20100924") and S8	US-PGPUB; USPAT;	OR	OFF	2017/09/15 09:00

			USOCR; DERWENT			
S10	2	(("20090141907") or ("20040161121")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2017/09/18 08:52
S11	198	delay same distance same (origin or source or reference adj point) and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:22
S12	115	(@rlad<="20100924" or @ad<="20100924") and S11	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:22
S13	38	S12 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:31
S14	69	delay same distance same (origin or source or reference adj point) same angle and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:49
S15	27	(@rlad<="20100924" or @ad<="20100924") and S14	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:49
S16	7	S15 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/18 10:49
S17	1	("8358766").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2017/09/18 13:08
S18	1	("8861698").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2017/09/18 14:58
S19	6	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/19 09:22
S20	5	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/19 10:05
S21	2	(("9648162") or ("20050231588")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2017/09/19 12:24
S22	3	(("9648162") or ("20050231588") or ("20130242030")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2017/09/19 12:25
S23	25	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/19 13:10
S24	3	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay with sample$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/19 13:10
S25	80	(@rlad<="20100924" or	US-PGPUB;	OR	OFF	2017/09/20

		@ad<="20100924") and (elevation with distance) same azimuth and microphone and delay	USPAT; USOCR; DERWENT			09:06
S26	320	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 09:06
S27	52	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 09:06
S28	16	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone with array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 09:09
S29	5	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and adaptive adj filter$4	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 12:40
S30	2	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 12:42
S31	1204	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 12:42
S32	5	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank same beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2017/09/20 12:42

EAST Search History (Interference)

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9/26/2017 6:51:43 AM
C:\Users\oescalante\Documents\EAST\Workspaces\15293626.wsp

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SEARCH REPORT

STIC Database Tracking Number: 553414

To: Ovidio Escalante
Location: MDE-4C01
Art Unit: N/A
Wednesday, September 27, 2017

Case Serial Number: 15/293626

From: Kimberly Johnson
Location: EIC2600
KNX-8B59B
Phone: (571)272-3493

kimberly.johnson@uspto.gov

Search Notes

1 of 1 DOCUMENT

UNITED STATES PATENT AND TRADEMARK OFFICE GRANTED PATENT

8861756

Link to Claims Section

October 14, 2014

Microphone array system

REISSUE:
October 14, 2016 - Reissue Application filed, Ex. Gp.: 2654; Re. S.N.: 15/293,626 , (O.G. December 27, 2016)

INVENTOR: Zhu, Manli - Pearl River, New York, United States of America (US) ; Li, Qi - New Providence, New Jersey, United States of America (US)

APPL-NO: 049877 (13)

FILED-DATE: March 16, 2011

GRANTED-DATE: October 14, 2014

PRIORITY: March 16, 2011 - 13049877, United States of America (US)

ASSIGNEE-PRE-ISSUE:
March 21, 2011 - ASSIGNMENT OF ASSIGNORS INTEREST (SEE DOCUMENT FOR DETAILS)., LI CREATIVE TECHNOLOGIES, INC., 25B HANOVER ROAD, SUITE 140, FLORHAM PARK, NEW JERSEY, UNITED STATES OF AMERICA (US), 07932, Reel and Frame Number: 026003/0985

ASSIGNEE-AT-ISSUE:
LI Creative Technologies, Inc., Florham Park, New Jersey, United States of America (US), United States company or corporation (02)

LEGAL-STATUS:

March 21, 2011 - ASSIGNMENT

CORE TERMS: array, microphone, target, sensor, noise, adaptive, beamforming, filter, spatial, exemplarily, reduction, omega, directivity, ambient, region, frequency, minus, sub-band, tau, bank, centered, computer, localization, algorithm, matrix, filtering, beamformer, simulation, enhancing, circular

ENGLISH-ABST:

A method and system for enhancing a target sound signal from multiple sound signals is provided. An array of an

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

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1. US Patent Issued to Li Creative Technologies on Oct. 14 for "Microphone array system" (New Jersey, New York Inventors)

US Fed News, 294 words

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... Oct. 14 -- United States Patent no. 8,861,756 , issued on Oct. 14, was assigned to Li Creative Technologies ...

Date
Oct 14, 2014

2. US Patent granted to Li Creative Technologies, Inc (NJ) on October 14 titled as "Microphone array system"

US Official News, 255 words

... States Patent and Trademark Office has granted patent no. 8,861,756 , on October 14, 2014, to Li Creative Technologies, Inc (NJ), ...

Date
Oct 20, 2014

3. Li Creative Technologies Granted United States Patent for Microphone Array System

Global IP News. Electronics Patent News, 336 words

... Application Number: 13/049,877 Patent Publication Number: 8,861,756 International Patent Classification Codes: G01S 3/8055 (20130101), H04R ...
... 14 -- Li Creative Technologies has been granted a patent (8,861,756) for microphone array system. This invention was developed by Zhu ...

Date
Oct 14, 2014

4. Li Creative Technologies Granted United States Patent for Microphone Array System

Global IP News. Electronics Patent News, 334 words

... Electronics Patent News Patent Application Number: 13/049,877 Patent Publication Number: 8,861,756 International Patent Classification Codes: G01S 3/8055 (20130101), H04R 3/005 (20130101), ...
... 14 -- Li Creative Technologies has been granted a patent (8,861,756) for microphone array system. This invention was developed by Zhu ...
... Electronics Patent News Patent Application Number: 13/049,877 Patent Publication Number: 8,861,756 International Patent Classification Codes: G01S 3/8055 (20130101), H04R 3/005 (20130101), ...
... 14 -- Li Creative Technologies has been granted a patent (8,861,756) for microphone array system. This invention was developed by Zhu ...

Date
Oct 14, 2014

5. HANLEY: A 24-hour darts marathon [...];In brief

The Sentinel (Stoke), NEWS; Pg. 7, 59 words

... who is interested should telephone organiser Paul Barker on 0784 8861756...

Date
Oct 01, 2011

6. Radio Kombi Baden-Württemberg

GlobalAdSource (German), 41 words

... USD Media Type Print Country Germany Source Werben & Verkaufen (W&V) ...

Date
Jul 28, 2005

1

User Name: KIMBERLY JOHNSON
Date and Time: Wednesday, September 27, 2017 11:56:00 AM EDT
Job Number: 54041809

Document (1)

1. US Patent Issued to LI Creative Technologies on Oct. 14 for "Microphone array system" (New Jersey, New York Inventors)

    Client/Matter: -None-

    Search Terms: 8861756 or 8,861,756

    Search Type: Terms and Connectors

    Narrowed by:

Content Type	Narrowed by
News	-None-

US Patent Issued to LI Creative Technologies on Oct. 14 for "Microphone array system" (New Jersey, New York Inventors)

US Fed News

October 14, 2014 Tuesday 11:27 PM EST

Length: 294 words

Dateline: ALEXANDRIA, Va.

Body

ALEXANDRIA, Va., Oct. 14 -- United States Patent no. ***8,861,756***, issued on Oct. 14, was assigned to LI Creative Technologies Inc. (Florham Park, N.J.).

"Microphone array system" was invented by Manli Zhu (Pearl River, N.Y.) and Qi Li (New Providence, N.J.). According to the abstract* released by the U.S. Patent & Trademark Office: "A method and system for enhancing a target sound signal from multiple sound signals is provided. An array of an arbitrary number of sound sensors positioned in an arbitrary configuration receives the sound signals from multiple disparate sources. The sound signals comprise the target sound signal from a target sound source, and ambient noise signals. A sound source localization unit, an adaptive beamforming unit, and a noise reduction unit are in operative communication with the array of sound sensors. The sound source localization unit estimates a spatial location of the target sound signal from the received sound signals. The adaptive beamforming unit performs adaptive beamforming by steering a directivity pattern of the array of sound sensors in a direction of the spatial location of the target sound signal, thereby enhancing the target sound signal and partially suppressing the ambient noise signals, which are further suppressed by the noise reduction unit." The patent was filed on March 16, 2011, under Application No. 13/049,877. *For further information, including images, charts and tables, please visit: http://patft.uspto.gov/netacgi/nph-Parser?Sect1=PTO2&Sect2=HITOFF&p=1&u=%2Fnetahtml%2FPTO%2Fsearch-bool.html&r=1&f=G&l=50&co1=AND&d=PTXT&s1=8861756&OS=8861756&RS=8861756 For any query with respect to this article or any other content requirement, please contact Editor at htsyndication@hindustantimes.com

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Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	15293626	ZHU ET AL.
	Examiner	**Art Unit**
	OVIDIO ESCALANTE	3992

CPC- SEARCHED

Symbol	Date	Examiner
G01S: 3/8055; 5/22; 3/801	9/26/17	OE
H04R: 3/005; 1/406; 2201/403; 2201/401	9/26/17	OE
H04M: 3/568	9/26/17	OE

CPC COMBINATION SETS - SEARCHED

Symbol	Date	Examiner

US CLASSIFICATION SEARCHED

Class	Subclass	Date	Examiner
381	300. 57	9/26/17	OE
381	92, 94.1, 93	9/26/17	OE

* See search history printout included with this form or the SEARCH NOTES box below to determine the scope of the search.

SEARCH NOTES

Search Notes	Date	Examiner
Requested Litigation Search	9/26/17	OE
EAST Search	9/26/17	OE
Reviewed Patent Prosecution History	9/26/17	OE

INTERFERENCE SEARCH

US Class/ CPC Symbol	US Subclass / CPC Group	Date	Examiner

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~~ 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 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~~ 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 (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 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~~ <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 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 <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 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

/a tankha/
Ashok Tankha
Attorney for Applicant
Reg. No: 33,802

Correspondence Address
Lipton Weinberger &Husick

36 Greenleigh Drive
Sewell, NJ 08080
Fax: 856-374-0246
Phone: 856-266-5145
Email: ash@ipprocure.com

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REISSUE APPLICATION DECLARATION BY THE ASSIGNEE	Docket Number (optional) CreativeTech_01RE_US

I hereby declare that:

The residence and mailing address of the inventor or joint inventors are stated below.

I am authorized to act on behalf of the following assignee: Li Creative Technologies, Inc.

The entire title to the patent identified below is vested in said assignee.

Inventor Li, Qi(Peter)

Residence: City	State	Country
New Providence	NJ	USA

Mailing Address
225 Runnymede Parkway

City	State	Zip	Country
New Providence	NJ	07974	USA

☑ Additional Inventors are named on separately numbered sheets attached hereto.

Patent Number US8861756 | Date of Patent Issued 14 October, 2014

I believe said inventor(s) to be the original inventor or original joint inventors of the subject matter which is described and claimed in said patent, for which a reissue patent is sought on the invention titled:

Microphone array system

the specification of which

☐ is attached hereto.

☑ was filed on 10/14/2016 as reissue application number 15/293,626 .

The above-identified application was made or authorized to be made by me.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

I believe the original patent to be wholly or partly inoperative or invalid, for the reasons described below.
(Check all boxes that apply.)

☐ by reason of a defective specification or drawing.

☑ by reason of the patentee claiming more or less than he had the right to claim in the patent.

☑ by reason of other errors.

[Page 1 of 2]

REISSUE APPLICATION DECLARATION BY THE ASSIGNEE	Docket Number (Optional)

At least one error upon which reissue is based is described below. If the reissue is a broadening reissue, a claim that the application seeks to broaden must be identified and the box below must be checked:

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.

[Attach additional sheets, if needed.]

☑ **The application for the original patent was filed under 37 CFR 1.46 by the assignee of the entire interest.**

I hereby appoint:

☐ Practitioners associated with Customer Number:

OR

☑ Practitioner(s) named below:

Name	Registration Number
Ashok Tankha	33802

as my/our attorney(s) or agent(s) to prosecute the application identified above, and to transact all business in the United States Patent and Trademark Office connected therewith.

Correspondence Address: Direct all communications about the application to:

☐ The address associated with Customer Number:

OR

☑ Firm or Individual Name	Ashok Tankha				
Address	36 Greenleigh drive				
City	Sewell	State	NJ	Zip	08080
Country	US				
Telephone	856-266-5145	Email	ash@ipprocurement.com		

WARNING:

Petitioner/applicant is cautioned to avoid submitting personal information in documents filed in a patent application that may contribute to identity theft. Personal information such as social security numbers, bank account numbers, or credit card numbers (other than a check or credit card authorization form PTO-2038 submitted for payment purposes) is never required by the USPTO to support a petition or an application. If this type of personal information is included in documents submitted to the USPTO, petitioners/applicants should consider redacting such personal information from the documents before submitting them to the USPTO. Petitioner/applicant is advised that the record of a patent application is available to the public after publication of the application (unless a non-publication request in compliance with 37 CFR 1.213(a) is made in the application) or issuance of a patent. Furthermore, the record from an abandoned application may also be available to the public if the application is referenced in a published application or an issued patent (see 37 CFR 1.14). Checks and credit card authorization forms PTO-2038 submitted for payment purposes are not retained in the application file and therefore are not publicly available.

Signature	Date (Optional) January 29, 2018
Full name of person signing (given name, family name) Qi Li	
Address of Assignee New Providence, NJ, USA	

[Page 2 of 2]

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.
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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.

PTO/AIA/22 (03-13)
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PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a)

Docket Number (Optional): CreativeTech_01RE_US

Application Number 15/293,626	Filed 10/14/2016
For Microphone Array System	
Art Unit 3992	Examiner Escalante, Ovidio

This is a request under the provisions of 37 CFR 1.136(a) to extend the period for filing a reply in the above-identified application.

The requested extension and fee are as follows (check time period desired and enter the appropriate fee below):

	Fee	Small Entity Fee	Micro Entity Fee	
☑ One month (37 CFR 1.17(a)(1))	$200	$100	$50	$ 100
☐ Two months (37 CFR 1.17(a)(2))	$600	$300	$150	$
☐ Three months (37 CFR 1.17(a)(3))	$1,400	$700	$350	$
☐ Four months (37 CFR 1.17(a)(4))	$2,200	$1,100	$550	$
☐ Five months (37 CFR 1.17(a)(5))	$3,000	$1,500	$750	$

☑ Applicant asserts small entity status. See 37 CFR 1.27.

☐ Applicant certifies micro entity status. See 37 CFR 1.29.
Form PTO/SB/15A or B or equivalent must either be enclosed or have been submitted previously.

☐ A check in the amount of the fee is enclosed.

☐ Payment by credit card. Form PTO-2038 is attached.

☐ The Director has already been authorized to charge fees in this application to a Deposit Account.

☑ The Director is hereby authorized to charge any fees which may be required, or credit any overpayment, to Deposit Account Number 600314.

☐ Payment made via EFS-Web.

WARNING: Information on this form may become public. Credit card information should not be included on this form. Provide credit card information and authorization on PTO-2038.

I am the

☐ applicant.

☑ attorney or agent of record. Registration number 33802.

☐ attorney or agent acting under 37 CFR 1.34. Registration number ____________.

/a tankha/	01/29/2018
Signature	Date
Ashok Tankha	856-266-5145
Typed or printed name	Telephone Number

NOTE: This form must be signed in accordance with 37 CFR 1.33. See 37 CFR 1.4 for signature requirements and certifications. Submit multiple forms if more than one signature is required, see below*.

☐ * Total of ____________ forms are submitted.

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.

Electronic Patent Application Fee Transmittal

Application Number:	15293626
Filing Date:	14-Oct-2016
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Filer:	Ashok Tankha
Attorney Docket Number:	CreativeTech_01RE_US

Filed as Small Entity

Filing Fees for **Utility under 35 USC 111(a)**

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Extension - 1 month with $0 paid	2251	1	100	100
Miscellaneous:				
		Total in USD ($)		**100**

Electronic Acknowledgement Receipt

EFS ID:	31639725
Application Number:	15293626
International Application Number:	
Confirmation Number:	4199
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Customer Number:	64188
Filer:	Ashok Tankha
Filer Authorized By:	
Attorney Docket Number:	CreativeTech_01RE_US
Receipt Date:	29-JAN-2018
Filing Date:	14-OCT-2016
Time Stamp:	22:54:05
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$100
RAM confirmation Number	013018INTEFSW22572500
Deposit Account	
Authorized User	

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	CreativeTech_01RE_US_Transmittal_Letter.pdf	22374 afbb9c7090778a807f738c3f6ed7c1c926c447c3	no	2
Warnings:					
Information:					
2	Amendment/Req. Reconsideration-After Non-Final Reject	CreativeTech_01RE_US_Response.pdf	170776 da390c960c765b3df0bddb27aa5fc7017d13d06e	no	35
Warnings:					
Information:					
3	Oath or Declaration filed	CreativeTech_01RE_US_Declaration_aia0006.pdf	1272180 3b80d3ffb7ad53b27a587d7489f8a24629b8ffaa	no	3
Warnings:					
Information:					
4	Extension of Time	CreativeTech_01RE_US_time_extension_aia0022.pdf	843789 7e9e9a277053c395bce1bfdc2dc5a92c7e71aad5	no	2
Warnings:					
Information:					
5	Fee Worksheet (SB06)	fee-info.pdf	30023 e3f4ed32354db6065cd6917ec25a1d607d1049e2	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	2339142		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

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

TRANSMITTAL LETTER

1. **Attachments:** We are transmitting herewith the attached papers for the above identified patent application:

§ **Response to office action (35 pages)**;
§ **Petition for 1 month time extension, Form PTO/AIA/22;** and
§ **Revised reissue application declaration by the Assignee (3 pages)**.

2. **Fees:**

a. **$100** for one month time extension.

3. **General Authorization to charge or credit fees:** The Director is hereby authorized to charge any underpayment of fee or any other fee that may be required to deposit account # **600314**.

4. **Certificate Of Transmission Under 37 § CFR 1.8**: The undersigned hereby certifies that this Transmittal Letter and the papers as described in paragraph 1 hereinabove, are being electronically transmitted to the United States Patent and Trademark Office **via the USPTO electronic filing system** on this **29th** day of **January 2018.**

Respectfully submitted,

Date: Jan. 29, 2018

/a tankha/
Ashok Tankha
Attorney For Applicant
Reg. No. 33,802

Correspondence Address
Lipton, Weinberger & Husick
36 Greenleigh Drive
Sewell, NJ 08080
Phone: 856-266-5145
Fax: 856-374-0246

Email: ash@ipprocure.com

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875	Application or Docket Number 15/293,626	Filing Date 10/14/2016	☐ To be Mailed

ENTITY: ☐ LARGE ☒ SMALL ☐ MICRO

APPLICATION AS FILED – PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	RATE ($)	FEE ($)
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	
☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	minus 20 =	*	X $ =	
INDEPENDENT CLAIMS (37 CFR 1.16(h))	minus 3 =	*	X $ =	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	

APPLICATION AS AMENDED – PART II

AMENDMENT	01/29/2018	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	* 35	Minus	** 35	= 0	x $50 =	0
	Independent (37 CFR 1.16(h))	* 12	Minus	***12	= 0	x $230 =	0
	☐ Application Size Fee (37 CFR 1.16(s))						
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
						TOTAL ADD'L FEE	**0**

AMENDMENT		CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	*	Minus	**	=	X $ =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X $ =	
	☐ Application Size Fee (37 CFR 1.16(s))						
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
						TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

LIE
VIOLA ROGERS

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

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
ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

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.

/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

EAST Search History

EAST Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	3298	delay same sound adj source same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L2	174	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L3	10	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L4	252	delay same sound adj source same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L5	16	delay same sound adj source same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L6	13605	delay same source same distance same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L7	245	delay same source same distance same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L8	26	delay same source same distance same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L9	19	(@rlad<="20100924" or @ad<="20100924") and L8	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L10	2	(("20090141907") or ("20040161121")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L11	209	delay same distance same (origin or source or reference adj point) and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L12	115	(@rlad<="20100924" or @ad<="20100924") and L11	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L13	38	L12 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L14	73	delay same distance same (origin or source or reference adj point) same angle and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L15	27	(@rlad<="20100924" or @ad<="20100924") and L14	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32

L16	7	L15 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L17	1	("8358766").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L18	1	("8861698").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L19	6	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L20	5	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L21	2	(("9648162") or ("20050231588")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L22	3	(("9648162") or ("20050231588") or ("20130242030")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L23	25	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L24	3	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay with sample$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L25	81	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L26	326	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L27	54	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L28	18	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone with array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L29	7	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and adaptive adj filter$4	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L30	2	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L31	1235	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L32	5	(@rlad<="20100924" or	US-PGPUB;	OR	OFF	2018/02/16

		@ad<="20100924") and analysis adj filter adj bank same beamformer	USPAT; USOCR; DERWENT			11:32
L33	9	"8861756"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L34	0	WO2010020162	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L35	3	"2010020162"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L36	1	2010-C11185.NRAN.	DERWENT	OR	OFF	2018/02/16 11:32
L37	58	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L38	9	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size and motion	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L39	10	"8554970"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L40	1	("7590075").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L41	1	("8861756").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L42	216	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L43	151	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L44	2	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L45	17	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L46	6	(digital adj signal adj processor or DSP) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/02/16 11:32
L47	5	(processor) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/02/16 11:32

EAST Search History (Interference)

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2/16/2018 11:41:17 AM
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Index of Claims	Application/Control No. 15293626	Applicant(s)/Patent Under Reexamination ZHU ET AL.
	Examiner OVIDIO ESCALANTE	Art Unit 3992

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

☐ Claims renumbered in the same order as presented by applicant ☐ CPA ☐ T.D. ☐ R.1.47

CLAIM		DATE									
Final	Original	09/26/2017	02/16/2018								
	1	✓	✓								
	2	✓	✓								
	3	✓	✓								
	4	✓	✓								
	5	✓	✓								
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	35	✓	✓								

Search Notes	**Application/Control No.** 15293626	**Applicant(s)/Patent Under Reexamination** ZHU ET AL.
	Examiner OVIDIO ESCALANTE	**Art Unit** 3992

CPC- SEARCHED

Symbol	Date	Examiner
G01S: 3/8055; 5/22; 3/801	9/26/17	OE
H04R: 3/005; 1/406; 2201/403; 2201/401	9/26/17	OE
H04M: 3/568	9/26/17	OE
Update	2/16/2018	OE

CPC COMBINATION SETS - SEARCHED

Symbol	Date	Examiner

US CLASSIFICATION SEARCHED

Class	Subclass	Date	Examiner
381	300. 57	9/26/17	OE
381	92, 94.1, 93	9/26/17	OE
Update	Update	2/16/18	OE

* See search history printout included with this form or the SEARCH NOTES box below to determine the scope of the search.

SEARCH NOTES

Search Notes	Date	Examiner
Requested Litigation Search	9/26/17	OE
EAST Search	9/26/17	OE
Reviewed Patent Prosecution History	9/26/17	OE
Updated EAST Search	2/16/18	OE

INTERFERENCE SEARCH

US Class/ CPC Symbol	US Subclass / CPC Group	Date	Examiner

EAST Search History

EAST Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	3298	delay same sound adj source same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L2	174	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L3	10	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L4	252	delay same sound adj source same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L5	16	delay same sound adj source same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L6	13605	delay same source same distance same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L7	245	delay same source same distance same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L8	26	delay same source same distance same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L9	19	(@rlad<="20100924" or @ad<="20100924") and L8	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L10	2	(("20090141907") or ("20040161121")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L11	209	delay same distance same (origin or source or reference adj point) and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L12	115	(@rlad<="20100924" or @ad<="20100924") and L11	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L13	38	L12 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L14	73	delay same distance same (origin or source or reference adj point) same angle and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L15	27	(@rlad<="20100924" or @ad<="20100924") and L14	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32

L16	7	L15 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L17	1	("8358766").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L18	1	("8861698").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L19	6	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L20	5	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L21	2	(("9648162") or ("20050231588")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L22	3	(("9648162") or ("20050231588") or ("20130242030")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L23	25	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L24	3	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay with sample$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L25	81	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L26	326	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L27	54	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L28	18	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone with array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L29	7	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and adaptive adj filter$4	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L30	2	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L31	1235	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L32	5	(@rlad<="20100924" or	US-PGPUB;	OR	OFF	2018/02/16

		@ad<="20100924") and analysis adj filter adj bank same beamformer	USPAT; USOCR; DERWENT			11:32
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L34	0	WO2010020162	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L35	3	"2010020162"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L36	1	2010-C11185.NRAN.	DERWENT	OR	OFF	2018/02/16 11:32
L37	58	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L38	9	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size and motion	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L39	10	"8554970"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L40	1	("7590075").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L41	1	("8861756").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/02/16 11:32
L42	216	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L43	151	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
L44	2	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/02/16 11:32
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L46	6	(digital adj signal adj processor or DSP) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/02/16 11:32
L47	5	(processor) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/02/16 11:32

EAST Search History (Interference)

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2/16/2018 11:42:33 AM
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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 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] <u>a linear, circular, or other</u> configuration, a sound source localization unit, an adaptive beamforming unit, and a noise reduction unit, <u>wherein said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are integrated in a digital signal processor, and</u> 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 <u>array of</u> sound sensors [and] <u>in</u> 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 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 (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 (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 (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

/a tankha/
Ashok Tankha
Attorney for Applicant
Reg. No: 33,802

Correspondence Address
Lipton Weinberger &Husick
36 Greenleigh Drive
Sewell, NJ 08080
Fax: 856-374-0246
Phone: 856-266-5145
Email: ash@ipprocurement.com

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REISSUE APPLICATION DECLARATION BY THE INVENTOR	Docket Number (Optional) CreativeTech_01RE_US

I hereby declare that:
Each inventor's residence and mailing address are stated below next to their name.
I believe I am the original inventor or an original joint inventor of the subject matter which is described and claimed in patent number US8861756, granted 14 October, 2014 and for which a reissue patent is sought on the invention titled Microphone Array System,

the specification of which

☐ is attached hereto

☑ was filed on 10/14/2016 as reissue application number 15/293,626 .

The above-identified application was made or authorized to be made by me.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

I believe the original patent to be wholly or partly inoperative or invalid, for the reasons described below. (Check all boxes that apply.)

☐ by reason of a defective specification or drawing.

☐ by reason of the patentee claiming more or less than he had the right to claim in the patent.

☑ by reason of other errors.

At least one error upon which reissue is based is described below. If the reissue is a broadening reissue, a claim that the application seeks to broaden must be identified:

Independent claims 1, 9, 20-22, 26, and 30-35 have been amended for more clarity (see attached sheets)

(REISSUE APPLICATION DECLARATION BY THE INVENTOR, page 2)

Docket Number (Optional)
CreativeTech_01RE_US

Note: To appoint a power of attorney, use form PTO/AIA/81.

Correspondence Address: Direct all communications about the application to:

☐ The address associated with Customer Number:

OR

☑ Firm or Individual Name	Ashok Tankha				
Address	36 Greenleigh Drive				
City	Sewell,	State	NJ	Zip	08080
Country	USA				
Telephone	856-266-5145	Email	ash@ipprocurement.com		

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Petitioner/applicant is cautioned to avoid submitting personal information in documents filed in a patent application that may contribute to identity theft. Personal information such as social security numbers, bank account numbers, or credit card numbers (other than a check or credit card authorization form PTO-2038 submitted for payment purposes) is never required by the USPTO to support a petition or an application. If this type of personal information is included in documents submitted to the USPTO, petitioners/applicants should consider redacting such personal information from the documents before submitting them to the USPTO. Petitioner/applicant is advised that the record of a patent application is available to the public after publication of the application (unless a non-publication request in compliance with 37 CFR 1.213(a) is made in the application) or issuance of a patent. Furthermore, the record from an abandoned application may also be available to the public if the application is referenced in a published application or an issued patent (see 37 CFR 1.14). Checks and credit card authorization forms PTO-2038 submitted for payment purposes are not retained in the application file and therefore are not publicly available.

Legal name of sole or first inventor (*E.g.*, Given Name (first and middle (if any) and Family Name or Surname)

Manli Zhu

Inventor's Signature: Manli Zhu		Date (Optional): 4/13/2018	
Residence: City: New City	State: NY	Country: USA	
Mailing Address: 49 Woodside Dr			
City: New City	State: NY	Zip: 10956	Country: USA

[■] Additional joint inventors are named on the One supplemental sheet(s) PTO/AIA/10 attached hereto.

[Page 2 of 2]

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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.

Attached Sheet

The limitation: "*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*" in claim 1 has been amended by replacing the terms "*an arbitrary*" to "*a linear, circular, or other*" to add more clarity. The term "arbitrary" is vague. It is replaced by "*a linear, circular, or other*", which better describes the configuration of the microphone array system. Furthermore, in the same limitation, to further add more clarity to the structure of the microphone array system, the terms "*said sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are integrated in a digital signal processor*" have been added to show the integration of sound source localization unit, adaptive beamforming unit and noise reduction unit within a digital signal processor.

Similar amendment has been made in the other independent claims 9, 20-22, 26 and 30-35.

Doc Code: A.NE.AFCP
Document Description: After Final Consideration Pilot Program Request

PTO/SB/434 (05-13)

CERTIFICATION AND REQUEST FOR CONSIDERATION UNDER THE AFTER FINAL CONSIDERATION PILOT PROGRAM 2.0

Practitioner Docket No.:	Application No.:	Filing Date:
CreativeTech_01RE_US	15/293,626	10/14/2016
First Named Inventor:	Title:	
Manli Zhu	Microphone Array System	

APPLICANT HEREBY CERTIFIES THE FOLLOWING AND REQUESTS CONSIDERATION UNDER THE AFTER FINAL CONSIDERATION PILOT PROGRAM 2.0 (AFCP 2.0) OF THE ACCOMPANYING RESPONSE UNDER 37 CFR 1.116.

1. The above-identified application is (i) an original utility, plant, or design nonprovisional application filed under 35 U.S.C. 111(a) [a continuing application (*e.g.*, a continuation or divisional application) is filed under 35 U.S.C. 111(a) and is eligible under (i)], or (ii) an international application that has entered the national stage in compliance with 35 U.S.C. 371(c).
2. The above-identified application contains an outstanding final rejection.
3. Submitted herewith is a response under 37 CFR 1.116 to the outstanding final rejection. The response includes an amendment to at least one independent claim, and the amendment does not broaden the scope of the independent claim in any aspect.
4. This certification and request for consideration under AFCP 2.0 is the only AFCP 2.0 certification and request filed in response to the outstanding final rejection.
5. Applicant is willing and available to participate in any interview requested by the examiner concerning the present response.
6. This certification and request is being filed electronically using the Office's electronic filing system (EFS-Web).
7. Any fees that would be necessary consistent with current practice concerning responses after final rejection under 37 CFR 1.116, *e.g.*, extension of time fees, are being concurrently filed herewith. [There is no additional fee required to request consideration under AFCP 2.0.]
8. By filing this certification and request, applicant acknowledges the following:

- Reissue applications and reexamination proceedings are not eligible to participate in AFCP 2.0.
- The examiner will verify that the AFCP 2.0 submission is compliant, *i.e.*, that the requirements of the program have been met (see items 1 to 7 above). For compliant submissions:
    - The examiner will review the response under 37 CFR 1.116 to determine if additional search and/or consideration (i) is necessitated by the amendment and (ii) could be completed within the time allotted under AFCP 2.0. If additional search and/or consideration is required but cannot be completed within the allotted time, the examiner will process the submission consistent with current practice concerning responses after final rejection under 37 CFR 1.116, *e.g.*, by mailing an advisory action.
    - If the examiner determines that the amendment does not necessitate additional search and/or consideration, or if the examiner determines that additional search and/or consideration is required and could be completed within the allotted time, then the examiner will consider whether the amendment places the application in condition for allowance (after completing the additional search and/or consideration, if required). If the examiner determines that the amendment does not place the application in condition for allowance, then the examiner will contact the applicant and request an interview.
        - The interview will be conducted by the examiner, and if the examiner does not have negotiation authority, a primary examiner and/or supervisory patent examiner will also participate.
        - If the applicant declines the interview, or if the interview cannot be scheduled within ten (10) calendar days from the date that the examiner first contacts the applicant, then the examiner will proceed consistent with current practice concerning responses after final rejection under 37 CFR 1.116.

Signature	Date
/a tankha/	04/18/2018
Name (Print/Typed) Ashok Tankha	Practitioner Registration No. 33802

***Note**: This form must be signed in accordance with 37 CFR 1.33. See 37 CFR 1.4(d) for signature requirements and certifications. Submit multiple forms if more than one signature is required, see below*.*

☐ * Total of ________ forms are submitted.

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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.

SUPPLEMENTAL SHEET FOR DECLARATION

ADDITIONAL INVENTOR(S)
Supplemental Sheet (for PTO/AIA/08,09) Page 1 of 1

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)

Qi Li

Inventor's Signature: [signature] Date (Optional):

Residence: City	State	Country
New Providence	NJ	USA

Mailing Address: 116 Hansell Road

City	State	Zip	Country
New Providence	NJ	07974	USA

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)

Inventor's Signature: Date (Optional):

Residence: City	State	Country

Mailing Address:

City	State	Zip	Country

Legal Name of Additional Joint Inventor, if any:

(*E.g.*, Given Name (first and middle (if any)) and Family Name or Surname)

Inventor's Signature: Date (Optional):

Residence: City	State	Country

Mailing Address:

City	State	Zip	Country

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).
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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.
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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.

Electronic Acknowledgement Receipt

EFS ID:	32368143
Application Number:	15293626
International Application Number:	
Confirmation Number:	4199
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Customer Number:	64188
Filer:	Ashok Tankha
Filer Authorized By:	
Attorney Docket Number:	CreativeTech_01RE_US
Receipt Date:	18-APR-2018
Filing Date:	14-OCT-2016
Time Stamp:	04:49:37
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	CreativeTech_01RE_US_Transmittal_Letter.pdf	24920 ed597fcc58932711861bd6a435b4b5506004e7c0	no	2

Warnings:

Information:					
2	Response After Final Action	CreativeTech_01RE_US_Response.pdf	159311 f40efc73a90b6d5bfc939aa973be9f7e06f8a5f0	no	22
Warnings:					
Information:					
3	Oath or Declaration filed	CreativeTech_01RE_US_Reissue_Declaration_First_Inventor.pdf	2005436 aa1caa3f6919720708840 44bd70576534f8643a4	no	3
Warnings:					
Information:					
4	Miscellaneous Incoming Letter	CreativeTech_01RE_US_Attached_Sheet.pdf	32571 dc37c8d20ef13400cc8068c675cfaab102bdb022	no	1
Warnings:					
Information:					
5	After Final Consideration Program Request	CreativeTech_01RE_US_sb0434.pdf	226515 a50e07f24cf137a528cc07dc9b03d093863ca1f4	no	2
Warnings:					
Information:					
6	Oath or Declaration filed	CreativeTech_01RE_US_Declaration_Second_Inventor.pdf	799220 5f1b324454d3c348d5dee632e8bec7bbc77ee115	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	3247973		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.
National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.
New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

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

TRANSMITTAL LETTER

1. **Attachments:** We are transmitting herewith the attached papers for the above identified patent application:

- **Response to final office action under "After Final Consideration Pilot Program 2.0" (22 pages); and**
- **Form SB434 for consideration under "After Final Consideration Pilot Program 2.0";**
- **Reissue application declaration by the inventor, Form PTO/AIA/05;**
- **Supplemental sheet for declaration, Form PTO/AIA/10; and**
- **Attached sheet (1 page).**

2. **General Authorization to charge or credit fees:** The Director is hereby authorized to charge any underpayment of fee or any other fee that may be required to deposit account # **600314**.

3. **Certificate Of Transmission Under 37 § CFR 1.8**: The undersigned hereby certifies that this Transmittal Letter and the papers as described in paragraph 1 hereinabove, are being electronically transmitted to the United States Patent and Trademark Office **via the USPTO electronic filing system** on this **18th** day of **April 2018.**

Respectfully submitted,

Date: April 18, 2018

/a tankha/
Ashok Tankha
Attorney For Applicant
Reg. No. 33,802

Correspondence Address
Lipton, Weinberger & Husick
36 Greenleigh Drive
Sewell, NJ 08080

Phone: 856-266-5145
Fax: 856-374-0246
Email: ash@ipprocurement.com

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875	Application or Docket Number 15/293,626	Filing Date 10/14/2016	☐ To be Mailed

ENTITY: ☐ LARGE ☒ SMALL ☐ MICRO

APPLICATION AS FILED – PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	RATE ($)	FEE ($)
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	
☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	minus 20 =	*	X $ =	
INDEPENDENT CLAIMS (37 CFR 1.16(h))	minus 3 =	*	X $ =	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	

APPLICATION AS AMENDED – PART II

AMENDMENT

04/18/2018	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE ($)	ADDITIONAL FEE ($)
Total (37 CFR 1.16(i))	* 35	Minus	** 35	= 0	x $50 =	0
Independent (37 CFR 1.16(h))	* 12	Minus	***12	= 0	x $230 =	0
☐ Application Size Fee (37 CFR 1.16(s))						
☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
					TOTAL ADD'L FEE	**0**

AMENDMENT

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE ($)	ADDITIONAL FEE ($)
Total (37 CFR 1.16(i))	*	Minus	**	=	X $ =	
Independent (37 CFR 1.16(h))	*	Minus	***	=	X $ =	
☐ Application Size Fee (37 CFR 1.16(s))						
☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
					TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

LIE
PATRICIA F. LEWIS

PTO/SB/06 (09-11)
Approved for use through 1/31/2014. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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.

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875	Application or Docket Number 15/293,626	Filing Date 10/14/2016	☐ To be Mailed

ENTITY: ☐ LARGE ☒ SMALL ☐ MICRO

APPLICATION AS FILED – PART I

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	RATE ($)	FEE ($)
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	
☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	minus 20 =	*	X $ =	
INDEPENDENT CLAIMS (37 CFR 1.16(h))	minus 3 =	*	X $ =	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	

APPLICATION AS AMENDED – PART II

AMENDMENT		(Column 1)		(Column 2)	(Column 3)		
	04/18/2018	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	* 35	Minus	** 35	= 0	x $50 =	0
	Independent (37 CFR 1.16(h))	* 12	Minus	***12	= 0	x $230 =	0
	☐ Application Size Fee (37 CFR 1.16(s))						
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
						TOTAL ADD'L FEE	**0**

AMENDMENT		(Column 1)		(Column 2)	(Column 3)		
		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	*	Minus	**	=	X $ =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X $ =	
	☐ Application Size Fee (37 CFR 1.16(s))						
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))						
						TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

LIE
PATRICIA F. LEWIS

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

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)

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

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. ______________________

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

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 05/02/2018

ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

EXAMINER	
ESCALANTE, OVIDIO	

ART UNIT	PAPER NUMBER
3992	

DATE MAILED: 05/02/2018

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

Search Notes	**Application/Control No.** 15293626	**Applicant(s)/Patent Under Reexamination** ZHU ET AL.
	Examiner OVIDIO ESCALANTE	**Art Unit** 3992

CPC- SEARCHED		
Symbol	**Date**	**Examiner**
G01S: 3/8055; 5/22; 3/801	9/26/17	OE
H04R: 3/005; 1/406; 2201/403; 2201/401	9/26/17	OE
H04M: 3/568	9/26/17	OE
Update	2/16/2018	OE
Update	4/20/18	OE

CPC COMBINATION SETS - SEARCHED		
Symbol	**Date**	**Examiner**

US CLASSIFICATION SEARCHED			
Class	**Subclass**	**Date**	**Examiner**
381	300. 57	9/26/17	OE
381	92, 94.1, 93	9/26/17	OE
Update	Update	2/16/18	OE
Update	Update	4/20/18	OE

* See search history printout included with this form or the SEARCH NOTES box below to determine the scope of the search.

SEARCH NOTES		
Search Notes	**Date**	**Examiner**
Requested Litigation Search	9/26/17	OE
EAST Search	9/26/17	OE
Reviewed Patent Prosecution History	9/26/17	OE
Updated EAST Search	2/16/18	OE
Updated EAST Search	4/20/18	OE

INTERFERENCE SEARCH			
US Class/ CPC Symbol	**US Subclass / CPC Group**	**Date**	**Examiner**
G01S:	3/8055; 5/22; 3/801	4/20/18	OE
H04R:	3/005; 1/406; 2201/403; 2201/401	4/20/18	OE
H04M:	3/568	4/20/18	OE
381	92, 94.1, 93	4/20/18	OE
381	300. 57	4/20/18	OE

OK TO ENTER: /O.E./

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.

UNITED STATES PATENT AND TRADEMARK OFFICE

Reissue Terminal Disclaimer Review Form

Application No. 15/293,626

Art Unit: 3992

Examiner: Ovidio Escalante

Original Patent Number of Patent to be Reissued is: 8861756

The Maintenance fee status is:

(●) up to date.

( ) not up to date. (Consult with SPRS)

Is there a terminal disclaimer filed and accepted during the prosecution of (i) the current reissue application, (ii) the underlying patent, and/or (iii) reexamination proceeding(s) of the underlying patent?

(●) NO

( ) YES (Complete the rest of the form)

This reissue patent is subject to Terminal Disclaimer(s) that was/were:

☐ filed and accepted (DISQ or DISQ.E.FILE) during the prosecution of the current reissue application. (Enter terminal disclaimer(s) filing date(s) below).

1. ____________ **2.** ____________ **3.** ____________

The underlying patent of the current reissue application is subject to Terminal Disclaimer(s) that was/were:

☐ accepted (DISQ or DISQ.E.FILE) and of record in the prosecution of the underlying patent and/or reexamination proceeding(s) of the underlying patent. (Enter application/control no(s) and terminal disclaimer(s) filing date(s) below).

1. ____________ ____________ **2.** ____________ ____________ **3.** ____________ ____________

(**Examiner's note:** Assign Doc Code "REIS.REVFORM" to this form.)

Bibliographic Data

Application No: 15293626

Foreign Priority claimed:	○ Yes	◉ No	
35 USC 119 (a-d) conditions met:	☐ Yes	☑ No	☐ Met After Allowance
Verified and Acknowledged:	/Ovidio Escalante/		
	Examiner's Signature		Initials
Title:	Microphone Array System		

FILING or 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
10/14/2016	381	3992	CreativeTech_01RE_US
RULE			

APPLICANTS

LI Creative Technologies, Inc., Florham Park, NJ, UNITED STATES

INVENTORS

Manli Zhu, Pearl River, NY, UNITED STATES

Qi Li, New Providence, NJ, UNITED STATES

CONTINUING DATA

This application is a REI of 13049877 03/16/2011 PAT 8861756

13049877 has PRO of 61403952 09/24/2010

FOREIGN APPLICATIONS

IF REQUIRED, FOREIGN LICENSE GRANTED**

11/07/2016

STATE OR COUNTRY

UNITED STATES

ADDRESS

ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080
UNITED STATES

FILING FEE RECEIVED

$4,080

Index of Claims	Application/Control No. 15293626	Applicant(s)/Patent Under Reexamination ZHU ET AL.
	Examiner OVIDIO ESCALANTE	Art Unit 3992

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

☒ Claims renumbered in the same order as presented by applicant ☐ CPA ☐ T.D. ☐ R.1.47

CLAIM		DATE								
Final	Original	09/26/2017	02/16/2018	04/20/2018						
	1	✓	✓	=						
	2	✓	✓	=						
	3	✓	✓	=						
	4	✓	✓	=						
	5	✓	✓	=						
	6	✓	✓	=						
	7	✓	✓	=						
	8	✓	✓	=						
	9	✓	✓	=						
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	32	✓	✓	=						
	33	✓	✓	=						
	34	✓	✓	=						
	35	✓	✓	=						

EAST Search History

EAST Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
S206	1531	G01S5/22.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:47
S205	57	G01S3/8055.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:47
S204	646	G01S3/801.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:47
S203	1914	H04R2201/403.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:46
S202	2486	H04R2201/401.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:46
S201	0	H04R221/401.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:46
S200	4796	H04R1/406.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:46
S199	11726	H04R3/005.cpc.	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:46
S198	1203	((381/300) or (381/57)).CCLS.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:45
S197	1	("8861756").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:44
S196	1	("6643355").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S195	1	("20050216580").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41

S194	1	("7113090").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S193	256	"7113090"	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:41
S192	5	(processor) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:41
S191	6	(digital adj signal adj processor or DSP) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:41
S190	17	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S189	2	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S188	151	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S187	216	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S186	1	("8861756").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S185	1	("7590075").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S184	11	"8554970"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S183	9	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size and motion	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S182	58	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S181	1	2010-C11185.NRAN.	DERWENT	OR	OFF	2018/04/20 06:41
S180	3	"2010020162"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S179	0	WO2010020162	US-PGPUB; USPAT; USOCR;	OR	OFF	2018/04/20 06:41

			DERWENT			
S178	9	"8861756"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S177	5	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank same beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S176	1243	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S175	2	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S174	7	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and adaptive adj filter$4	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S173	18	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone with array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S172	54	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S171	328	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S170	82	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S169	3	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay with sample$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S168	25	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S167	3	(("9648162") or ("20050231588") or ("20130242030")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S166	2	(("9648162") or ("20050231588")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S165	5	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S164	6	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41

S163	1	("8861698").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S162	1	("8358766").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S161	7	S160 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S160	27	(@rlad<="20100924" or @ad<="20100924") and S159	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S159	76	delay same distance same (origin or source or reference adj point) same angle and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S158	38	S157 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S157	115	(@rlad<="20100924" or @ad<="20100924") and S156	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S156	212	delay same distance same (origin or source or reference adj point) and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S155	2	(("20090141907") or ("20040161121")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S154	19	(@rlad<="20100924" or @ad<="20100924") and S153	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S153	26	delay same source same distance same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S152	248	delay same source same distance same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S151	13715	delay same source same distance same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S150	16	delay same sound adj source same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S149	257	delay same sound adj source same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S148	11	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S147	182	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S146	3351	delay same sound adj source same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S145	5	(processor) same beamform$3 same noise adj reduc$5 same	US-PGPUB; USPAT; USOCR;	OR	OFF	2018/04/20 06:41

		sound adj source	FPRS; EPO; JPO; DERWENT; IBM_TDB			
S144	6	(digital adj signal adj processor or DSP) same beamform$3 same noise adj reduc$5 same sound adj source	US-PGPUB; USPAT; USOCR; FPRS; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2018/04/20 06:41
S143	17	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S142	2	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S141	151	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1 and analog$2 adj digital	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S140	216	(@rlad<="20100924" or @ad<="20100924") and beamformer same digital adj analog$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S139	1	("8861756").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S138	1	("7590075").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S137	11	"8554970"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S136	9	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size and motion	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S135	58	(@rlad<="20100924" or @ad<="20100924") and beamformer same adaptive adj filter$3 and step adj size	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S134	1	2010-C11185.NRAN.	DERWENT	OR	OFF	2018/04/20 06:41
S133	3	"2010020162"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S132	0	WO2010020162	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S131	9	"8861756"	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S130	5	(@rlad<="20100924" or @ad<="20100924") and analysis adj filter adj bank same beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S129	1243	(@rlad<="20100924" or	US-PGPUB;	OR	OFF	2018/04/20

		@ad<="20100924") and analysis adj filter adj bank	USPAT; USOCR; DERWENT			06:41
S128	2	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and analysis adj filter adj bank	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S127	7	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and adaptive adj filter$4	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S126	18	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone with array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S125	54	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth same delay and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S124	328	(@rlad<="20100924" or @ad<="20100924") and (distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S123	82	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S122	3	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay with sample$1	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S121	25	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same azimuth and microphone adj array and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S120	3	(("9648162") or ("20050231588") or ("20130242030")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S119	2	(("9648162") or ("20050231588")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S118	5	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth and delay	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S117	6	(@rlad<="20100924" or @ad<="20100924") and (elevation with distance) same microphone adj array and azimuth	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S116	1	("8861698").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S115	1	("8358766").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S114	7	S113 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S113	27	(@rlad<="20100924" or @ad<="20100924") and S112	US-PGPUB; USPAT; USOCR;	OR	OFF	2018/04/20 06:41

			DERWENT			
S112	76	delay same distance same (origin or source or reference adj point) same angle and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S111	38	S110 and microphone adj array	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S110	115	(@rlad<="20100924" or @ad<="20100924") and S109	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S109	212	delay same distance same (origin or source or reference adj point) and beamformer and spatial	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S108	2	(("20090141907") or ("20040161121")).PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2018/04/20 06:41
S107	19	(@rlad<="20100924" or @ad<="20100924") and S106	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S106	26	delay same source same distance same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S105	248	delay same source same distance same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S104	13715	delay same source same distance same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S103	16	delay same sound adj source same (origin or source or reference adj point) and beamformer and spatial adj location	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S102	257	delay same sound adj source same (origin or source or reference adj point) and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S101	11	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle and beamformer	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S100	182	delay same sound adj source same (origin or source or reference adj point) and azimuth adj angle	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41
S99	3351	delay same sound adj source same (origin or source or reference adj point)	US-PGPUB; USPAT; USOCR; DERWENT	OR	OFF	2018/04/20 06:41

EAST Search History (Interference)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
S218	21	S216 and (microphone and sound and array and beamform$4 and noise and angle).clm.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:55
S217	65	S216 and (microphone and sound and array and beamform$4 and noise).clm.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:55

S216	7525	S207 S208 S209 S210 S211 S212 S213 S214 S215	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:54
S215	1246	H04M3/568.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:49
S214	385	G01S5/22.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:49
S213	18	G01S3/8055.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:49
S212	174	G01S3/801.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:49
S211	541	H04R2201/403.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:48
S210	717	H04R2201/401.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:48
S209	1471	H04R1/406.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:48
S208	3650	H04R3/005.cpc.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:48
S207	1165	((381/300) or (381/57)).CCLS.	US-PGPUB; USPAT	OR	OFF	2018/04/20 06:45

4/20/2018 8:34:23 AM
C:\Users\oescalante\Documents\EAST\Workspaces\15293626.wsp

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	15293626	ZHU ET AL.
	Examiner	**Art Unit**
	OVIDIO ESCALANTE	3992

CPC					
Symbol				**Type**	**Version**
G01S	3	/	8055	F	2013-01-01
G01S	3	/	801	I	2013-01-01
H04M	3	/	568	A	2013-01-01
H04R	1	/	406	I	2013-01-01
H04R	2201	/	401	A	2013-01-01
H04R	2201	/	403	A	2013-01-01
H04R	3	/	005	I	2013-01-01
G01S	5	/	22	I	2013-01-01
		/			
		/			
		/			
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		/			
		/			
		/			

CPC Combination Sets							
Symbol				**Type**	**Set**	**Ranking**	**Version**
		/					
		/					

NONE (Assistant Examiner)	(Date)	**Total Claims Allowed:** 35	
/OVIDIO ESCALANTE/ Primary Examiner.Art Unit 3992 (Primary Examiner)	4/20/2018 (Date)	O.G. Print Claim(s) 22	O.G. Print Figure 1

Issue Classification	**Application/Control No.** 15293626	**Applicant(s)/Patent Under Reexamination** ZHU ET AL.
	Examiner OVIDIO ESCALANTE	**Art Unit** 3992

US ORIGINAL CLASSIFICATION	
CLASS	**SUBCLASS**
381	300

CROSS REFERENCE(S)

CLASS	**SUBCLASS (ONE SUBCLASS PER BLOCK)**				
381	57				

INTERNATIONAL CLASSIFICATION

CLAIMED					**NON-CLAIMED**
H	0	4	R	25 / 00 (2006.0)	
H	0	3	G	3 / 20 (2006.01.01)	

NONE (Assistant Examiner) (Date)	**Total Claims Allowed:** 35	
/OVIDIO ESCALANTE/ Primary Examiner.Art Unit 3992 4/20/2018 (Primary Examiner) (Date)	O.G. Print Claim(s) 22	O.G. Print Figure 1

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	15293626	ZHU ET AL.
	Examiner	**Art Unit**
	OVIDIO ESCALANTE	3992

☒ **Claims renumbered in the same order as presented by applicant** ☐ **CPA** ☐ **T.D.** ☐ **R.1.47**

Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original

NONE (Assistant Examiner)	(Date)	**Total Claims Allowed:** 35	
/OVIDIO ESCALANTE/ Primary Examiner.Art Unit 3992 (Primary Examiner)	4/20/2018 (Date)	O.G. Print Claim(s) 22	O.G. Print Figure 1

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)

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

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 Ash Tankha
2 Lipton, Weinberger & Husick
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: LI Creative Technologies, Inc.

(B) RESIDENCE: (CITY and STATE OR COUNTRY): Florham Park, NJ (US)

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 /a tankha/ Date August 02, 2018

Typed or printed name Ashok Tankha Registration No. 33802

Electronic Patent Application Fee Transmittal

Application Number:	15293626
Filing Date:	14-Oct-2016
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Filer:	Ashok Tankha
Attorney Docket Number:	CreativeTech_01RE_US

Filed as Small Entity

Filing Fees for **Utility under 35 USC 111(a)**

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
REISSUE ISSUE FEE	2511	1	500	500

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**500**

Electronic Acknowledgement Receipt

EFS ID:	33351082
Application Number:	15293626
International Application Number:	
Confirmation Number:	4199
Title of Invention:	Microphone Array System
First Named Inventor/Applicant Name:	Manli Zhu
Customer Number:	64188
Filer:	Ashok Tankha
Filer Authorized By:	
Attorney Docket Number:	CreativeTech_01RE_US
Receipt Date:	02-AUG-2018
Filing Date:	14-OCT-2016
Time Stamp:	01:16:09
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$500
RAM confirmation Number	080218INTEFSW01473400
Deposit Account	
Authorized User	

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Issue Fee Payment (PTO-85B)	CreativeTech_01RE_US_Issue_Fee.pdf	380888 a7aa66d09736b6c1c5ac6b0fbdc0470383d238ee	no	1
Warnings:					
Information:					
2	Fee Worksheet (SB06)	fee-info.pdf	29682 254f095347b382e8cc106b459f1f6558cb219025	no	2
Warnings:					
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Notice of References Cited	Application/Control No. 15/293,626	Applicant(s)/Patent Under Reexamination ZHU ET AL.	
	Examiner OVIDIO ESCALANTE	Art Unit 3992	Page 1 of 1

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*	B	US-2010/0241426 A1	09-2010	ZHANG; Chen	G10L21/0208	704/226
*	C	US-2011/0096915 A1	04-2011	Nemer; Elias	H04M3/568	379/158
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	G	US-				
	H	US-				
	I	US-				
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	K	US-				
	L	US-				
	M	US-				

FOREIGN PATENT DOCUMENTS

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	N	WO 2010020162 A1	02-2010	~~CHINA~~ WO	WANG DONGQI	H04R1/403
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	P					
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	T					

Change(s) applied
to document,
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5/30/2018

NON-PATENT DOCUMENTS

*		Include as applicable: Author, Title Date, Publisher, Edition or Volume, Pertinent Pages)
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APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
15/293,626	09/18/2018	RE47049	CreativeTech_01RE_US	4199

64188 7590 08/29/2018

ASHOK TANKHA
36 GREENLEIGH DRIVE
SEWELL, NJ 08080

ISSUE NOTIFICATION

The projected patent number and issue date are specified above.

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 Application Assistance Unit (AAU) of the Office of Data Management (ODM) at (571)-272-4200.

APPLICANT(s) (Please see PAIR WEB site http://pair.uspto.gov for additional applicants):

Manli Zhu, Pearl River, NY;
LI Creative Technologies, Inc., Florham Park, NJ, Assignee (with 37 CFR 1.172 Interest);
Qi Li, New Providence, NJ;

AO 120 (Rev. 08/10)

TO: **Mail Stop 8 Director of the U.S. Patent and Trademark Office P.O. Box 1450 Alexandria, VA 22313-1450**	**REPORT ON THE FILING OR DETERMINATION OF AN ACTION REGARDING A PATENT OR TRADEMARK**

In Compliance with 35 U.S.C. § 290 and/or 15 U.S.C. § 1116 you are hereby advised that a court action has been filed in the U.S. District Court for the Eastern District of Texas - Marshall Division on the following

☐ Trademarks or ☒ Patents. (☐ the patent action involves 35 U.S.C. § 292.):

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:19-cv-123	April 15, 2019	Eastern District of Texas - Marshall Division

PLAINTIFF	DEFENDANT
VOCALIFE LLC	AMAZON.COM, INC. and AMAZON.COM LLC

PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 8,861,756	October 14, 2014	VOCALIFE LLC
2 RE47,049	September 18, 2018	VOCALIFE LLC
3		
4		
5		

In the above—entitled case, the following patent(s)/ trademark(s) have been included:

DATE INCLUDED	INCLUDED BY ☐ Amendment ☐ Answer ☐ Cross Bill ☐ Other Pleading	
PATENT OR TRADEMARK NO.	**DATE OF PATENT OR TRADEMARK**	**HOLDER OF PATENT OR TRADEMARK**
1		
2		
3		
4		
5		

In the above—entitled case, the following decision has been rendered or judgement issued:

DECISION/JUDGEMENT

CLERK	(BY) DEPUTY CLERK	DATE

Copy 1—Upon initiation of action, mail this copy to Director Copy 3—Upon termination of action, mail this copy to Director
Copy 2—Upon filing document adding patent(s), mail this copy to Director Copy 4—Case file copy

Trials@uspto.gov
571-272-7822

Paper 22
Entered: October 28, 2020

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

AMAZON.COM, INC.,
Petitioner,

v.

VOCALIFE LLC,
Patent Owner.

IPR2020-00864
Patent RE47,049 E

Before AMANDA F. WIEKER, MONICA S. ULLAGADDI, and JASON M. REPKO, *Administrative Patent Judges.*

REPKO, *Administrative Patent Judge.*

DECISION
Denying Institution of *Inter Partes* Review
35 U.S.C. § 314

I. INTRODUCTION

Amazon.com, Inc. ("Petitioner") filed a petition to institute *inter partes* review of claims 1–8, 19, 20, 22–25, and 30 of U.S. Patent No. RE47,049 E (Ex. 1001, "the '049 patent"). Paper 1 ("Pet."). Vocalife LLC ("Patent Owner") filed a Preliminary Response. Paper 8 ("Prelim. Resp."). We authorized additional briefing to address Patent Owner's argument that we should deny institution of the Petition under § 314(a). Paper 9. Petitioner filed a Reply. Paper 10 ("Reply"). Patent Owner filed a Sur-reply. Paper 12 ("Sur-reply"). After the conclusion of the parallel trial in district court, we authorized the parties to file another set of briefs. Paper 19 ("Pet. Post-Trial Brief"), Paper 21 ("PO Post-Trial Brief").

To institute an *inter partes* review, we must determine "that there is a reasonable likelihood that the petitioner would prevail with respect to at least 1 of the claims challenged in the petition." 35 U.S.C. § 314(a). But the Board has discretion to deny a petition even when a petitioner meets that threshold. *Id.*; *see, e.g.*, *Cuozzo Speed Techs., LLC v. Lee*, 136 S. Ct. 2131, 2140 (2016) ("[T]he agency's decision to deny a petition is a matter committed to the Patent Office's discretion."); *NHK Spring Co. v. Intri-Plex Techs., Inc.*, IPR2018-00752, Paper 8 (PTAB Sept. 12, 2018) (precedential); Patent Trial and Appeal Board Consolidated Trial Practice Guide 64 (Nov. 20, 2019), http://www.uspto.gov/TrialPracticeGuideConsolidated (identifying considerations that may warrant exercise of this discretion).

For the reasons discussed below, we exercise our discretion under § 314(a) to deny institution.

A. Related Matters

According to the parties, the ’049 patent is involved in *Vocalife LLC v. Amazon.com, Inc.*, No. 2:19-cv-00123-JRG (E.D. Tex. filed Apr. 16, 2019). Pet. 90; Paper 4, 2.

B. The ’049 Patent

The ’049 patent generally relates to enhancing a target sound signal, such as a speech signal, while suppressing ambient noise. *See* Ex. 1001, 2:5–11. This enhancement can be applied to signals from a microphone array, like those in mobile phones, for example. *See, e.g.*, *id.* at 18:49–55. According to the patent, conventional microphone arrays are used for radar and sonar. *Id.* at 1:42–46. Narrow-band techniques used by these systems, though, are unsuitable for speech signals captured by smaller devices because those signals have an extremely wide bandwidth relative to the center frequency. *Id.* at 1:46–50. And conventional arrays for broadband speech are too bulky to be used in mobile devices. *Id.* at 1:50–55.

To enhance the target sound signal in broadband-speech applications, the ’049 patent uses sound-source localization, adaptive beamforming, and noise reduction. *Id.* at 2:11–14. Figure 2, below, shows an example system. *Id.* at 3:66–67.

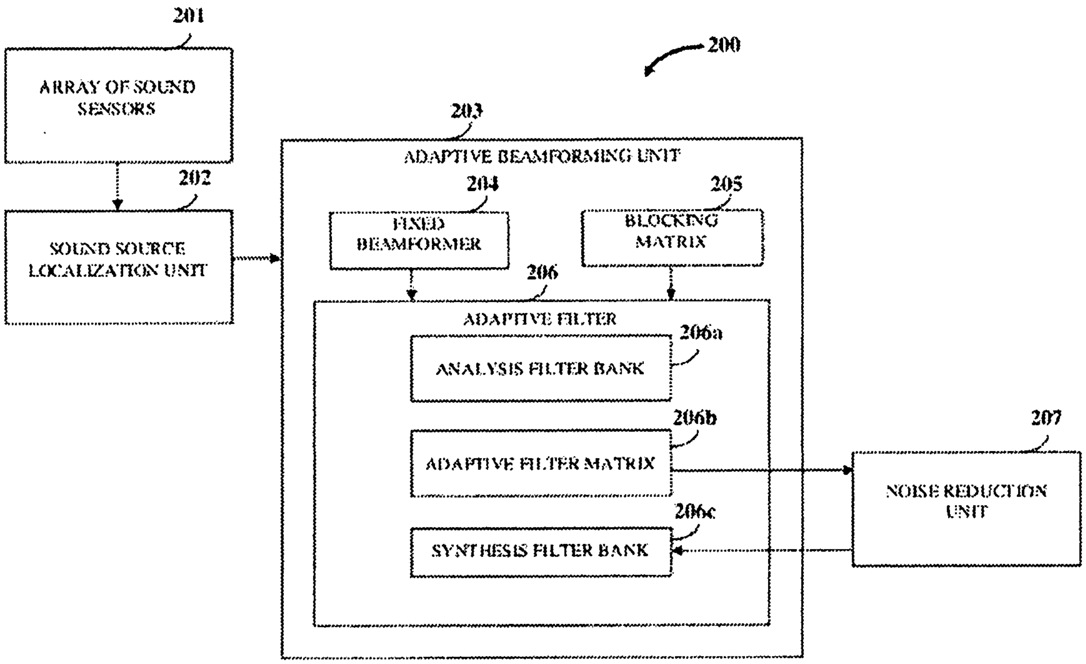

FIG. 2

Figure 2, above, shows system 200 with sound-source localization unit 202, adaptive-beamforming unit 203, and noise-reduction unit 207. *Id.* at 6:32–38.

In system 200, array 201 receives the sound signal. *Id.* at 6:48–53. Sound source localization unit 202 estimates a target sound signal's location. *Id.* at 6:54–56. Adaptive beamforming unit 203 steers the array's directivity pattern to the target sound signal. *Id.* at 6:60–64. This enhances the target sound signal and partially suppresses ambient noise signals. *Id.* Noise reduction unit 207 then further suppresses the ambient noise signals. *Id.* at 7:9–11.

Claims 1, 20, 22, and 30 are independent. Claim 1 is reproduced below.

> 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] *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.

Ex. 1001, 21:27–22:3

C. Evidence

Name	Reference	Exhibit No.
Saric	WO 2008/041878 A2, published April 10, 2008	1005
Dmochowski	Jacek Dmochowski et al., *Direction of Arrival Estimation Using the Parameterized Spatial Correlation Matrix*, 15 IEEE Transactions on Audio, Speech, and Language Processing 4, 1327–39 (2007)	1006
Li	Qi (Peter) Li et al., *A Portable USB-Based Microphone Array Device for Robust Speech Recognition*, 2009 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2009), 1301–04 (2009)	1007
Brandstein	Michael Brandstein & Darren Ward (Eds.), *Microphone Arrays: Signal Processing Techniques And Applications* (Springer-Verlag Berlin Heidelberg 2001)	1010
Abutalebi	US 2004/0071284 A1, published Apr. 15, 2004	1011
Greenberg	Julie E. Greenberg et al., *Evaluation of an Adaptive Beamforming Method for Hearing Aids*, Journal of the Acoustical Society of America 91 (3), 1662–76 (1992)	1012
Hoshuyama	Osamu Hoshuyama et al., *A Realtime Robust Adaptive Microphone Array Controlled by an SNR Estimate*, Proceedings of the 1998 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP '98), 3605–08 (1998)	1013

D. Asserted Grounds

Petitioner asserts that claims 1–8, 19, 20, 22–25, and 30 are unpatentable on the following grounds. Pet. 10–11.

Claims Challenged	Pre-AIA[1] 35 U.S.C. §	Reference(s)/Basis
1, 7, 19, 20, 22, 30	103	Saric, Dmochowski
1–4, 7, 19, 20, 22–24, 30	103	Saric, Dmochowski, Brandstein
6, 24	103	Saric, Dmochowski, Brandstein, Greenberg
6, 24	103	Saric, Dmochowski, Brandstein, Hoshuyama
5, 8, 25	103	Saric, Dmochowski, Brandstein, Abutalebi
1–4, 6, 7, 19, 22–24	103	Li, Brandstein
1–4, 7, 19, 20, 22–24, 30	103	Li, Brandstein, Dmochowski
6, 24	103	Li, Brandstein, Dmochowski, Greenberg
6, 24	103	Li, Brandstein, Dmochowski, Hoshuyama
5, 8, 25	103	Li, Brandstein, Dmochowski, Abutalebi

II. ANALYSIS

A. 35 U.S.C. § 314(a)

Under § 314(a), the Director has discretion to deny institution. In determining whether to exercise that discretion on behalf of the Director, we are guided by the Board's precedential decision in *NHK Spring Co. v. Intri-Plex Technologies, Inc.*, IPR2018-00752, Paper 8 (PTAB Sept. 12, 2018).

[1] Congress amended § 103 when it passed the Leahy-Smith America Invents Act (AIA). Pub. L. No. 112–29, § 3(c), 125 Stat. 284, 287 (2011). Here, the previous version of § 103 applies.

In *NHK*, the Board found that the "advanced state of the district court proceeding" was a "factor that weighs in favor of denying" the petition under § 314(a). *NHK*, Paper 8 at 20. The Board determined that "[i]nstitution of an *inter partes* review under these circumstances would not be consistent with 'an objective of the AIA . . . to provide an effective and efficient alternative to district court litigation.'" *Id.* (citing *Gen. Plastic Indus. Co. v. Canon Kabushiki Kaisha*, IPR2016-01357, Paper 19 at 16–17 (PTAB Sept. 6, 2017) (precedential as to § II.B.4.i).

"[T]he Board's cases addressing earlier trial dates as a basis for denial under *NHK* have sought to balance considerations such as system efficiency, fairness, and patent quality." *Apple Inc. v. Fintiv, Inc.*, IPR2020-00019, Paper 11 at 5 (PTAB Mar. 20, 2020) (precedential) (collecting cases) ("the *Fintiv* Order"). The *Fintiv* Order sets forth six non-exclusive factors for determining "whether efficiency, fairness, and the merits support the exercise of authority to deny institution in view of an earlier trial date in the parallel proceeding." *Id.* at 6. These factors consider

1. whether the court granted a stay or evidence exists that one may be granted if a proceeding is instituted;
2. proximity of the court's trial date to the Board's projected statutory deadline for a final written decision;
3. investment in the parallel proceeding by the court and the parties;
4. overlap between issues raised in the petition and in the parallel proceeding;
5. whether the petitioner and the defendant in the parallel proceeding are the same party; and
6. other circumstances that impact the Board's exercise of discretion, including the merits.

Id. In the sections that follow, we discuss each factor and perform a holistic analysis of the facts and evidence underlying these factors.

Factor 1: Whether the court granted a stay or evidence exists that one may be granted if a proceeding is instituted

"A district court stay of the litigation pending resolution of the PTAB trial allays concerns about inefficiency and duplication of efforts. This fact has strongly weighed against exercising the authority to deny institution under *NHK*." *Id.*

According to the parties, the parallel district-court proceeding has not been stayed. *See, e.g.*, Prelim. Resp. 25. Nor has Petitioner sought a stay. *Id.* at 11; Sur-reply 2. Patent Owner argues that there is no evidence that a stay would be granted. Prelim. Resp. 11. In fact, the trial in district court concluded on October 8, 2020—several weeks before this institution deadline. Ex. 1042, 1 (Fifth Amended Docket Control Order); Ex. 3001 (Verdict Form).

Absent specific evidence, we decline to speculate how the district court would rule on a stay request. *See Sand Revolution II LLC v. Cont'l Intermodal Grp.*, IPR2019-01393, Paper 24, 7 (PTAB June 16, 2020) (informative, designated July 13, 2020) (declining to predict how the district court in the related litigation will proceed). No specific evidence in this case suggests that the district court will grant a stay. So this factor is neutral.

Factor 2: Proximity of the court's trial date to the Board's projected statutory deadline for a final written decision

"If the court's trial date is earlier than the projected statutory deadline, the Board generally has weighed this fact in favor of exercising authority to deny institution under *NHK*." *Fintiv* Order at 9.

Here, the trial date has passed. Ex. 1042, 1 (Fifth Amended Docket Control Order); Ex. 3001 (Verdict Form). The projected deadline for the Board's final decision is a year from now. Because the trial date is substantially earlier than the projected statutory deadline for the Board's final decision, this factor strongly favors exercising our discretion to deny institution.

Factor 3: Investment in the parallel proceeding by the court and the parties

"The Board also has considered the amount and type of work already completed in the parallel litigation by the court and the parties at the time of the institution decision." *Fintiv* Order at 9. "[M]ore work completed by the parties and court in the parallel proceeding tends to support the arguments that the parallel proceeding is more advanced, a stay may be less likely, and instituting would lead to duplicative costs." *Id.* at 10.

Patent Owner asserts that the parties and the court will have invested a substantial amount in the parallel litigation by the time this decision issues. *See* Prelim. Resp. 20–25. We agree. The parties have litigated the proceeding through trial with only the potential for post-trial briefing remaining. For example, a *Markman* hearing has been held, and a claim construction order has been issued. *Id.* at 21. The parties and the court invested in, and completed, expert and fact discovery. *Id.* at 20–22. The trial began October 1, 2020 and concluded with a jury verdict on October 8, 2020. Ex. 1042 (Fifth Amended Docket Control Order); Ex. 3001 (Verdict Form).

Considering the investment in the parallel proceeding by the court and the parties, we determine that this factor strongly favors exercising our discretion to deny institution.

Factor 4: Overlap between issues raised in the petition and in the parallel proceeding

"[I]f the petition includes the same or substantially the same claims, grounds, arguments, and evidence as presented in the parallel proceeding, this fact has favored denial." *Fintiv* Order at 12.

Petitioner argues that the trial involved claims 1 and 8, but thirteen additional claims are challenged in the Petition. Pet. Post-Trial Brief 1. Yet the other independent claims that are challenged here are similar to claim 1. As noted in the Petition, "Claim 22 differs slightly from claim 1." Pet. 79. In fact, claim 22 lacks some limitations found in claim 1: Claim 22 recites "providing a microphone array system comprising an array of sound sensors," without claim 1's limitations about how to position the sensors. *Id.* Claim 22 recites "a beamforming unit" instead of "an adaptive" one, as recited in claim 1. *Id.* Also, claim 22 recites a delay with respect to a reference point, which enables two or more sensors instead of the "plurality of configurations" in claim 1. *Id.* As for claim 30, Petitioner asserts that this claim "is the same as claim 20," except for limitations that "would have been obvious for the same reasons as claim 22." *Id.* at 36. Thus, the similarities between claim 1 and the other independent claims will likely lead to substantially the same arguments and evidence here as presented in the parallel proceeding.

The prior-art combinations considered in the district-court proceeding cover nearly all the claims challenged in the Petition. In particular, Patent

Owner argues that Petitioner's invalidity case through pretrial and trial were based on various combinations of Saric, Dmochowski, Brandstein, Abutalebi, and Li, which are the basis for most challenges in this Petition. PO Post-Trial Brief 1; Pet. 10–11. Petitioner argues that the jury did not consider Grounds 1a–1e of the Petition, which are based on Saric. Pet. Post-Trial Brief 1.

Even though Petitioner did not present Saric to the jury at trial, the grounds based on Li and Brandstein (Grounds 2a–2e) cover the same claims as the grounds based on Saric (Grounds 1a–1e). *See* Pet. 10–11. And the grounds based on Li and Brandstein, which were presented at trial, account for half the prior-art combinations in this proceeding. *See id.*; Ex. 1043 (Excerpts of Trial Transcript).

To be sure, Hoshuyama and Greenberg, which are relied upon in the Petition, were not before the district court. Reply 3 (citing Ex. 2002, 1). Petitioner, though, uses Hoshuyama and Greenberg only in the challenges to dependent claims 6 and 24. Pet. 10–11.

As noted above, Petitioner argues that fifteen claims are challenged here, but only two were tried in district court. Pet. Post-Trial Brief 2. According to Petitioner, Patent Owner has not granted it a covenant not to sue on the other claims, and invalidating those claims would require another proceeding. Pet. 88; Reply 2–3. This argument, though, is speculative and does not outweigh our concerns about inefficiency and the possibility of conflicting decisions. Petitioner has not persuasively explained any benefit to resolving the patentability of the additionally challenged claims when they are not asserted against Petitioner.

Concerns over inefficiency and the possibility of the Board and the district court issuing conflicting decisions underlie this factor. *Fintiv* Order at 12. Should the Board institute, there would be some overlap between the proceedings, which would implicate these concerns for at least half the prior art combinations and claims 1 and 8. On balance, this factor favors exercising our discretion to deny institution.

Factor 5: Whether the petitioner and the defendant in the parallel proceeding are the same party

If the petitioner and the defendant in the parallel proceeding are the same and the validity issues are scheduled to be determined in the parallel proceeding first, this factor weighs in favor of denial. *Fintiv,* Paper 15 at 15 (informative) (applying *Fintiv* Factor 5); *Sand Revolution*, Paper 24 at 12–13 (informative) ("Although it is far from an unusual circumstance that a petitioner in *inter partes* review and a defendant in a parallel district court proceeding are the same, or where a district court is scheduled to go to trial before the Board's final decision would be due in a related *inter partes* review, this factor weighs in favor of discretionary denial.").

Here, the parties do not dispute that Petitioner is the defendant in the parallel proceeding. *See, e.g.*, Pet. 90, Prelim. Resp. 11. Also, the validity of claims 1 and 8 have already been determined in the parallel proceeding. So this factor, when considered in light of the circumstances of this case, strongly favors exercising our discretion to deny institution.

Factor 6: Other circumstances that impact the Board's exercise of discretion, including the merits

When considering whether to exercise discretion to deny a petition, we assess "all the relevant circumstances in the case, including the merits." *Fintiv* Order at 14. Under *Fintiv* Factor 6, we consider the strengths and weaknesses of the Petition's merits: stronger merits typically favor institution, and weaker merits favor denial. *Id.* at 15–16.

In the parallel proceeding, the jury found that claims 1 and 8 were not invalid. Pet. Post-Trial Brief 1; Ex. 3001 (Verdict Form). Li and the other references used in Grounds 2a–2e of the Petition were presented at trial. *See, e.g.*, Ex. 1043 (Excerpts from Trial Transcripts). According to the parties, the expert did not present Saric as part of his testimony. PO Post-Trial Brief 1–2; Pet. Post-Trial Brief 1–2. Petitioner uses Saric in Grounds 1a–1e of the Petition. Pet. 10–11.

A full merits evaluation is not necessary in the analysis under *Fintiv* Factor 6. *Fintiv* Order at 15. Here, it is sufficient to look only at the grounds based on Saric (Grounds 1a–1e). Every claim challenged under Grounds 1a–1e of the Petition was also challenged under Grounds 2a–2e. *See* Pet. 10–11. And according to the parties, the jury reached its verdict based on a presentation of prior art used in Grounds 2a–2e. *See* Pet. Post-Trial Brief; PO Post-Trial Brief. As discussed in detail below, the merits of the grounds based on Saric are weak.

In particular, Petitioner challenges independent claims 1, 20, and 22 using prior-art combinations involving Saric. Pet. 10–11. Claim 1 recites, in part:

> 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*.

Ex. 1001, 21:29–36. Claims 20 and 22 recite similar limitations. *Id.* at 24:40–46 (claim 20), 25:56–61 (claim 22).

The patent explains that a digital signal processor receives digital sound signals and "implements the sound source localization unit 202, the adaptive beamforming unit 203, and the noise reduction unit 207." *Id.* at 15:25–29. Petitioner asserts that the recited "source localization unit" should be limited to a digital signal processor executing the SRP-PHAT[2] algorithm. Pet. 18. Petitioner applies this construction when discussing the prior art. *See id.* at 23–26. Although Petitioner does not provide an explicit construction for the other units (*id.* at 15–18), Petitioner interprets them in a similar way—i.e., as an algorithm on a digital signal processor—when discussing the prior art (*see id.* at 23–26).

Specifically, Petitioner asserts that the combination of Saric and Dmochowski teaches the recited digital signal processor and the three units: (1) it would have been obvious to incorporate Dmochowski's SRP (steered-response power) algorithm for sound source localization in Saric to obtain the sound source localization unit; (2) Saric teaches the adaptive beamforming unit by using "a microphone array processing signal algorithm for adaptive beam forming (ABF)"; and (3) Saric teaches the noise reduction unit because Saric describes an algorithm for reducing stationary noise, non-steady noise, and residual echo. Pet. 23–25 (citing Ex. 1005, 4:24–33, 6:31–36, 8:2–5, 8:24–26, 16:3–29, Figs. 3 and 5; Ex. 1006, 1330; Ex. 1015 ¶¶ 107, 112–115). Petitioner asserts that Saric integrates the three recited units in a digital signal processor. *Id.* at 25–26 (citing Ex. 1005, 4:29–35, Fig. 1; Ex. 1015 ¶¶ 116–118).

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[2] SRP-PHAT means steered response power-phase transform. Ex. 1001, 11:28.

Patent Owner argues that Petitioner has not sufficiently shown that the three recited units "are integrated in a digital signal processor," as recited. Prelim. Resp. 27. We agree.

For the recited integration, the Petition relies on one sentence and a figure from Saric. Pet. 25 (citing Ex. 1005, 4:29-35; Ex. 1015 ¶¶ 116-118). The cited sentence is,

> *DSP run* a few complex algorithms: acoustic echo canceling algorithm (AEC), microphone array processing signal algorithm for adaptive beam forming (ABF) and its directivity characteristics, estimation algorithm for direction of arrival (DOA) of useful signal for indoor localization of speaker, in other words speaker room localization, algorithm for reduction of stationary noise, non-steady noise and residual echo (NR-Noise Reduction) and algorithm for system automatic gain control (AGC), because of compensation between different speaker distance from the microphone array.

Ex. 1005, 4:29–35 (emphasis added). Thus, the issue of whether Saric teaches a digital signal processor running all the recited algorithms turns on the meaning of the phrase "DSP run." "[H]ardware engineers use 'DSP' to mean Digital Signal *Processor*," but "algorithm developers use 'DSP' to mean Digital Signal *Processing*." Ex. 1033, 5 ("The Scientist's and Engineer's Guide to Digital Signal Processing") (emphasis added). In describing the invention, Saric defines DSP as "digital signal *processing*." Ex. 1005, 15:26 (emphasis added). Thus, we are not persuaded that Saric's phrase "DSP run" refers to a digital signal processor executing the algorithms, as the claims require. *See id.*

Petitioner also cites Saric's Figure 1. Pet. 25. That figure, below, shows a single box labeled with the word "DSP." Ex. 1005, Fig. 1.

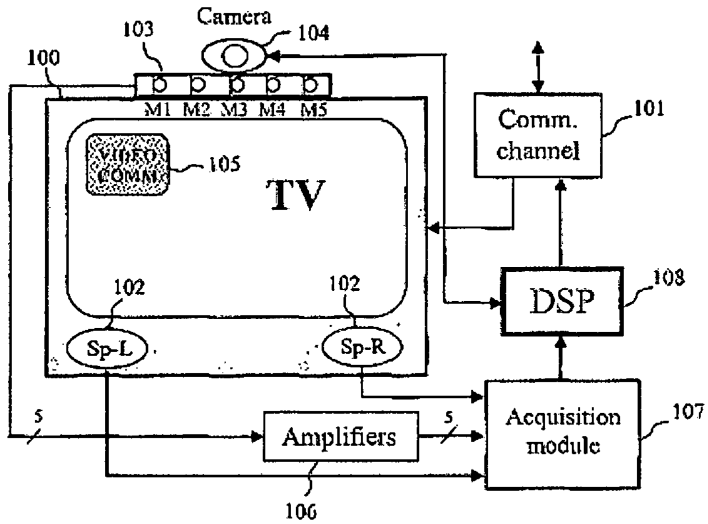

FIG. 1.

Saric describes Figure 1, above, as having "elements." *Id.* at 6. Saric divides those elements into the boxes shown Figure 3, reproduced below with annotations. Pet. 25.

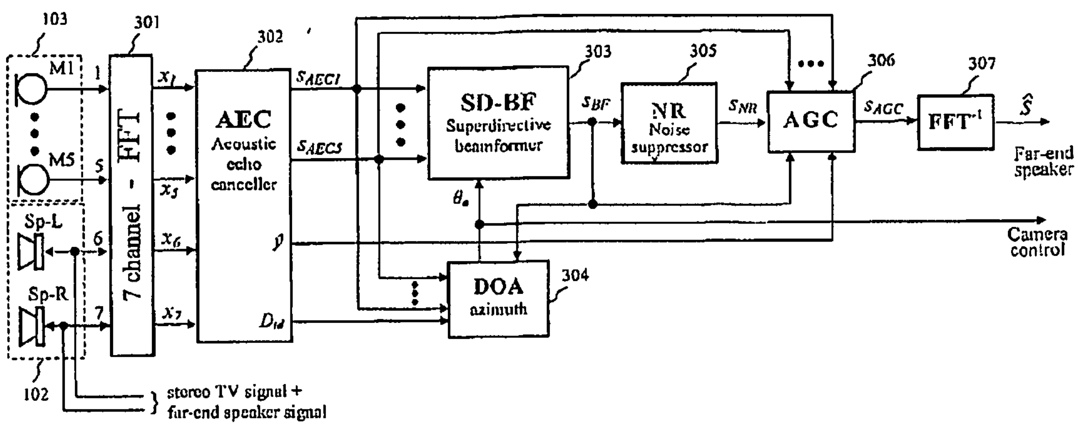

FIG. 3.

Saric's Figure 3 is shown above with Petitioner's annotations on noise suppressor 305 (blue), DOA azimuth 304 (purple), and SD-BF super directive beamformer 303 (pink). *Id.* Petitioner maps these three units to the recited noise reduction unit, sound source localization unit, and adaptive beamforming unit, which "are integrated in a digital signal processor." *See* *id.* at 25–26. Yet Saric shows them as three distinct boxes in Figure 3. This suggests that Saric's diagrams are merely illustrating various levels of abstraction for the process, instead of signifying a specific hardware processor executing those algorithms.

To the extent that Saric's boxes can be interpreted as hardware components, the description of the invention suggests that there are multiple processors that execute the digital signal processing:

> For example, these techniques can be implemented into the hardware, software, or in the combination two of them. In the hardware implementation we can use specific integrated circuits (ASIC), *processors of the digital signal processing (DSP),* programmable logical devices (PDL or FPGA) and others electronic circuits, designed in that way, to be able to accomplish a given invention functions.

Ex. 1005, 15:23–28 (emphasis added). That is, this passage states that Saric's digital signal processing is accomplished by multiple processors—not all three units integrated in a digital signal processor as required by the claims.

Petitioner's obviousness rationale is based on the premise that Saric uses a specialized processor. Pet. 21–22. For example, the district court determined that "digital signal processor" means "microprocessor that is specialized for mathematical processing of digital signals." Ex. 1028, 24 (Claim Construction Memorandum Opinion and Order). In its analysis, the district court expressly rejected the construction that a digital signal processor is simply "a device that processes digital signals." *Id.* at 21–22. Although Petitioner argues that this term need not be construed here (Pet. 15), Petitioner's obviousness rationale refers to a digital signal processor's ability handle "computation load" (*see, e.g.*, *id.* at 22; Ex. 1015 ¶ 109). Yet Petitioner has not sufficiently shown on this record that

Saric's processors confer the advantages of a digital signal processor. *See* Pet. 21–22, 23–26. Rather, Saric simply refers to "processors" generally.[3] Ex. 1005, 15:23–28.

In sum, Petitioner has not shown a reasonable likelihood of prevailing in showing that Saric, in combination with the other references, teaches or suggests that the "sound source localization unit, said adaptive beamforming unit, and said noise reduction unit are integrated in a digital signal processor." At best, Saric's disclosure is ambiguous.

In Grounds 1a–1e, Petitioner relies on Saric in combination with other references to teach or suggest the units integrated in a digital signal processor recited in all challenged independent claims. *See* Pet. 25 (claim 1), 34–36 (claim 22), 36–38 (claim 20). So none of the challenges based on Saric meet the Board's institution standard. Thus, *Fintiv* Factor 6 favors denial.

B. Summary

We consider the *Fintiv* factors as part of a holistic analysis. *Fintiv* Order at 6. Under that analysis, the factors favor exercising our discretion to deny institution of *inter partes* review: the parallel district-court proceeding has concluded trial, the parties have invested a substantial amount in to the parallel proceeding, the Petition and the issues resolved in the parallel proceeding partly overlap, and for the grounds that arguably do not overlap, the merits are weak. Thus, efficiency and integrity of the system are best served by denying review. *See NHK*, Paper 8 at 20; Consolidated TPG at 58 (quoting 35 U.S.C. § 316(b)).

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[3] Saric mentions a "commercial platform of digital signal processor (DSP)" in the *Background Art* section. Ex. 1005, 2:30–32. But this is only in reference to problems found in the prior art. *See id.* at 1:11–2:32.

III. CONCLUSION

We exercise our discretion under § 314(a) to deny institution.

IV. ORDER

It is

ORDERED that the Petition is *denied*.

PETITIONER:

Colin B. Heideman
Joseph R. Re
Joshua J. Stowell
Jeremy A. Anapol
KNOBBE, MARTENS, OLSON & BEAR, LLP
2cbh@knobbe.com
2JRR@knobbe.com
2jys@knobbe.com
2jaa@knobbe.com
BoxSEAZNL1608LP@knobbe.com

PATENT OWNER:

Vincent J. Rubino, III
Peter Lambrianakos
Enrique W. Iturralde
Jialin Zhong
FABRICANT LLP
vrubino@fabricantllp.com
plambrianakos@fabricantllp.com
eiturralde@fabricantllp.com
zhong@fabricantllp.com

AO 120 (Rev. 08/10)

TO: **Mail Stop 8** **Director of the U.S. Patent and Trademark Office** **P.O. Box 1450** **Alexandria, VA 22313-1450**	**REPORT ON THE FILING OR DETERMINATION OF AN ACTION REGARDING A PATENT OR TRADEMARK**

In Compliance with 35 U.S.C. § 290 and/or 15 U.S.C. § 1116 you are hereby advised that a court action has been filed in the U.S. District Court for the Eastern District of Texas - Marshall Division on the following

☐ Trademarks or ☒ Patents. (☐ the patent action involves 35 U.S.C. § 292.):

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:19-cv-123	April 15, 2019	Eastern District of Texas - Marshall Division

PLAINTIFF	DEFENDANT
VOCALIFE LLC	AMAZON.COM, INC. and AMAZON.COM LLC

PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 8,861,756	October 14, 2014	VOCALIFE LLC
2 RE47,049	September 18, 2018	VOCALIFE LLC
3		
4		
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In the above—entitled case, the following patent(s)/ trademark(s) have been included:

DATE INCLUDED	INCLUDED BY ☐ Amendment ☐ Answer ☐ Cross Bill ☐ Other Pleading	
PATENT OR TRADEMARK NO.	**DATE OF PATENT OR TRADEMARK**	**HOLDER OF PATENT OR TRADEMARK**
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☐ Trademarks or ☐ Patents. (☐ the patent action involves 35 U.S.C. § 292.):

DOCKET NO. 2:21-cv-00128	DATE FILED 4/8/2021	U.S. DISTRICT COURT for the Eastern District of Texas, Marshall Division
PLAINTIFF VOCALIFE LLC		DEFENDANT BOSE CORPORATION

PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 8,861,756	10/14/2014	Vocalife LLC
2 RE47,049	9/18/2021	Vocalife LLC
3 RE48,371	12/29/2020	Vocalife LLC
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DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:21-cv-00129	4/8/2021	for the Eastern District of Texas, Marshall Division

PLAINTIFF	DEFENDANT
VOCALIFE LLC	SONOS, INC.

PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 8,861,756	10/14/2014	Vocalife LLC
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DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:21-cv-00124	4/2/2021	for the Eastern District of Texas, Marshall Division

PLAINTIFF	DEFENDANT
VOCALIFE LLC	GOOGLE LLC

PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 8,861,756	10/14/2014	Vocalife LLC
2 RE47,049	9/18/2021	Vocalife LLC
3 RE48,371	12/29/2020	Vocalife LLC
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PLAINTIFF	DEFENDANT
VOCALIFE LLC	HARMAN INTERNATIONAL INDUSTRIES INC.

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