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
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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	

CONFIRMATION NO. 9714

21125
NUTTER MCCLENNEN & FISH LLP
SEAPORT WEST
155 SEAPORT BOULEVARD
BOSTON, MA 02210-2604

POA ACCEPTANCE LETTER

OC000000132842574

Date Mailed: 03/31/2022

NOTICE OF ACCEPTANCE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 03/30/2022.

The Power of Attorney in this application is accepted. Correspondence in this application will be mailed to the above address as provided by 37 CFR 1.33.

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**.

/nrhayden/

Sony v. Jawbone
U.S. Patent No. 7,246,058
Sony Ex. 1014

UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	JAL-055

CONFIRMATION NO. 9714

15516
Kokka & Backus, PC
P.O. Box 60280
265 Cambridge Avenue
Palo Alto, CA 94306

POWER OF ATTORNEY NOTICE

OC000000132842561

Date Mailed: 03/31/2022

NOTICE REGARDING CHANGE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 03/30/2022.

- The Power of Attorney to you in this application has been revoked by the assignee who has intervened as provided by 37 CFR 3.71. Future correspondence will be mailed to the new address of record(37 CFR 1.33).

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**.

/nrhayden/

DocuSign Envelope ID: C544647C-BDCC-4496-BBED-120D984CBFFC

Doc Code: PA..
Document Description: Power of Attorney

PTO/AIA/82B (07-13)
Approved for use through 03/31/2021. OMB 0651-0035
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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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.)

[x] 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: 21125

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:

[x] The address associated with the above-mentioned Customer Number

OR

[ ] The address associated with Customer Number:

OR

[ ] Firm or Individual Name					
Address					
City		State		Zip	
Country					
Telephone		Email			

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

Jawbone Innovations, LLC

[ ] Inventor or Joint Inventor (title not required below)

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

[x] 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	DocuSigned by: York Eggleston 9708B74C8825434...	Date (Optional)	5/25/2021
Name	York Eggleston		
Title	Manager		

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 1 forms are submitted.

PTO/SB/96 (11-18)
Approved for use through 11/30/2020. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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STATEMENT UNDER 37 CFR 3.73(b)

Applicant/Patent Owner: Gregory C. Burnett

Application No./Patent No.: 7,246,058 Filed/Issue Date: July 17, 2007

Titled: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Jawbone Innovations, LLC , a Limited Liability Company
(Name of Assignee) (Type of Assignee, e.g., corporation, partnership, university, government agency, etc.

states that it is:

1. [x] the assignee of the entire right, title, and interest in;
2. [ ] an assignee of less than the entire right, title, and interest in
(The extent (by percentage) of its ownership interest is _____ %); or
3. [ ] the assignee of an undivided interest in the entirety of (a complete assignment from one of the joint inventors was made) the patent application/patent identified above, by virtue of either:

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 _____, Frame _____, or a copy* is attached.

OR

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

1. From: Gregory C. Burnett To: AliphCom, Inc.
The document was recorded in the United States Patent and Trademark Office at
Reel 013855 , Frame 0906 , or a copy* is attached.

2. From: Gregory C. Burnett To: AliphCom
The document was recorded in the United States Patent and Trademark Office at
Reel 035532 , Frame 0905 , or a copy* is attached.

3. From: AliphCom dba Jawbone To: AliphCom, LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 043637 , Frame 0796 , or a copy* thereof is attached.

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

[x] *As required by 37 CFR 3.73(b)(1)(i), if a copy/copies is/are attached, 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 302.1

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

/Mark S. Leonardo/	March 29, 2022
Signature	Date
Mark S. Leonardo	41,433
Printed or Typed Name	Title or Registration Number

STATEMENT UNDER 37 CFR 3.73(b)

4. From: AliphCom To: AliphCom (assignment for the benefit of creditors), LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 043711 0001 , or for which a copy thereof is attached.

5. From: AliphCom, LLC To: JAWB Acquisition, LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 043638 , Frame 0025 , or for which a copy thereof is attached.

6. From: AliphCom (assignment for the benefit of creditors), LLC To: JAWB Acquisition LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 043746 , Frame 0693 , or for which a copy thereof is attached.

7. From: JAWB Acquisition LLC To: JI Audio Holdings LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 056320 , Frame 0195 , or for which a copy thereof is attached.

8. From: JI Audio Holdings LLC To: Jawbone Innovations, LLC
The document was recorded in the United States Patent and Trademark Office at
Reel 056323 , Frame 0728 , or for which a copy thereof is attached.

9. 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.

Electronic Acknowledgement Receipt	
EFS ID:	45348036
Application Number:	10159770
International Application Number:	
Confirmation Number:	9714
Title of Invention:	DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS
First Named Inventor/Applicant Name:	Gregory C. Burnett
Customer Number:	15516
Filer:	Mark S. Leonardo
Filer Authorized By:	
Attorney Docket Number:	JAL-055
Receipt Date:	30-MAR-2022
Filing Date:	30-MAY-2002
Time Stamp:	08:07:04
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	Power of Attorney	122174_POA_Jawbone_executed.pdf	223726 c3be042904ae588b8cd483853d8861bee83ff9f3	no	1

Warnings:

The PDF file has been signed with a digital signature and the legal effect of the document will be based on the contents of the file not the digital signature.					
Information:					
2	Assignee showing of ownership per 37 CFR 3.73	122174-10101_Statement_3_73.pdf	22536 aedf9cade8b6c691a5fbec0424f7c5f812893be0	no	2
Warnings:					
Information:					
			Total Files Size (in bytes):	246262	

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.

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
2:21-cv-00435	11/29/2021	for the Eastern District of Texas, Marshall Division

PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Amazon.com, Inc. and Amazon.com Services, Inc.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	8,019,091	9/13/2011	Jawbone Innovations, LLC
2	7,246,058	7/17/2007	Jawbone Innovations, LLC
3	8,280,072	10/2/2012	Jawbone Innovations, LLC
4	8,321,213	11/17/2012	Jawbone Innovations, LLC
5	8,326,611	12/4/2012	Jawbone Innovations, LLC

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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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**

AO 120 (Rev. 08/10)

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PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Amazon.com, Inc. and Amazon.com Services, Inc.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	10,779,080	9/15/2020	Jawbone Innovations, LLC
2	11,122,357	9/14/2021	Jawbone Innovations, LLC
3	8,467,543	6/18/2013	Jawbone Innovations, LLC
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In the above—entitled case, the following patent(s)/ trademark(s) have been included:

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AMENDED

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

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:21-cv-00186-JRG	10/26/2021	for the Eastern District of Texas, Marshall Division

PLAINTIFF	DEFENDANT
JAWBONE INNOVATIONS, LLC	SAMSUNG ELECTRONICS CO. LTD. and SAMSUNG ELECTRONICS AMERICA, INC.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	8,019,091	9/13/2011	Jawbone Innovations, LLC
2	8,280,072	10/2/2012	Jawbone Innovations, LLC
3	7,246,058	7/17/2007	Jawbone Innovations, LLC
4	10,779,080	9/15/2020	Jawbone Innovations, LLC
5	11,122,357	9/14/2021	Jawbone Innovations, LLC

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

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DECISION/JUDGEMENT

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Copy 2—Upon filing document adding patent(s), mail this copy to Director Copy 4—Case file copy

AMENDED

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DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
2:21-cv-00186-JRG	10/26/2021	for the Eastern District of Texas, Marshall Division

PLAINTIFF	DEFENDANT
JAWBONE INNOVATIONS, LLC	SAMSUNG ELECTRONICS CO. LTD. and SAMSUNG ELECTRONICS AMERICA, INC.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	8,467,543	6/18/2013	Jawbone Innovations, LLC
2	8,503,691	8/6/2013	Jawbone Innovations, LLC
3			
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In the above—entitled case, the following patent(s)/ trademark(s) have been included:

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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 Western District of Texas - Waco Division on the following

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

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
6:21-cv-00984	September 23, 2021	Western District of Texas - Waco Division

PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Apple Inc.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	8,019,091	2/8/2005	Jawbone Innovations, LLC
2	7,246,058	7/17/2007	Jawbone Innovations, LLC
3	8,280,072	10/2/2012	Jawbone Innovations, LLC
4	8,321,213	11/27/2012	Jawbone Innovations, LLC
5	8,326,611	12/4/2012	Jawbone Innovations, LLC

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

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DECISION/JUDGEMENT

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Copy 2—Upon filing document adding patent(s), mail this copy to Director **Copy 4—Case file copy**

AO 120 (Rev. 08/10)

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

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
6:21-cv-00984	September 23, 2021	Western District of Texas - Waco Division

PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Apple Inc.

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	10,799,080	9/15/2020	Jawbone Innovations, LLC
2	11,122,357	9/14/2021	Jawbone Innovations, LLC
3	8,467,543	6/18/2014	Jawbone Innovations, LLC
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☐ Trademarks or ☒ Patents. (☐ the patent action involves 35 U.S.C. § 292.):

DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
6:21-cv-00985	September 23, 2021	Western District of Texas - Waco Division

PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Google LLC

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	8,019,091	2/8/2005	Jawbone Innovations, LLC
2	7,246,058	7/17/2007	Jawbone Innovations, LLC
3	8,280,072	10/2/2012	Jawbone Innovations, LLC
4	8,321,213	11/27/2012	Jawbone Innovations, LLC
5	8,326,611	12/4/2012	Jawbone Innovations, LLC

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

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DOCKET NO.	DATE FILED	U.S. DISTRICT COURT
6:21-cv-00985	September 23, 2021	Western District of Texas - Waco Division

PLAINTIFF	DEFENDANT
Jawbone Innovations, LLC	Google LLC

	PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1	10,799,080	9/15/2020	Jawbone Innovations, LLC
2	11,122,357	9/14/2021	Jawbone Innovations, LLC
3	8,467,543	6/18/2014	Jawbone Innovations, LLC
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In the above—entitled case, the following patent(s)/ trademark(s) have been included:

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PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
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Copy 2—Upon filing document adding patent(s), mail this copy to Director **Copy 4—Case file copy**

JAWBONE®

JAWB Acquisition LLC

321 West 44th Street, Suite 1000

New York, NY 10036

Hello,

This letter is to assert that JAWB Acquisition LLC is a small entity with four employees. We are the current assignee and would like to submit a payment for the patent below as a small entity:

PATENT #
7246058
APPLICATION #
10159770
FILING DATE
05/30/2002
ISSUE DATE
07/17/2007

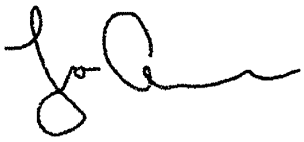
Please contact Luke Conticello or Daniel Setton for additional information for JAWB Acquisition LLC.

Thanks,

JAWB Acquisition LLC

631 495 2954

Patents@aliph.com



Luke Conticello, Director

PATENT ASSIGNMENT COVER SHEET

Electronic Version v1.1
Stylesheet Version v1.2

EPAS ID: PAT3502102

SUBMISSION TYPE:	NEW ASSIGNMENT
NATURE OF CONVEYANCE:	SECURITY INTEREST

CONVEYING PARTY DATA

Name	Execution Date
ALIPHCOM	08/26/2015
MACGYVER ACQUISITION LLC	08/26/2015
ALIPH, INC.	08/26/2015
BODYMEDIA, INC.	08/26/2015
PROJECT PARIS ACQUISITION LLC	08/26/2015

RECEIVING PARTY DATA

Name:	BLACKROCK ADVISORS, LLC
Street Address:	1 UNIVERSITY SQUARE DRIVE
Internal Address:	C/O GLOBAL ALLOCATION GROUP
City:	PRINCETON
State/Country:	NEW JERSEY
Postal Code:	08540

PROPERTY NUMBERS Total: 604

Property Type	Number
Application Number:	10159770
Application Number:	13069244
Application Number:	13069264
Application Number:	13069275
Application Number:	13753441
Application Number:	29333427
Application Number:	29333428
Application Number:	29333430
Application Number:	29333431
Application Number:	29333432
Application Number:	10400292
Application Number:	10667207
Application Number:	10769302
Application Number:	11199856
Application Number:	11704552

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Application Number:	11859460
Application Number:	11860004
Application Number:	11969737
Application Number:	11982956
Application Number:	12006607
Application Number:	12039718
Application Number:	12123364
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Application Number:	29516015
Application Number:	29516016
Application Number:	29516017
Application Number:	29517054
Application Number:	29517056

CORRESPONDENCE DATA

Fax Number: (617)235-9802

Correspondence will be sent to the e-mail address first; if that is unsuccessful, it will be sent using a fax number, if provided; if that is unsuccessful, it will be sent via US Mail.

Phone: 6179517860
Email: ronald.duvernay@ropesgray.com
Correspondent Name: JOSHUA S. JACKSON
Address Line 1: PRUDENTIAL TOWER, 800 BOYLSTON STREET
Address Line 2: ROPES & GRAY LLP
Address Line 4: BOSTON, MASSACHUSETTS 02199-3600

ATTORNEY DOCKET NUMBER:	MFOG-652
NAME OF SUBMITTER:	JOSHUA S. JACKSON
SIGNATURE:	/j jackson/
DATE SIGNED:	08/27/2015
	This document serves as an Oath/Declaration (37 CFR 1.63).

Total Attachments: 32
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AMENDED AND RESTATED PATENT SECURITY AGREEMENT

THIS AMENDED AND RESTATED PATENT SECURITY AGREEMENT (as it may be amended, restated or otherwise modified from time to time, the "Patent Security Agreement") is entered into as of August 26, 2015 by and among (i) AliphCom, a California corporation (the "Company"), MacGyver Acquisition LLC, a Delaware limited liability company, Aliph, Inc., a Delaware Corporation, BodyMedia, Inc., a Delaware corporation, and Project Paris Acquisition LLC, a Delaware limited liability company (each, a "Grantor", and collectively, the "Grantors"), and (ii) BlackRock Advisors, LLC, a Delaware limited liability company (the "Agent").

W I T N E S S E T H:

WHEREAS, each of the Grantors, the Purchasers identified therein and the Agent are entering into the Note Purchase Agreement, dated as of April 28, 2015, as amended by the First Amendment to Note Purchase Agreement; Omnibus Amendment to Secured Convertible Promissory Notes and Limited Consent, dated as of July 22, 2015 (as amended, restated, supplemented, or otherwise modified from time to time, the "Existing Note Purchase Agreement"), pursuant to which the Company issued certain secured convertible promissory notes (the "Notes") and the other Grantors agreed to Guarantee the Obligations;

WHEREAS, each of the Grantors and the Agent entered into the original Security Agreement, dated as of April 28, 2015 (as amended, restated, supplemented, or otherwise modified from time to time, the "Original Security Agreement"), in order to induce the Purchasers to purchase the Notes and to secure the Secured Obligations; and

WHEREAS, pursuant to the Original Security Agreement, each of the Grantors is required to execute and deliver to Agent the original Patent Security Agreement, dated as of April 28, 2015 (the "Original Patent Security Agreement");

WHEREAS, each of the Grantors, the Purchasers identified therein and the Agent entered into the Note Purchase Agreement, dated as of July 21, 2015 (as amended, restated, supplemented, or otherwise modified from time to time, the "July 2015 Note Purchase Agreement"), pursuant to which the Company issued certain additional Notes and the other Grantors agreed to Guarantee the Secured Obligations;

WHEREAS, each of the Grantors and the Agent entered into the Amended and Restated Security Agreement, dated as of July 22, 2015 (as amended, restated, supplemented, or otherwise modified from time to time, the "Security Agreement"), in order to induce the Purchasers to purchase the additional Notes and to secure the Secured Obligations; and

WHEREAS, the parties to the Original Patent Security Agreement wish to amend and restate the Original Patent Security Agreement in its entirety;

NOW, THEREFORE, in consideration of the premises and mutual covenants herein contained and for other good and valuable consideration, the receipt and sufficiency of which are hereby acknowledged, each Grantor hereby agrees as follows:

1. DEFINED TERMS. All initially capitalized terms used but not otherwise defined herein have the meanings given to them in the Security Agreement, and this Patent Security Agreement shall be subject to the rules of construction set forth in the Security Agreement, which rules of construction are incorporated herein by this reference, *mutatis mutandis*.

2. GRANT OF SECURITY INTEREST IN PATENT COLLATERAL. Each Grantor does hereby unconditionally grant, assign, and pledge to Agent, and agrees to unconditionally grant, assign, and pledge to Agent, for its benefit and the benefit of each of the Noteholders, to secure the Secured Obligations, as applicable, a continuing security interest (referred to in this Patent Security Agreement as the "Security Interest") in all of such Grantor's entire right, title and interest in and to the following, whether now owned or hereafter acquired or arising (collectively, the "Patent Collateral"):

(a) all of such Grantor's Patents and Patent Intellectual Property Licenses to which it is a party including those referred to on Schedule I;

(b) all applications, improvements, divisionals, continuations, continuations-in-part, reissues, reexaminations, or extensions of the foregoing, foreign counterparts, and the inventions covered thereby;

(c) all files and records relating to the prosecution, exploitation, and defense of any of the foregoing, and all rights of action pertaining to the Grantor's Patents; and

(d) all products and proceeds of the foregoing, including any claim by such Grantor against third parties for past, present or future infringement of any Patent or any Patent exclusively licensed under any Intellectual Property License, including the right to receive damages, or right to receive license fees, royalties, and other compensation under any Patent Intellectual Property License.

3. SECURITY FOR SECURED OBLIGATIONS. This Patent Security Agreement and the Security Interest created hereby secures the payment and performance of the Secured Obligations, whether now existing or arising hereafter. Without limiting the generality of the foregoing, this Patent Security Agreement secures the payment of all amounts which constitute part of the Secured Obligations and would be owed by Grantors, or any of them, to Agent, or the Noteholders, whether or not they are unenforceable or not allowable due to the existence of an Insolvency Proceeding involving any Grantor.

4. SECURITY AGREEMENT. The Security Interest granted pursuant to this Patent Security Agreement is granted in conjunction with the security interests granted to Agent, for its benefit and the benefit of the Noteholders, pursuant to the Security Agreement. Each Grantor hereby acknowledges and affirms that the rights and remedies of Agent with respect to the Security Interest in the Patent Collateral made and granted hereby are more fully set forth in the Security Agreement, the terms and provisions of which are incorporated by reference herein as if fully set forth herein. To the extent there is any inconsistency between this Patent Security Agreement and the Security Agreement, the Security Agreement shall control.

5. AUTHORIZATION TO SUPPLEMENT. If any Grantor shall obtain rights to any new patent application or issued patent or become entitled to the benefit of any patent application or patent for any improvement, divisional, continuation, continuation-in-part, reissue, or reexamination of any existing patent or patent application, or any inventions covered thereby, the provisions of this Patent Security Agreement shall automatically apply thereto. Grantors hereby authorize Agent unilaterally to modify this Patent Security Agreement by amending Schedule I to include any such new patent rights of each Grantor. Notwithstanding the foregoing, no failure to so modify this Patent Security Agreement or amend Schedule I shall in any way affect, invalidate or detract from Agent's continuing security interest in all Collateral, whether or not listed on Schedule I.

6. COUNTERPARTS. This Patent Security Agreement is a Note Document. This Patent Security Agreement may be executed in any number of counterparts and by different parties on separate counterparts, each of which, when executed and delivered, shall be deemed to be an original, and all of which, when taken together, shall constitute but one and the same Patent Security Agreement. Delivery

of an executed counterpart of this Patent Security Agreement by telefacsimile or other electronic method of transmission shall be equally as effective as delivery of an original executed counterpart of this Patent Security Agreement. Any party delivering an executed counterpart of this Patent Security Agreement by telefacsimile or other electronic method of transmission also shall deliver an original executed counterpart of this Patent Security Agreement but the failure to deliver an original executed counterpart shall not affect the validity, enforceability, and binding effect of this Patent Security Agreement.

7. CHOICE OF LAW AND VENUE, JURY TRIAL WAIVER. THIS PATENT SECURITY AGREEMENT SHALL BE SUBJECT TO THE PROVISIONS REGARDING CHOICE OF LAW AND VENUE AND JURY TRIAL WAIVER SET FORTH IN SECTIONS 8.16, 8.17 AND 8.18 OF THE SECURITY AGREEMENT, AND SUCH PROVISIONS ARE INCORPORATED HEREIN BY THIS REFERENCE, *MUTATIS MUTANDIS*.

8. ORIGINAL PATENT SECURITY AGREEMENT AMENDED AND RESTATED. This Patent Security Agreement is an amendment and restatement of the Original Patent Security Agreement. This Patent Security Agreement is in no way intended to constitute a novation of the Original Patent Security Agreement or the "Obligations" (as defined in the Existing Note Purchase Agreement and the July 2015 Note Purchase Agreement). Any security granted pursuant to or in connection with the Original Patent Security Agreement and the other documents executed in connection therewith shall continue to secure the Secured Obligations.

[signature page follows]

IN WITNESS WHEREOF, the parties hereto have caused this Patent Security Agreement to be executed and delivered as of the day and year first above written.

GRANTORS:

ALIPHCOM,
a California corporation

By: DocuSigned by: Hosain Rahman 0E4D8FA187D24C0...
Name: Hosain Rahman
Title: Chief Executive Officer

ALIPH, INC.,
a Delaware corporation

By: DocuSigned by: Hosain Rahman 0E4D8FA187D24C0...
Name: Hosain Rahman
Title: Chief Executive Officer

MACGYVER ACQUISITION, LLC,
a Delaware limited liability company
by ALIPHCOM, as sole member

By: DocuSigned by: Hosain Rahman 0E4D8FA187D24C0...
Name: Hosain Rahman
Title: Chief Executive Officer

BODYMEDIA, INC.,
a Delaware corporation

By: DocuSigned by: Hosain Rahman 0E4D8FA187D24C0...
Name: Hosain Rahman
Title: Chief Executive Officer

PROJECT PARIS ACQUISITION LLC, a
Delaware limited liability company
by ALIPHCOM, as sole member

By: DocuSigned by: Hosain Rahman 0E4D8FA187D24C0...
Name: Hosain Rahman
Title: Chief Executive Officer

[SIGNATURE PAGE TO AMENDED AND RESTATED PATENT SECURITY AGREEMENT]

AGENT:

ACCEPTED AND ACKNOWLEDGED BY:

BLACKROCK ADVISORS, LLC

By: [signature]
Name: Lisa O'Donnell
Title: Managing Director, Authorized Signatory

[SIGNATURE PAGE TO AMENDED AND RESTATED PATENT SECURITY AGREEMENT]

SCHEDULE I
to
PATENT SECURITY AGREEMENT

Patents

Current Owner	Country	Patent No.	Application No.	Filed
Aliph, Inc.	U.S.	7246058B2	10159770	5/30/2002
Aliph, Inc.	U.S.		13069244	3/22/2011
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Aliph, Inc.	U.S.	8,942,383	13753441	1/29/2013
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Aliphcom, Inc.	U.S.	8467543B2	10400282	3/27/2003
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AliphCom, Inc.	U.S.	8055307	12039718	2/18/2008
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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009

CONFIRMATION NO. 9714

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POWER OF ATTORNEY NOTICE

OC000000051334153

Date Mailed: 12/12/2011

NOTICE REGARDING CHANGE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 11/30/2011.

- The Power of Attorney to you in this application has been revoked by the assignee who has intervened as provided by 37 CFR 3.71. Future correspondence will be mailed to the new address of record(37 CFR 1.33).

/atesfai/

Office of Data Management, Application Assistance Unit (571) 272-4000, or (571) 272-4200, or 1-888-786-0101

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009

CONFIRMATION NO. 9714

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POA ACCEPTANCE LETTER

OC000000051334174

Date Mailed: 12/12/2011

NOTICE OF ACCEPTANCE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 11/30/2011.

The Power of Attorney in this application is accepted. Correspondence in this application will be mailed to the above address as provided by 37 CFR 1.33.

/atesfai/

Office of Data Management, Application Assistance Unit (571) 272-4000, or (571) 272-4200, or 1-888-786-0101

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POWER OF ATTORNEY TO PROSECUTE APPLICATIONS BEFORE THE USPTO

I hereby revoke all previous powers of attorney given in the application identified in the attached statement under 37 CFR 3.73(b).

I hereby appoint:

☑ Practitioners associated with the Customer Number: 15516

OR

☐ Practitioner(s) named below (If more than ten patent practitioners are to be named, then a customer number must be used):

Name	Registration Number	Name	Registration Number

as attorney(s) or agent(s) to represent the undersigned before the United States Patent and Trademark Office (USPTO) in connection with any and all patent applications assigned only to the undersigned according to the USPTO assignment records or assignment documents attached to this form in accordance with 37 CFR 3.73(b).

Please change the correspondence address for the application identified in the attached statement under 37 CFR 3.73(b) to:

☑ The address associated with Customer Number: 15516

OR

☐ Firm or Individual Name			
Address			
City		State	Zip
Country			
Telephone		Email	

Assignee Name and Address:

AlphCom, Inc
410 Jessie Street
San Francisco, California 94103

A copy of this form, together with a statement under 37 CFR 3.73(b) (Form PTO/SB/96 or equivalent) is required to be filed in each application in which this form is used. The statement under 37 CFR 3.73(b) may be completed by one of the practitioners appointed in this form if the appointed practitioner is authorized to act on behalf of the assignee, and must identify the application in which this Power of Attorney is to be filed.

SIGNATURE of Assignee of Record

The individual whose signature and title is supplied below is authorized to act on behalf of the assignee

Signature		Date	12-2-2011
Name	Michael Edward Smith Luna	Telephone	(877) 254-7426
Title	Chief Technology Officer, AlphCom, Inc		

This collection of information is required by 37 CFR 1.31, 1.32 and 1.33. 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 1.11 and 1.14. This collection is estimated to take 3 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: **Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

PTO/SB/96 (07-09)
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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.

STATEMENT UNDER 37 CFR 3.73(b)

Applicant/Patent Owner: Gregory C. Burnett

Application No./Patent No.: 10/159,770 / 7,246,058 Filed/Issue Date: 05/30/2002; 07/17/2007

Titled: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

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

states that it is:

1. [X] the assignee of the entire right, title, and interest in;

2. [ ] an assignee of less than the entire right, title, and interest in
(The extent (by percentage) of its ownership interest is _____ %); or

3. [ ] the assignee of an undivided interest in the entirety of (a complete assignment from one of the joint inventors was made)

the patent application/patent identified above, by virtue of either:

A. [X] 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 013855, Frame 0906, or for which a copy therefore is attached.

OR

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: _____

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

3. From: _____ To: _____

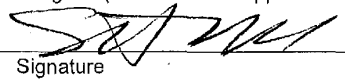
The document was recorded in the United States Patent and Trademark Office at
Reel _____, Frame _____, or for which a copy thereof is attached.

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

[X] As required by 37 CFR 3.73(b)(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.



Signature December 2, 2011
Date

Scott S. Kokka Attorney of Record
Printed or Typed Name Title

This collection of information is required by 37 CFR 3.73(b). 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 1.11 and 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

Electronic Acknowledgement Receipt

EFS ID:	11537223
Application Number:	10159770
International Application Number:	
Confirmation Number:	9714
Title of Invention:	DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS
First Named Inventor/Applicant Name:	Gregory C. Burnett
Customer Number:	98195
Filer:	Scott Susumu Kokka
Filer Authorized By:	
Attorney Docket Number:	ALPH.P009
Receipt Date:	02-DEC-2011
Filing Date:	30-MAY-2002
Time Stamp:	21:25:32
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		ALI-055_GPOA_and_stmt_37CFR373b_asfiled.pdf	385207 5ad7a2bbcd8e5cf828594c1ef8d43cd181f5154b	yes	2

	Multipart Description/PDF files in .zip description		
	Document Description	**Start**	**End**
	Power of Attorney	1	1
	Assignee showing of ownership per 37 CFR 3.73(b).	2	2
Warnings:			
Information:			
	Total Files Size (in bytes):	385207	

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.

PTO/SB/80 (11-08)
Approved for use through 11/30/2011. OMB 0651-0035
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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POWER OF ATTORNEY TO PROSECUTE APPLICATIONS BEFORE THE USPTO

I hereby revoke all previous powers of attorney given in the application identified in the attached statement under 37 CFR 3.73(b).

I hereby appoint:

☑ Practitioners associated with the Customer Number: 15516

OR

☐ Practitioner(s) named below (if more than ten patent practitioners are to be named, then a customer number must be used):

Name	Registration Number	Name	Registration Number

as attorney(s) or agent(s) to represent the undersigned before the United States Patent and Trademark Office (USPTO) in connection with any and all patent applications assigned only to the undersigned according to the USPTO assignment records or assignment documents attached to this form in accordance with 37 CFR 3.73(b).

Please change the correspondence address for the application identified in the attached statement under 37 CFR 3.73(b) to:

☑ The address associated with Customer Number: 15516

OR

☐ Firm or Individual Name			
Address			
City		State	Zip
Country			
Telephone		Email	

Assignee Name and Address:

AliphCom, Inc.
410 Jessie Street, Unit 601
San Francisco, CA 94103

A copy of this form, together with a statement under 37 CFR 3.73(b) (Form PTO/SB/96 or equivalent) is required to be filed in each application in which this form is used. The statement under 37 CFR 3.73(b) may be completed by one of the practitioners appointed in this form if the appointed practitioner is authorized to act on behalf of the assignee, and must identify the application in which this Power of Attorney is to be filed.

SIGNATURE of Assignee of Record
The individual whose signature and title is supplied below is authorized to act on behalf of the assignee

Signature		Date	11-21-2011
Name	Michael Edward Smith Luna	Telephone	(877) 254-7426
Title	Chief Technology Officer, AliphCom, Inc.		

This collection of information is required by 37 CFR 1.31, 1.32 and 1.33. 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 1.11 and 1.14. This collection is estimated to take 3 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

PTO/SB/96 (07-09)
Approved for use through 07/31/2012. OMB 0651-0031
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.

STATEMENT UNDER 37 CFR 3.73(b)

Applicant/Patent Owner: Gregory C. Burnett

Application No./Patent No.: 10/159,770/7,246,058 Filed/Issue Date: 05/30/2002; 07/17/2007

Titled: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

AliphCom, Inc. , a Corporation

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

states that it is:

1. [X] the assignee of the entire right, title, and interest in;

2. [ ] an assignee of less than the entire right, title, and interest in
(The extent (by percentage) of its ownership interest is _____ %); or

3. [ ] the assignee of an undivided interest in the entirety of (a complete assignment from one of the joint inventors was made)

the patent application/patent identified above, by virtue of either:

A. [X] 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 013855, Frame 0906, or for which a copy therefore is attached.

OR

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: _____

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

3. From: _____ To: _____

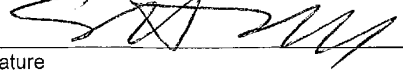
The document was recorded in the United States Patent and Trademark Office at
Reel _____, Frame _____, or for which a copy thereof is attached.

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

[X] As required by 37 CFR 3.73(b)(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.



Signature

November 30, 2011
Date

Scott S. Kokka
Printed or Typed Name

Attorney of Record
Title

This collection of information is required by 37 CFR 3.73(b). 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 1.11 and 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 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.
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 Acknowledgement Receipt	
EFS ID:	11516248
Application Number:	10159770
International Application Number:	
Confirmation Number:	9714
Title of Invention:	DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS
First Named Inventor/Applicant Name:	Gregory C. Burnett
Customer Number:	98195
Filer:	Scott Susumu Kokka
Filer Authorized By:	
Attorney Docket Number:	ALPH.P009
Receipt Date:	30-NOV-2011
Filing Date:	30-MAY-2002
Time Stamp:	23:34:48
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		ALI-055_GPOA_and_Stmt_37CFR373b_asfiled.pdf	671858 b5b44595031b6b76c63d3accd7c230fe87c2d37b	yes	3

	Multipart Description/PDF files in .zip description		
	Document Description	**Start**	**End**
	Power of Attorney	1	1
	Assignee showing of ownership per 37 CFR 3.73(b).	2	3
Warnings:			
Information:			
	Total Files Size (in bytes):	671858	

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.

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

APPLICATION NUMBER	PATENT NUMBER	GROUP ART UNIT	FILE WRAPPER LOCATION
10/159,770	7246058	2626	9200



Correspondence Address/Fee Address Change

The following fields have been set to Customer Number 98195 on 08/09/2010

- **Correspondence Address**
- **Maintenance Fee Address**
- **Power of Attorney Address**

The address of record for Customer Number 98195 is:

98195
Gregory & Martensen LLP
2018 Bissonnet Street
Houston, TX 77005

PART 1 - ATTORNEY/APPLICANT COPY
page 1 of 1

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

APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	07/17/2007	7246058	ALPH.P009	9714

53186 7590 06/27/2007

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

ISSUE NOTIFICATION

The projected patent number and issue date are specified above.

Determination of Patent Term Adjustment under 35 U.S.C. 154 (b)
(application filed on or after May 29, 2000)

The Patent Term Adjustment is 688 day(s). Any patent to issue from the above-identified application will include an indication of the adjustment on the front page.

If a Continued Prosecution Application (CPA) was filed in the above-identified application, the filing date that determines Patent Term Adjustment is the filing date of the most recent CPA.

Applicant will be able to obtain more detailed information by accessing the Patent Application Information Retrieval (PAIR) WEB site (http://pair.uspto.gov).

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 (571)-272-4200.

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

Gregory C. Burnett, Livermore, CA;

OIPE JC14
NOV 21 2002
PATENT & TRADEMARK OFFICE

Please type a plus sign (+) inside this box → [+]

PTO/SB/08A (08-00)
Approved for use through 10/31/2002. OMB 0651-0031
U.S. Patent and Trademark Office: U.S. DEPARTMENT OF COMMERCE
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Substitute for form 1449A/PTO

INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(use as many sheets as necessary)

Sheet 1 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	~~2644~~ 2654
Examiner Name	~~Unknown~~ ABAD
Attorney Docket Number	ALPH.P009

U.S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	U.S. Patent Document Number	Kind Code[2] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
AA		5,212,764		Ariyoshi	05-18-1993	
		5,400,409		Linhard	03-21-1995	
		5,414,776		Sims, Jr.	05-09-1995	
		5,539,859		Robbe et al.	07-23-1996	
		5,633,935		Kanamori et al.	05-27-1997	
		5,835,608		Wamaka et al.	11-10-1998	
		5,917,921		Sasaki et al.	06-29-1999	
		~~6,000,396~~		Nagata	12-28-1999	
		6,191,724	B1	McEwan	02-20-2001	
		5,986,600		McEwan	11-16-1999	
		5,966,090		McEwan	10-12-1999	
AA		6,006,175		Holzrichter	12-21-1999	

6,009,396

RECEIVED
NOV 2 2 2002
Technology Center 2600

FOREIGN PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Foreign Patent Document Office[3]	Number[4]	Kind Code[5] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear	T[6]

Examiner Signature		Date Considered	12/10/05

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

[1] Unique citation designation number. [2] See attached Kinds of U.S. Patent Documents. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST. 16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

Burden Hour Statement: This form is estimated to take 2.0 hours to complete. Time will vary depending upon the needs of the individual case. Any comments on the amount of time you are required to complete this form should be sent to the Chief Information Officer, U. S. Patent and Trademark Office, Washington, DC 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Assistant Commissioner for Patents, Washington, DC 20231.

OIPE JC14B
NOV 05 2004
PATENT & TRADEMARK OFFICE

PTO/SB/08A (08-03)
Approved for use through 07/31/2006. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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Substitute for form 1449/PTO

INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(Use as many sheets as necessary)

Sheet 1 of 1

Complete if Known	
Application Number	10/159,770
Filing Date	05/30/2002
First Named Inventor	Gregory C. Burnett
Art Unit	~~2644~~ 2654
Examiner Name	~~Not yet assigned~~ A ZAD
Attorney Docket Number	ALPH.P009

U. S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Document Number Number-Kind Code[2] (If known)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
A. A		US- 2002/039425 A1	04-04-2002	Burnett et al.	
		US- 3,789,166	01-29-1974	Sebesta	
		US- 4,006,318	02-01-1977	Sebesta et al.	
		US- 4,591,668	05-27-1986	Iwata	
		US- 4,901,354	02-13-1990	Gollmar et al.	
		US- 5,473,702	12-05-1995	Yoshida et al.	
		US- 5,515,865	05-14-1996	Scanlon et al.	
		US- 5,649,055	07-15-1997	Gupta et al.	
		US- 5,684,460	11-04-1997	Scanlon et al.	
		US- 5,729,694	03-17-1998	Holzrichter et al.	
		US- 5,754,665	05-19-1998	Hosoi et al.	
		US- 5,853,005	12-29-1998	Scanlon	
		US- 6,069,963	05-30-2000	Martin et al.	
		US- 6,191,724	02-20-2001	McEwan	
		US- 6,266,422	07-24-2001	Ikeda	
		US- 6,430,295	08-06-2002	Handel et al.	
A. A		US- 5,097,515 3-17-1992 ~~03-17-2002~~		Baba	
		US-			
		US			

WD 6/7/07

FOREIGN PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Foreign Patent Document Country Code[3]-Number[4]-Kind Code[5] (If known)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages Or Relevant Figures Appear	T[6]
A. A		EP 0 637 187 A	02-01-1995	Pan Communicati		
A. A		EP 0 795 851 A2	09-17-1997	Tokyo Shibaura		
A. A		EP 0 984 660 A2	03-08-2000	Nippon T&T Corp		
A. A		JP 2000 312 395	11-07-2000	Alpine Electron		
A. A		JP 2001 189 987	07-10-2001	Pioneer Electro		
A. A		WO 02 07151	01-24-2002	ALIPHCOM, Inc.		

Examiner Signature	A. A.	Date Considered	12/10/05

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant. [1] Applicant's unique citation designation number (optional). [2] See Kinds Codes of USPTO Patent Documents at www.uspto.gov or MPEP 901.04. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

This collection of information is required by 37 CFR 1.97 and 1.98. 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 2 hours 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, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.

If you need assistance in completing the form, call 1-800-PTO-9199 (1-800-786-9199) and select option 2.

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

MAY 25 2007

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)

53186 7590 05/22/2007

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

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.

Barbara B. Courtney (Depositor's name)
(Signature)
May 25/2007 (Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

TITLE OF INVENTION: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	NO	$1400	$300	$0	$1700	08/22/2007

EXAMINER	ART UNIT	CLASS-SUBCLASS
AZAD, ABUL K	2626	704-214000

05/29/2007 TDESHAM2 00000036 10159770

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 Courtney Staniford & Gregory LLP
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: Aliph, Inc.
(B) RESIDENCE: (CITY and STATE OR COUNTRY): San Francisco, CA

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 credit any overpayment, to Deposit Account Number 503616 (enclose an extra copy of this form).

5. **Change in Entity Status** (from status indicated above)
☐ a. Applicant claims SMALL ENTITY status. See 37 CFR 1.27.
☐ b. Applicant is no longer claiming SMALL ENTITY status. See 37 CFR 1.27(g)(2).

NOTE: The Issue Fee and Publication Fee (if required) will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the United States Patent and Trademark Office.

Authorized Signature _____ Date May 25, 2007
Typed or printed name Barbara B. Courtney Registration No. 42,442

This collection of information 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.

PAGE 2/3 * RCVD AT 5/25/2007 4:51:53 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2732885 * CSID:14083421909 * DURATION (mm-ss):01-10

OIPE IAP78 MAY 25 2007 PATENT & TRADEMARK OFFICE

Attorney Docket No. ALPH.P009 Patent

Transmittal of Issue Fee

Certification Under 37 C.F.R. §1.8(a)

Transmitted to USPTO Central Facsimile No.: **571-273-2885**

May 25, 2007 Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the Facsimile number above on the date indicated above.

Barbara B. Courtney (Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:
Gregory C. Burnett
Application No. 10/159,770
Filed: May 30, 2002
For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Examiner: Abul K. Azad
Art Unit: 2626

3 pages total

VIA FACSIMILE: 571-273-2885
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

In response to the Notice of Allowance dated May 22, 2007, Applicant herewith submits the following:

1. Form PTOL-85B;
2. Authorization to charge our credit card in the amount of $1,700.00 to cover the following:
   a. Issue Fee Large Entity $1400.00; and
   b. Publication Fee $300.00.

Courtney Staniford & Gregory LLP
PO Box 9686
San Jose, CA 95157
Tel: 408-342-1902
Fax: 408-342-1909

PAGE 1/3 * RCVD AT 5/25/2007 4:51:53 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2732885 * CSID:14083421909 * DURATION (mm-ss):01-10

UNITED STATES PATENT AND TRADEMARK OFFICE

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

NOTICE OF ALLOWANCE AND FEE(S) DUE

53186 7590 05/22/2007

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

EXAMINER	
AZAD, ABUL K	
ART UNIT	PAPER NUMBER
2626	

DATE MAILED: 05/22/2007

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

TITLE OF INVENTION: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	NO	$1400	$300	$0	$1700	08/22/2007

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 SMALL ENTITY status shown above.

If the SMALL ENTITY is shown as YES, verify your current SMALL ENTITY status:

A. If the status is the same, pay the TOTAL FEE(S) DUE shown above.

B. If the status above is to be removed, check box 5b on Part B - Fee(s) Transmittal and pay the PUBLICATION FEE (if required) and twice the amount of the ISSUE FEE shown above, or

If the SMALL ENTITY is shown as NO:

A. Pay TOTAL FEE(S) DUE shown above, or

B. If applicant claimed SMALL ENTITY status before, or is now claiming SMALL ENTITY status, check box 5a on Part B - Fee(s) Transmittal and pay the PUBLICATION FEE (if required) and 1/2 the ISSUE FEE shown above.

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

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

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

PTOL-85 (Rev. 07/06) Approved for use through 05/31/2007.

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)

53186 7590 05/22/2007

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

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.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

TITLE OF INVENTION: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	NO	$1400	$300	$0	$1700	08/22/2007

EXAMINER	ART UNIT	CLASS-SUBCLASS
AZAD, ABUL K	2626	704-214000

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 credit any overpayment, to Deposit Account Number _____ (enclose an extra copy of this form).

5. **Change in Entity Status** (from status indicated above)
☐ a. Applicant claims SMALL ENTITY status. See 37 CFR 1.27. ☐ b. Applicant is no longer claiming SMALL ENTITY status. See 37 CFR 1.27(g)(2).

NOTE: The Issue Fee and Publication Fee (if required) will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the United States Patent and Trademark Office.

Authorized Signature _____ Date _____

Typed or printed name _____ Registration No. _____

This collection of information 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.

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

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

53186 7590 05/22/2007

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

EXAMINER	
AZAD, ABUL K	
ART UNIT	PAPER NUMBER
2626	

DATE MAILED: 05/22/2007

Determination of Patent Term Adjustment under 35 U.S.C. 154 (b)
(application filed on or after May 29, 2000)

The Patent Term Adjustment to date is 688 day(s). If the issue fee is paid on the date that is three months after the mailing date of this notice and the patent issues on the Tuesday before the date that is 28 weeks (six and a half months) after the mailing date of this notice, the Patent Term Adjustment will be 688 day(s).

If a Continued Prosecution Application (CPA) was filed in the above-identified application, the filing date that determines Patent Term Adjustment is the filing date of the most recent CPA.

Applicant will be able to obtain more detailed information by accessing the Patent Application Information Retrieval (PAIR) WEB site (http://pair.uspto.gov).

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.

PTOL-85 (Rev. 07/06) Approved for use through 05/31/2007.

Notice of Allowability	Application No.	Applicant(s)	
	10/159,770	BURNETT, GREGORY C.	
	Examiner	Art Unit	
	ABUL K. AZAD	2626	

-- 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 *the communication filed on April 9, 2007*.

2. ☒ The allowed claim(s) is/are *6-10*.

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

4. ☐ A SUBSTITUTE OATH OR DECLARATION must be submitted. Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL PATENT APPLICATION (PTO-152) which gives reason(s) why the oath or declaration is deficient.

5. ☐ CORRECTED DRAWINGS (as "replacement sheets") must be submitted.
   (a) ☐ including changes required by the Notice of Draftsperson's Patent Drawing Review (PTO-948) attached
      1) ☐ hereto or 2) ☐ to Paper No./Mail Date _____.
   (b) ☐ 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. ☐ Notice of Draftperson's Patent Drawing Review (PTO-948)
3. ☐ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date _____
4. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
5. ☐ Notice of Informal Patent Application
6. ☐ Interview Summary (PTO-413), Paper No./Mail Date _____ .
7. ☐ Examiner's Amendment/Comment
8. ☐ Examiner's Statement of Reasons for Allowance
9. ☐ Other _____.

4/28/07

ABUL K. AZAD
Primary Examiner
Art Unit: 2626

Attorney Docket No.: 37212.8009.US00

SN: 10/159770

INVENTORSHIP DECLARATION BY SOLE INVENTOR

I HEREBY DECLARE THAT:

My residence, mailing address, and citizenship are stated next to my name in PART A on hereof.

I believe I am the original, first, and sole inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

the specification of which:

☒ is attached hereto.

☐ was filed on _____ as Application Serial No. _____ and was amended on _____ (if applicable).

I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56.

I hereby claim priority benefits under Title 35, United States Code §119 of any foreign application(s) for patent or inventor's certificate listed in PART B hereof and have also identified in PART B hereof any foreign application for patent or inventor's certificate having a filing date before that of the application of which priority is claimed.

I hereby claim the benefit under Title 35, United States Code, §119(e) and §120 of any United States application(s) listed in PART C hereof and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph by the first paragraph of Title 35, United States Code §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations §1.56 which occurred between the filing date of the prior application and any national or PCT international filing date of this application.

I hereby declare that all statements made herein of my knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Please direct all telephone calls to Richard L. Gregory, Jr. at (650) 838-4408. Address all correspondence to:

Customer No. 22918
Perkins Coie LLP
101 Jefferson Drive
Menlo Park, California 94025-1114
(650) 838-4300

See page 2 attached, signed, and made a part hereof.

2005.07.15 15:50:50 FOT

INVENTORSHIP DECLARATION BY SOLE INVENTOR

PART A: INVENTOR INFORMATION AND SIGNATURE

Full name of Sole inventor: Gregory C. Burnett

Citizenship: U.S.A.

Mailing Address: 675 South H Street
Livermore, CA 94550

Residence (if different): _____

Inventor's Signature: _____ Date: _____

PART B: CLAIM TO PRIORITY OF FOREIGN APPLICATION(S) UNDER 35 U.S.C. 119(a-d)

Country	App. No.	Filing Date

PART C: CLAIM TO PRIORITY OF U.S. PROVISIONAL AND NONPROVISIONAL APPLICATION(S) UNDER 35 U.S.C. 119(e) AND 120

Serial No.	Filing Date	Status
60/294,383	5/30/01	☒ Pending
09/905,361	7/12/01	☒ Pending
60/335,100	10/30/01	☒ Pending
60/332,202	11/21/01	☒ Pending
09/990,847	11/21/01	☒ Pending
60/362,162	3/5/02	☒ Pending
60/362,103	3/5/02	☒ Pending
60/362,170	3/5/02	☒ Pending
60/361,981	3/5/02	☒ Pending
60/362,161	3/5/02	☒ Pending
60/368,209	3/27/02	☒ Pending
60/368,208	3/27/02	☒ Pending
60/368,343	3/27/02	☒ Pending

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Typical methods for classifying voiced and unvoiced speech have relied mainly on the acoustic content of microphone data, which is plagued by problems with noise and the corresponding uncertainties in signal content. This is especially problematic now with the proliferation of portable communication devices like cellular telephones and personal digital assistants because, in many cases, the quality of service provided by the device depends on the quality of the voice services offered by the device. There are methods known in the art for suppressing the noise present in the speech signals, but these methods demonstrate performance shortcomings that include unusually long computing time, requirements for cumbersome hardware to perform the signal processing, and distorting the signals of interest.

BRIEF DESCRIPTION OF THE FIGURES

Figure 1 is a block diagram of a NAVSAD system, under an embodiment.

Figure 2 is a block diagram of a PSAD system, under an embodiment.

Figure 3 is a block diagram of a denoising system, referred to herein as the Pathfinder system, under an embodiment.

Figure 4 is a flow diagram of a detection algorithm for use in detecting voiced and unvoiced speech, under an embodiment.

Figure 5A plots the received GEMS signal for an utterance along with the mean correlation between the GEMS signal and the Mic 1 signal and the threshold for voiced speech detection.

Figure 5B plots the received GEMS signal for an utterance along with the standard deviation of the GEMS signal and the threshold for voiced speech detection.

Figure 6 plots voiced speech detected from an utterance along with the GEMS signal and the acoustic noise.

Figure 7 is a microphone array for use under an embodiment of the PSAD system.

Figure 8 is a plot of ΔM versus d_1 for several Δd values, under an embodiment.

Figure 9 shows a plot of the gain parameter as the sum of the absolute values of $H_1(z)$ and the acoustic data or audio from microphone 1.

Figure 10 is an alternative plot of acoustic data presented in Figure 9.

In the figures, the same reference numbers identify identical or substantially similar elements or acts.

Any headings provided herein are for convenience only and do not necessarily affect the scope or meaning of the claimed invention.

DETAILED DESCRIPTION

Systems and methods for discriminating voiced and unvoiced speech from background noise are provided below including a Non-Acoustic Sensor Voiced Speech Activity Detection (NAVSAD) system and a Pathfinder Speech Activity Detection (PSAD) system. The noise removal and reduction methods provided herein, while allowing for the separation and classification of unvoiced and voiced human speech from background noise, address the shortcomings of typical systems known in the art by cleaning acoustic signals of interest without distortion.

Figure 1 is a block diagram of a NAVSAD system 100, under an embodiment. The NAVSAD system couples microphones 10 and sensors 20 to at least one processor 30. The sensors 20 of an embodiment include voicing activity detectors or non-acoustic sensors. The processor 30 controls subsystems including a detection subsystem 50, referred to herein as a detection algorithm, and a denoising subsystem 40. Operation of the denoising subsystem 40 is described in detail in the Related Applications. The NAVSAD system works extremely well in any background acoustic noise environment.

Figure 2 is a block diagram of a PSAD system 200, under an embodiment. The PSAD system couples microphones 10 to at least one processor 30. The processor 30 includes a detection subsystem 50, referred to herein as a detection algorithm, and a denoising subsystem 40. The PSAD system is highly sensitive in low acoustic noise environments and relatively insensitive in high acoustic noise environments. The PSAD can operate independently or as a backup to the NAVSAD, detecting voiced speech if the NAVSAD fails.

Note that the detection subsystems 50 and denoising subsystems 40 of both the NAVSAD and PSAD systems of an embodiment are algorithms controlled by the processor 30, but are not so limited. Alternative embodiments of the NAVSAD and PSAD systems can include detection subsystems 50 and/or denoising subsystems 40 that comprise additional hardware, firmware, software, and/or combinations of hardware, firmware, and software. Furthermore, functions

10459700.050602

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

ABSTRACT OF THE DISCLOSURE

Systems and methods are provided for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise. The systems receive acoustic signals at two microphones, and generate difference parameters between the acoustic signals received at each of the two microphones. The difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals. The systems identify information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold, and identify information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold. Further, embodiments of the systems include non-acoustic sensors that receive physiological information to aid in identifying voiced speech.

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Atty. Docket No.: ALPH.P009 *Serial No.* 10/159,770

A-A 5/2/07 Amendment A.A. to be enter

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Attorney Docket No. ALPH.P009 Patent

Transmittal of Office Action Response

Certification of Transmission Under 37 C.F.R. §1.8

Transmitted to USPTO Central Facsimile No. 571-273-8300

APRIL 9, 2007
Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the facsimile number and on the date indicated above.

Barbara B. Courtney
(Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:

Greg Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NON-ACOUSTIC SENSORS

Examiner: Abul K. Azad

Art Unit: 2654

CENTRAL FAX NUMBER 571-273-8300
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

AMENDMENT AND REPONSE UNDER 37 C.F.R.§ 1.116(b)

Sir:

Response under 37 C.F.R. § 1.116(b)

This is a response to the Final office action mailed October 10, 2006. Applicants respectfully request entry of the following amendments and consideration of the

PAGE 4/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-6/21 * DNIS:2738300 * CSID:14083421989 * DURATION (mm-ss):01-48

Please enter after-final Amendment

A. A 4/28/07

IN THE CLAIMS

Amend the claims as indicated below.

Claims 1-5 Canceled.

6. (original) A system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

at least two microphones that receive the acoustic signals;

at least one voicing sensor that receives physiological information associated with human voicing activity; and

at least one processor coupled among the microphones and the voicing sensor, wherein the at least one processor;

generates cross correlation data between the physiological information and an acoustic signal received at one of the two microphones;

identifies information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generates difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifies information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

7. (original) A method for removing noise from acoustic signals, comprising:

receiving the acoustic signals at two receivers and receiving physiological information associated with human voicing activity at a voicing sensor;

generating cross correlation data between the physiological information and an acoustic signal received at one of the two receivers;

PAGE 6/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-6/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

Issue Classification	Application/Control No.	Applicant(s)/Patent under Reexamination	
	10/159,770	BURNETT, GREGORY C.	
	Examiner	Art Unit	
	ABUL K. AZAD	2626	

ISSUE CLASSIFICATION

ORIGINAL		CROSS REFERENCE(S)							
CLASS	SUBCLASS	CLASS	SUBCLASS (ONE SUBCLASS PER BLOCK)						
704	226	704	214						
INTERNATIONAL CLASSIFICATION									
G 1 0 L	11/06								
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(Assistant Examiner) (Date)

D. Johnson 5/1/07
(Legal Instruments Examiner) (Date)

ABUL K. AZAD 4/28/07
(Primary Examiner) (Date)

Total Claims Allowed: 5

O.G. Print Claim(s)	O.G. Print Fig.
6	7

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

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	30		60		90		120		150		180		210

Index of Claims

Application/Control No.	Applicant(s)/Patent under Reexamination
10/159,770	BURNETT, GREGORY C.
Examiner	**Art Unit**
ABUL K. AZAD	2626

√	Rejected	–	(Through numeral) Cancelled	N	Non-Elected	A	Appeal
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Claim		Date								
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Search Notes	Application/Control No.	Applicant(s)/Patent under Reexamination	
	10/159,770	BURNETT, GREGORY C.	
	Examiner	Art Unit	
	ABUL K. AZAD	2626	

SEARCHED			
Class	Subclass	Date	Examiner

INTERFERENCE SEARCHED			
Class	Subclass	Date	Examiner
EAST US-PGPUB searched		4/28/2007	AA

SEARCH NOTES (INCLUDING SEARCH STRATEGY)		
	DATE	EXMR
EAST Searched see attached	4/28/2007	AA

UNITED STATES PATENT AND TRADEMARK OFFICE

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Bib Data Sheet

CONFIRMATION NO. 9714

SERIAL NUMBER	FILING OR 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
10/159,770	05/30/2002 RULE	704	2626	ALPH.P009

APPLICANTS

Gregory C. Burnett, Livermore, CA;

**** CONTINUING DATA ***************************

This appln claims benefit of 60/294,383 05/30/2001 and claims benefit of.60/335,100 10/30/2001
and claims benefit of 60/332,202 11/21/2001
and claims benefit of 60/362,162 03/05/2002
and claims benefit of 60/362,103 03/05/2002
and claims benefit of 60/362,170 03/05/2002
and claims benefit of 60/361,981 03/05/2002
and claims benefit of 60/362,161 03/05/2002
and claims benefit of 60/368,209 03/27/2002
and claims benefit of 60/368,208 03/27/2002
and claims benefit of 60/368,343 03/27/2002

YES A.K

**** FOREIGN APPLICATIONS *********************

None A.A

IF REQUIRED, FOREIGN FILING LICENSE GRANTED
** 07/09/2002

Foreign Priority claimed	STATE OR COUNTRY	SHEETS DRAWING	TOTAL CLAIMS	INDEPENDENT CLAIMS
☐ yes ☒ no	CA	10	7	4

35 USC 119 (a-d) conditions met: ☐ yes ☑ no ☐ Met after Allowance

Verified and Acknowledged: A - A (Examiner's Signature / Initials)

ADDRESS
53186

TITLE

Detecting voiced and unvoiced speech using both acoustic and nonacoustic sensors

FILING FEE RECEIVED	FEES: Authority has been given in Paper	
954	No. _____ to charge/credit DEPOSIT ACCOUNT No. _____ for following:	☐ All Fees ☐ 1.16 Fees (Filing) ☐ 1.17 Fees (Processing Ext. of time) ☐ 1.18 Fees (Issue) ☐ Other _____ ☐ Credit

EAST Search History

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	490	704/208,214.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L2	291	L1 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:13
L3	26	("4032711" | "4052568" | "4357491" | "4401849" | "4410763" | "4433435" | "4531228" | "4535473" | "4552996" | "4627091" | "4630304" | "4696041" | "4718097" | "4815136" | "5151940" | "5222147" | "5305422" | "5313531" | "5323337" | "5479560" | "5579431" | "5617508" | "5649055" | "6038532" | "6266633" | "RE32172").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2007/04/28 17:12
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L5	58	("4052568").URPN.	USPAT	OR	OFF	2007/04/28 17:12
L6	490	704/208,214.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L7	291	L6 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L8	102	L7 and microphone	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L9	2582	voiced and unvoiced	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12

EAST Search History

L10	1187	L9 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L11	484	L10 and microphones	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L12	231	L11 and (background adj noise)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L13	155	L12 and vary$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L14	29	L10 and (two adj microphones)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L15	2	("5590241").PN.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2007/04/28 17:12
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EAST Search History

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L22	155	L21 and vary$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L23	116	L22 and gain	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L24	80	L18 and (first adj threshold) and (second adj threshold)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
L25	53	L24 and (background adj noise)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/04/28 17:12
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EAST Search History

L30	1830	voiced and unvoiced and noise	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/04/28 17:14
L31	97	l30 and physiological	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/04/28 17:15
L32	10	l31 and (cross correlation)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/04/28 17:17
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Atty. Docket No.: ALPH.P009 *Serial No.* 10/159,770

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Attorney Docket No. ALPH.P009 Patent

Transmittal of Office Action Response

Certification of Transmission Under 37 C.F.R. §1.8

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Facsimile No. **571-273-8300**

APRIL 9, 2007
Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the facsimile number and on the date indicated above.

Barbara B. Courtney
(Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:
Greg Burnett
Application No. 10/159,770
Filed: May 30, 2002
For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NON-ACOUSTIC SENSORS

Examiner: Abul K. Azad
Art Unit: 2654

CENTRAL FAX NUMBER **571-273-8300**
Commissioner for Patents
P.O. Box 1450
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AMENDMENT AND REPONSE UNDER 37 C.F.R.§ 1.116(b)

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PAGE 4/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

Remarks.

Petition for extension of time

This response is submitted with a Petition for an extension of time (3 months) under 37 C.F.R. § 1.136(a) to respond to the Final office action. The fee under 37 C.F.R. § 1.17(a)(3) is also submitted herewith.

PAGE 5/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

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generates difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifies information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

7. (original) A method for removing noise from acoustic signals, comprising:

receiving the acoustic signals at two receivers and receiving physiological information associated with human voicing activity at a voicing sensor;

generating cross correlation data between the physiological information and an acoustic signal received at one of the two receivers;

PAGE 6/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

identifying information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifying information of the acoustic signals as noise when the difference parameters are less than the gain thresbold.

8. (New) The method of claim 7, further comprising generating the gain threshold using standard deviations corresponding to the generation of the difference parameters.

9. (New) The method of claim 7, further comprising performing denoising on the identified noise.

10. (New) The method of claim 7, wherein the voicing sensor includes at least one detector selected from a group including radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

PAGE 7/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

REMARKS

Claims 1-7 are pending in the application. Claims 1-5 are canceled without prejudice herein. Claims 8-10 are added. No new matter is added by claims 8-10.

Allowable subject matter

Claims 6 and 7 were indicated to be allowable over the prior art of record. New claims 8-10 include further limitations on the patentable subject matter of claim 7, and are therefore allowable. Claim 8 includes the subject matter of claim 3. Claim 9 includes the subject matter of claim 4. Claim 10 includes the subject matter of claim 5.

Rejections under 35 U.S.C. § 103

Claims 1-5 were rejected under 35 U.S.C. § 103(a) as being unpatentable over Park et al. (U.S. Patent No. 5,590,241, hereinafter "Park"), in view of Cho et al (US Patent No. 6,233,551, hereinafter "Cho"). Claims 1-5 have been canceled herein without prejudice.

Applicant respectfully submits that the only claims remaining in the application are indicated to be allowable over the cited references.

PAGE 8/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

CONCLUSION

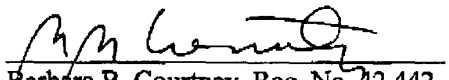
In view of the foregoing amendments and remarks, Applicants respectfully submit that claims 6-10 are in condition for allowance. The Examiner is respectfully requested to telephone the undersigned to resolve any remaining issues before allowance of the claims as amended.

AUTHORIZATION TO CHARGE DEPOSIT ACCOUNT

Please charge deposit account 503616 for any fees due, and not paid herewith, in connection with this application.

Respectfully submitted,
Courtney Staniford & Gregory LLP

Date: April 9, 2007



Barbara B. Courtney, Reg. No. 42,442
Tel. 408-342-1902

Courtney Staniford & Gregory LLP
P.O. Box 9686
San Jose CA 95157

PAGE 9/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

PTO/SB/22 (12-04)
Approved for use through 07/31/2006, OMB 0651-0031
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PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a) FY 2005 *(Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).)*	Docket Number (Optional) ALPH.P009
Application Number 10/159,770	Filed 05/30/2002
For Detecting voiced and unvoiced speech using both acoustic and non-acoustic sensors	
Art Unit 2626	Examiner Abul K. Azad

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	
☐	One month (37 CFR 1.17(a)(1))	$120	$60	$
☐	Two months (37 CFR 1.17(a)(2))	$450	$225	$
☑	Three months (37 CFR 1.17(a)(3))	$1020	$510	$ 510
☐	Four months (37 CFR 1.17(a)(4))	$1590	$795	$
☐	Five months (37 CFR 1.17(a)(5))	$2160	$1080	$

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

☐ 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 _____. I have enclosed a duplicate copy of this sheet.

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/inventor.

☐ assignee of record of the entire interest. See 37 CFR 3.71. Statement under 37 CFR 3.73(b) is enclosed (Form PTO/SB/96).

☑ attorney or agent of record. Registration Number 42,442

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

Signature	April 9, 2007 Date
Barbara B. Courtney Typed or printed name	408-342-1902 Telephone Number

NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below.

☐ Total of _____ forms are submitted.

This collection of information is required by 37 CFR 1.136(a). 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 1.11 and 1.14. This collection is estimated to take 6 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

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PAGE 2/9 * RCVD AT 4/9/2007 9:42:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/21 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-48

Courtney Staniford & Gregory LLP

Intellectual Property Attorneys

Tel: 408-342-1900
Fax: 408-342-1909

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San Jose, California 95157
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Fax

To: Art Unit 2626
USPTO

From: Barbara Courtney
Courtney Staniford & Gregory & LLP
Phone: 408-342-1902
Fax: 408-342-1909

Fax: **571-273-8300** Pages: 9 including cover

Phone: 571-272-7599 Date: April 9, 2007

Re: **Application Serial No. 10/159,770**
Art Unit 2626
Examiner Abul K. Azad

CC:

Atty Docket No. ALPH.P009

☑ Urgent ☑ For Review ☑ Please Comment ☑ Please Acknowledge

Attorney Docket No. CSHE.P007 Patent

Transmittal of Office Action Response

Certification of Transmission Under 37 C.F.R. §1.8

Transmitted to USPTO Central Facsimile No. 571-273-8300

April 9, 2007
Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the facsimile number and on the date indicated above under 37 CFR 1.8.

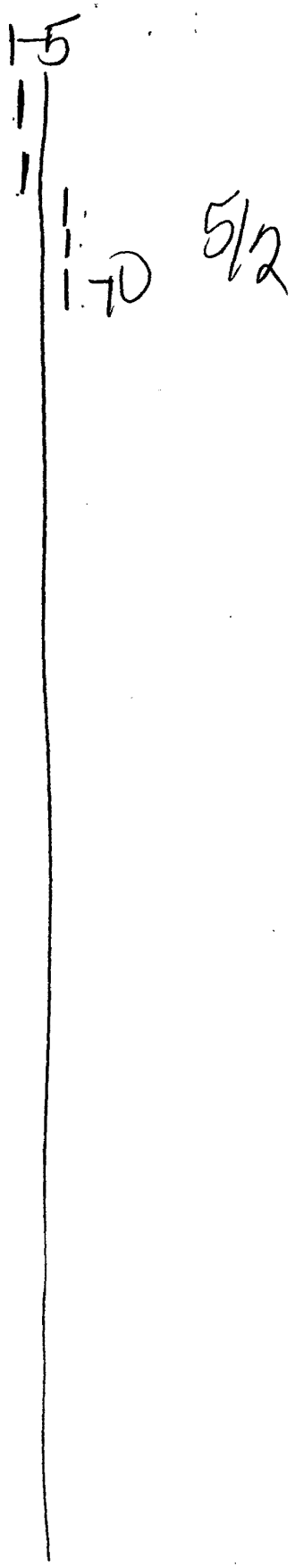
Barbara B. Courtney
(Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

1. Response to the Final Office Action mailed October 10, 2006
2. Petition for three month Extension of Time; and
3. Authorization to charge Credit Card.

Confidentiality Note

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I-5
I-70
5/2

10/189,770

PATENT APPLICATION FEE DETERMINATION RECORD
Effective October 1, 2001

Application or Docket Number: 37212.8009.US00

CLAIMS AS FILED - PART I

	(Column 1)	(Column 2)
TOTAL CLAIMS	7	
FOR	NUMBER FILED	NUMBER EXTRA
TOTAL CHARGEABLE CLAIMS	7 minus 20=	* 0
INDEPENDENT CLAIMS	4 minus 3 =	* 1
MULTIPLE DEPENDENT CLAIM PRESENT		☐

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

SMALL ENTITY TYPE ☐		OR	OTHER THAN SMALL ENTITY	
RATE	FEE		RATE	FEE
BASIC FEE	370.00	OR	BASIC FEE	740.00
X$ 9=		OR	X$18=	
X42=		OR	X84=	84
+140=		OR	+280=	
TOTAL		OR	TOTAL	824

4/9/07

CLAIMS AS AMENDED - PART II

AMENDMENT A

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
Total	* 5	Minus	** 20	=
Independent	* 2	Minus	*** 3	=
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

SMALL ENTITY		OR	OTHER THAN SMALL ENTITY	
RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT B

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
Total	*	Minus	**	=
Independent	*	Minus	***	=
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
Total	*	Minus	**	=
Independent	*	Minus	***	=
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. 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.

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

53186 7590 10/10/2006

COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

EXAMINER	
AZAD, ABUL K	

ART UNIT	PAPER NUMBER
2626	

DATE MAILED: 10/10/2006

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

PTO-90C (Rev. 10/03)

Office Action Summary	Application No.	Applicant(s)	
	10/159,770	BURNETT, GREGORY C.	
	Examiner	Art Unit	
	ABUL K. AZAD	2626	

-- 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 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, 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 *15 June 2006*.

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

3)☐ 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

4)☒ Claim(s) *1-7* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☒ Claim(s) *6 and 7* is/are allowed.

6)☒ Claim(s) *1-5* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ 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).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

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).

a)☐ All b)☐ Some * c)☐ None of:

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) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. This action is in response to the communication filed on June 17, 2006.

2. Claims 1-7 are pending in this action. Claims 1 and 2 have been amended.

3. The applicant's arguments with respect to claims 1-5 have been fully considered but they are not deemed to be persuasive. For examiner's response to the applicant's arguments or comments, see the detailed discussion in the Response to the Arguments section.

Claim Rejections - 35 USC § 103

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

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

5. Claims 1-5 are rejected under 35 U.S.C. 103(a) as being unpatentable over Park et al. (US 5,590,241) in view of Cho et al. (US 6,233,551).

As per claim 1, Park and Cho teach, "a system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise", comprising:

Park teaches, "at least two microphones for receiving the acoustic signals" (Fig. 1, elements 31 and 34);

"at least one processor coupled among the microphones, wherein the at least one processor" (Fig. 1, element 30);

"generates difference parameters between the acoustic signals received at each of the two microphones, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals" (Fig. 1, element 38);

Park teaches a signal predictor (Fig. 4, element 90) and determined voiced and unvoiced signal (col. 4, lines 10-36) but does not explicitly teach "identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold". However, Cho teaches, ""identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold" (col. 5, lines 10-56). Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to use Cho's teaching in the invention of Park because Cho teaches his invention provide a vocoder with improved sound quality by implementing voicing according to his invention (col. 6, lines 14-20).

As per claim 2, it is interpreted and thus rejected for the same reasons set forth in the rejection of claim 1.

As per claim 3, Park teaches, "generating the first and second thresholds using standard deviations corresponding to the generation of the difference parameters" (col. 6, lines 42-54).

As per claim 4, Park teaches, "identifying information of the acoustic signals as noise when the difference parameters are less than the first threshold; and performing denoising on the identified noise" (col. 7, lines 1-6).

As per claim 5, Park teaches, "further comprising receiving physiological information associated with human voicing activity, wherein the physiological information comprises receiving physiological data associated with human voicing using at least one detector selected from a group including radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors" (Fig. 1, element 34).

Allowable Subject Matter

6. Claims 6 and 7 are allowed over the prior art of record.

Response to Arguments

7. The applicant argues that element 34 is not a microphone is an accelerometer.

The examiner disagrees with applicant above assertion because Park teaches at the background section a microphone at col. 1, lines 22-44 and also he said that the microphone configuration could be as two microphone at col. 8, lines 14-25.

8. The applicant further argues that Cho does not teach detecting voice and unvoiced speech as claimed.

The examiner disagrees because voicing level is as voiced and unvoiced determinations please see at col. 3, lines 34-40.

9. In response to applicant's argument that propose combination does not result in the claimed invention, the test for obviousness is not whether the features of a secondary reference may be bodily incorporated into the structure of the primary reference; nor is it that the claimed invention must be expressly suggested in any one or all of the references. Rather, the test is what the combined teachings of the references would have suggested to those of ordinary skill in the art. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981).

Conclusion

10. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Contact Information

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to **Abul K. Azad** whose telephone number is **(571) 272-7599.** If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, **Richemond Dorvil**, can be reached at **(571) 272-7602.**

Any response to this action should be mailed to:

Commissioner for Patents

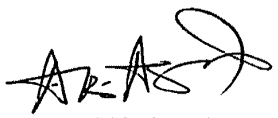
P.O. Box 1450

Alexandria, VA 22313-1450

Or faxed to: **(571) 273-8300.**

Hand-delivered responses should be brought to **401 Dulany Street, Alexandria, VA-22314** (Customer Service Window).

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).



September 29, 2006

Abul K. Azad
Primary Examiner
Art Unit 2626

Index of Claims

Application/Control No.	Applicant(s)/Patent under Reexamination
10/159,770	BURNETT, GREGORY C.
Examiner	**Art Unit**
ABUL K. AZAD	2626

√	Rejected	–	(Through numeral) Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

Claim		Date
Final	Original	9/29/06
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APPLICATION NUMBER	FILING OR 371 (c) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009

CONFIRMATION NO. 9714

22918
PERKINS COIE LLP
P.O. BOX 2168
MENLO PARK, CA 94026

OC000000019487760

Date Mailed: 07/05/2006

NOTICE REGARDING CHANGE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 06/17/2006.

- The Power of Attorney to you in this application has been revoked by the assignee who has intervened as provided by 37 CFR 3.71. Future correspondence will be mailed to the new address of record(37 CFR 1.33).

SHARON KUANG
PTOSS (703) 305-3006

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APPLICATION NUMBER	FILING OR 371 (c) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009

CONFIRMATION NO. 9714

53186
COURTNEY STANIFORD & GREGORY LLP
P.O. BOX 9686
SAN JOSE, CA 95157

OC000000019487866

Date Mailed: 07/05/2006

NOTICE OF ACCEPTANCE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 06/17/2006.

The Power of Attorney in this application is accepted. Correspondence in this application will be mailed to the above address as provided by 37 CFR 1.33.

SHARON KUANG
PTOSS (703) 305-3006

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JUN 17 2006

Attorney Docket No. ALPH.P009 Patent

Transmittal of Power of Attorney

Certification Under 37 C.F.R. §1.8(a)

Transmitted to USPTO Central Facsimile No. 571-273-8300

June 17, 2006
Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the Central Facsimile number on the date indicated above.

Barbara B. Courtney
(Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

Fax

To: USPTO

From: Barbara Courtney

Courtney Staniford & Gregory LLP
Phone: 408-342-1902
Fax: 408-342-1909

Fax: 571-273-8300 Pages: 3 including cover

Phone: Date: June 17, 2006

RE: U.S. Patent Application No. 10/159,770
Filed May 30, 2002

Attorney docket no. ALPH.P009

The following are submitted herewith on the date indicated above:

Power of Attorney (PTO/SB/80); and
Statement under 37 CFR 3.73(b) (PTO/SB/96).

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PAGE 1/3 * RCVD AT 6/17/2006 4:11:32 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-3/20 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):01-30

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I hereby revoke all previous powers of attorney given in the application identified in the attached statement under 37 CFR 3.73(b).

I hereby appoint:

☑ Practitioners associated with the Customer Number: 53186

OR

☐ Practitioner(s) named below (if more than ten patent practitioners are to be named, then a customer number must be used):

Name	Registration Number	Name	Registration Number

as attorney(s) or agent(s) to represent the undersigned before the United States Patent and Trademark Office (USPTO) in connection with any and all patent applications assigned only to the undersigned according to the USPTO assignment records or assignment documents attached to this form in accordance with 37 CFR 3.73(b).

Please change the correspondence address for the application identified in the attached statement under 37 CFR 3.73(b) to:

☑ The address associated with Customer Number: 53186

OR

☐ Firm or Individual Name			
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Country			
Telephone		Email	

Assignee Name and Address:

AliphCom, Inc., 150 Executive Park Blvd, Suite 4550, San Francisco, CA 94134

A copy of this form, together with a statement under 37 CFR 3.73(b) (Form PTO/SB/96 or equivalent) is required to be filed in each application in which this form is used. The statement under 37 CFR 3.73(b) may be completed by one of the practitioners appointed in this form if the appointed practitioner is authorized to act on behalf of the assignee, and must identify the application in which this Power of Attorney is to be filed.

SIGNATURE of Assignee of Record

The individual whose signature and title is supplied below is authorized to act on behalf of the assignee

Signature		Date	6/14/2006
Name	Gregory C. Burnett	Telephone	650-276-0515
Title	Chief Technical Officer		

This collection of information is required by 37 CFR 1.31, 1.32 and 1.33. 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 1.11 and 1.14. This collection is estimated to take 3 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.

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STATEMENT UNDER 37 CFR 3.73(b)

Applicant/Patent Owner: Allphcom, Inc.

Application No./Patent No.: 10/159,770 Filed/Issue Date: 05/30/2002

Entitled: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

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

states that it is:

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

2. ☐ an assignee of less than the entire right, title and interest.
The extent (by percentage) of its ownership interest is _____ %

in the patent application/patent identified above by virtue of either:

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 013855, Frame 0906, or for which a copy thereof is attached.

OR

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

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

☐ Copies of assignments or other documents in the chain of title are attached.
[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, if the assignment is to be recorded 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.

Signature	June 17, 2006 Date
Barbara B. Courtney Printed or Typed Name	408 342 1902 Telephone Number

Patent Attorney Representative of Assignee, Reg. No. 42,442
Title

This collection of information is required by 37 CFR 3.73(b). 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 1.11 and 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, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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Attorney Docket No. ALPH.P009 Patent

Transmittal of Response to Office Action

Certification Under 37 C.F.R. §1.8(a)

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June 15, 2006
Date of Transmission

I hereby certify that this document, and any other accompanying documents referred to herein are being transmitted via facsimile to the United States Patent Office to the facsimile number and on the date indicated above.

Richard L. Gregory, Jr.
(Print Name of Person Transmitting Facsimile)

(Signature of Person Transmitting Facsimile)

Atty. Docket No. ALPH.P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:

Gregory C. Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NON-ACOUSTIC SENSORS

Examiner: Abul K. Azad
Art Unit: 2654

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

AMENDMENT AND RESPONSE UNDER 37 C.F.R. § 1.111

Sir:

Response under 37 C.F.R. § 1.111

This is a response to the Office Action mailed December 15, 2005. Applicant respectfully requests entry of the following amendments and consideration of the Remarks.

PAGE 4/12 * RCVD AT 6/15/2006 5:06:33 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/18 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):03-04

Attorney Docket No. ALPH.P009

Petition for Extension of time under 37 C.F.R. § 1.136

Applicant submits herewith a petition for a three month extension of time, including the fee under 37 C.F.R. § 1.17.

PAGE 5/12 * RCVD AT 6/15/2006 5:06:33 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/18 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):03-04

In the Claims

Please amend the claims as follows.

1. (currently amended) A system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

at least two microphones for receiving the acoustic signals; and

at least one processor coupled among the microphones, wherein the at least one processor, ~~processor;~~

generates difference parameters between the acoustic signals received at each of the two microphones, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold; and

identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold.

2. (currently amended) A method for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

receiving the acoustic signals at two receivers;

generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of a ~~the~~ relative difference in signal gain between portions of the received acoustic signals;

identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold; and

identifying information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold.

3. (original) The method of claim 2, further comprising generating the first and second thresholds using standard deviations corresponding to the generation of the difference parameters.

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4. (original) The method of claim 2, further comprising:

identifying information of the acoustic signals as noise when the difference parameters are less than the first threshold; and

performing denoising on the identified noise.

5. (original) The method of claim 2, further comprising receiving physiological information associated with human voicing activity, wherein the physiological information comprises receiving physiological data associated with human voicing using at least one detector selected from a group including radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

6. (original) A system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

at least two microphones that receive the acoustic signals;

at least one voicing sensor that receives physiological information associated with human voicing activity; and

at least one processor coupled among the microphones and the voicing sensor, wherein the at least one processor;

generates cross correlation data between the physiological information and an acoustic signal received at one of the two microphones;

identifies information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generates difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifies information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

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7. (original) A method for removing noise from acoustic signals, comprising:

receiving the acoustic signals at two receivers and receiving physiological information associated with human voicing activity at a voicing sensor;

generating cross correlation data between the physiological information and an acoustic signal received at one of the two receivers;

identifying information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifying information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

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Attorney Docket No. ALPH.P009

REMARKS

Claims 1-7 are pending in the application. No claims have been allowed. Claims 1 and 2 have been amended. No new material is added by the amendments herein.

Claim Rejections 35 USC §103

Claims 1-7 were rejected under 35 USC §103(a) as being unpatentable over Park et al., United States Patent number 5,590,241 ("Park"), in view of Cho et al., United States Patent number 6,233,551 ("Cho"). Applicants respectfully submit that the claims would not have been obvious in view of the cited art.

Park teaches a speech processing system (30) that operates in a noisy environment (20) by performing adaptive prediction between inputs from two sensors positioned to transduce speech from a speaker, such as an accelerometer and a microphone. An adaptive filter (37) such as a finite impulse response (FIR) filter receives a digital accelerometer input signal, adjusts filter coefficients according to an estimation error signal, and provides an enhanced speech signal as an output. (Abstract).

Park teaches an adaptive filter that attempts to make a sensor signal more speech-like. This is in contrast to the claimed invention, which includes voice activity detection. There is no teaching or suggestion in Park regarding voice activity detection.

Regarding claim 1, the Examiner states Park discloses using at least two microphones (Figure 1, elements 31 and 34). Applicants respectfully disagree. Park is limited to teaching using one microphone only. Element 34 is an accelerometer, which is not a microphone.

The Exmainer further states that Park teaches at least one processor (Figure 1, element 30). Applicants note that element 30 is a system that includes the elements pointed to, but it is not a processor. Nor does the system 30 include a processor.

The Examiner further states that Park generates difference parameters between the acoustic signals received at the two microphones. First, Park teaches only one microphone. Second, Park specifies a summer (Figure 1, element 38) that sums the microphone signal and the adapted accelerometer signal as part of a standard adaptive filter. The summer does not generate difference parameters.

The Examiner further states that:

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Park teaches a signal predictor (Fig. 4, element 90), but does not explicitly teach "identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold". However, Cho teaches "identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold" (col. 5, lines 10-56). (column 5, lines 10-56).

Applicants respectfully disagree with this interpretation of Cho and of the combination of Park and Cho.

Cho teaches a single microphone system and method. In particular, Cho teaches a method and an apparatus for determining multiband voicing levels using a frequency moving method in a vocoder. Cho's method for determining the multiband voicing levels using the frequency moving method in the vocoder includes the steps of (a) applying a window to an input voice signal and obtaining a power spectrum from a voice spectrum obtained by Fourier converting a windowed signal, (b) moving the frequency of each subband to an origin after dividing the power spectrum into a predetermined number of subbands, (c) obtaining autocorrelation values of the respective subbands by inverse Fourier converting the power spectrum the frequency of which is moved to the origin, and (d) normalizing the respective autocorrelation values and determining the voicing levels of the subbands from the normalized autocorrelation values. (Abstract).

Applicants respectfully submit that Cho does not teach detecting voiced and unvoiced speech as claimed, but rather determining the voicing levels of the subbands (of the single microphone signal) from the normalized autocorrelation values.

Therefore, Applicants submit that the proposed combination does not result in the claimed invention which includes a multiple microphone system that generates difference parameters and identifies speech as voiced or unvoiced. Applicants respectfully submit that neither of the cited references contain any of these elements, nor do they suggest them. Further, at least because the suggested combination of the references would not result in the claimed invention, Applicants submit that one of ordinary skill in the art would find no motivation to combine the references. Neither of the references provide any motivation to combine to achieve the claimed system for detecting voiced and

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unvoiced speech in acoustic signals having varying levels of background noise. For example, Park refers to an adaptive filtering algorithm, while Cho refers to a vocoder. For all of these reasons, Applicants submit that claim 1 would not have been obvious in view of the cited references.

Claim 2 recited a method for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, including receiving the acoustic signals at two receivers, generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals, identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold, and identifying information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold. Applicants respectfully incorporate the above remarks with reference to claim 1 here. Applicants have shown that the cited references, alone or in combination, do not teach or suggest the invention of claim 2. Therefore, Applicants submit that claim 2 would not have been obvious in view of the cited references.

Dependent claims 3-5 include further limitations on claim 2. Therefore, Applicants submit that claims 3-5 are patentable over the cited references for the same reasons.

Claim 6 recites a system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, including at least two microphones that receive the acoustic signals. Further, the claimed system generates difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals. The claimed system further identifies information of the acoustic signals as unvoiced or unvoiced speech. As remarked with reference to claim 1, the cited references alone or together lack at lease these elements. Therefore, Applicants submit that claim 6 would not have been obvious in view of the cited references.

Claim 7 recites a method for removing noise from acoustic signals, including receiving the acoustic signals at two receivers and receiving physiological information

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associated with human voicing activity at a voicing sensor, generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals, and identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold. As discussed with reference to claim 1, the cited references alone or together lack at lease these elements. Therefore, Applicants submit that claim 7 would not have been obvious in view of the cited references.

Conclusion

In view of the foregoing amendments and remarks, Applicants respectfully submit that claims 1-7 as presented herein are in condition for allowance. Thus, allowance of the claims is requested. If in the opinion of Examiner Azad a telephone conference would expedite the prosecution of the subject application, or if there are any issues that remain to be resolved prior to allowance of the claims, Examiner Azad is encouraged to call Barbara Courtney at (408) 342-1902.

AUTHORIZATION TO CHARGE DEPOSIT ACCOUNT:

The USPTO is hereby authorized to charge our deposit account number 503616 for any underpayments required for this submission.

Respectfully submitted,
Courtney Staniford & Gregory LLP

Date: June 15, 2006

Richard L. Gregory, Jr.
Reg. No. 42,607
Tel. 408-342-1900

PAGE 12/12 * RCVD AT 6/15/2006 5:06:33 PM [Eastern Daylight Time] * SVR:USPTO-EFXRF-5/18 * DNIS:2738300 * CSID:14083421909 * DURATION (mm-ss):03-04

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Richard L. Gregory, Jr. R. Gregory
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Fax

To:	USPTO Art Unit 2654 Examiner: Abul K. Azad	From:	Barbara Courtney Courtney Staniford & Gregory LLP Phone: 408-342-1902 Fax: 408-342-1909
Fax:	571-273-8300	Pages:	12 including cover
Phone:	571-272-7599	Date:	June 15, 2006

RE: U.S. Patent Application No. 10/159,770

The following are submitted herewith on the date indicated above:

Response to the Final Office Action mailed December 15, 2005;
Petition for extension of time; and
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PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a) FY 2005 *(Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).)*	Docket Number (Optional) ALPH.P009
Application Number 10/159,770	Filed 05/30/2002
For Detecting Voiced and Unvoiced Speech using Both Acoustic and Nonacoustic Sensors	
Art Unit 2162	Examiner Abul K. Azad

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.

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		Fee	Small Entity Fee	
☐	One month (37 CFR 1.17(a)(1))	$120	$60	$
☐	Two months (37 CFR 1.17(a)(2))	$450	$225	$
☑	Three months (37 CFR 1.17(a)(3))	$1020	$510	$ 510
☐	Four months (37 CFR 1.17(a)(4))	$1590	$795	$
☐	Five months (37 CFR 1.17(a)(5))	$2160	$1080	$

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

☐ 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 50-3616. I have enclosed a duplicate copy of this sheet.

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I am the

☐ applicant/inventor.

☐ assignee of record of the entire interest. See 37 CFR 3.71.
Statement under 37 CFR 3.73(b) is enclosed (Form PTO/SB/96).

☑ attorney or agent of record. Registration Number 42,442

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

Signature — 06/15/2006 Date

Barbara B. Courtney — 408-342-1902

Typed or printed name — Telephone Number

06/16/2006 HDIRAS 00000021 10159770
01 FC:2253 510.00 OP

NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below.

☐ Total of _____ forms are submitted.

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/159,770	05/30/2002	Gregory C. Burnett	ALPH.P009	9714

22918 7590 12/15/2005

PERKINS COIE LLP
P.O. BOX 2168
MENLO PARK, CA 94026

EXAMINER	
AZAD, ABUL K	

ART UNIT	PAPER NUMBER
2654	

DATE MAILED: 12/15/2005

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

PTO-90C (Rev. 10/03)

Office Action Summary	Application No.	Applicant(s)	
	10/159,770	BURNETT, GREGORY C.	
	Examiner	Art Unit	
	ABUL K. AZAD	2654	

-- 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 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, 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 *30 May 2002*.

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

3)☐ 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

4)☒ Claim(s) *1-7* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-7* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *08 August 2002* 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).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

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).

a)☐ All b)☐ Some * c)☐ None of:

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) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date *11/5/04 12/27/02* 11/21/02
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

1. Claims 1-7 are pending in this Office Action.

Claim Rejections - 35 USC § 103

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

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

3. Claims 1-7 rejected under 35 U.S.C. 103(a) as being unpatentable over Park et al. (US 5,590,241) in view of Cho et al. (US 6,233,551).

As per claim 1, Park and Cho teach, "a system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise", comprising:

Park teaches, "at least two microphones for receiving the acoustic signals" (Fig. 1, elements 31 and 34);

"at least one processor coupled among the microphones, wherein the at least one processor" (Fig. 1, element 30);

"generates difference parameters between the acoustic signals received at each of the two microphones, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals" (Fig. 1, element 38);

Park teaches a signal predictor (Fig. 4, element 90) but does not explicitly teach "identifies information of the acoustic signals as unvoiced speech when the difference

parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold". However, Cho teaches, ""identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold and identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold" (col. 5, lines 10-56). Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to use Cho's teaching in the invention of Park because Cho teaches his invention provide a vocoder with improved sound quality by implementing voicing according to his invention (col. 6, lines 14-20).

As per claim 2, it is interpreted and thus rejected for the same reasons set forth in the rejection of claim 1.

As per claim 3, Park teaches, "generating the first and second thresholds using standard deviations corresponding to the generation of the difference parameters" (col. 6, lines 42-54).

As per claim 4, Park teaches, "identifying information of the acoustic signals as noise when the difference parameters are less than the first threshold; and performing denoising on the identified noise" (col. 7, lines 1-6).

As per claim 5, Park teaches, "further comprising receiving physiological information associated with human voicing activity, wherein the physiological information comprises receiving physiological data associated with human voicing using at least one detector selected from a group including radio frequency devices,

electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors" (Fig. 1, element 34).

As per claims 6 and 7, they are interpreted and thus rejected for the same reasons set forth in the rejection of claims 1 and 3-5.

Contact Information

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to **Abul K. Azad** whose telephone number is **(571) 272-7599.** If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, **Richemond Dorvil**, can be reached at **(571) 272-7602.**

Any response to this action should be mailed to:

Commissioner for Patents

P.O. Box 1450

Alexandria, VA 22313-1450

Or faxed to: **(571) 273-8300.**

Hand-delivered responses should be brought to **401 Dulany Street, Alexandria, VA-22314** (Customer Service Window).

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).

December 10, 2005

Abul K. Azad
Primary Examiner
Art Unit 2654

OIPE JC14 NOV 05 2004 PATENT & TRADEMARK OFFICE

PTO/SB/08A (08-03)
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Substitute for form 1449/PTO

INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(Use as many sheets as necessary)

Sheet 1 of 1

Complete if Known	
Application Number	10/159,770
Filing Date	05/30/2002
First Named Inventor	Gregory C. Burnett
Art Unit	~~2644~~ 2654
Examiner Name	~~Not yet assigned~~ A. Zad
Attorney Docket Number	ALPH.P009

U. S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Document Number Number-Kind Code[2] (if known)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
A. A		US- 2002/039425 A1	04-04-2002	Burnett et al.	
		US- 3,789,166	01-29-1974	Sebesta	
		US- 4,006,318	02-01-1977	Sebesta et al.	
		US- 4,591,668	05-27-1986	Iwata	
		US- 4,901,354	02-13-1990	Gollmar et al.	
		US- 5,473,702	12-05-1995	Yoshida et al.	
		US- 5,515,865	05-14-1996	Scanlon et al.	
		US- 5,649,055	07-15-1997	Gupta et al.	
		US- 5,684,460	11-04-1997	Scanlon et al.	
		US- 5,729,694	03-17-1998	Holzrichter et al.	
		US- 5,754,665	05-19-1998	Hosoi et al.	
		US- 5,853,005	12-29-1998	Scanlon	
		US- 6,069,963	05-30-2000	Martin et al.	
		US- 6,191,724	02-20-2001	McEwan	
		US- 6,266,422	07-24-2001	Ikeda	
A. A		US- 6,430,295	08-06-2002	Handel et al.	
A. A		US- 5,097,515	03-17-2002	Baba	
		US-			
		US			

FOREIGN PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Foreign Patent Document Country Code[3]-Number[4]-Kind Code[5] (if known)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages Or Relevant Figures Appear	T[6]
A. A		EP 0 637 187 A	02-01-1995	Pan Communicati		
A. A		EP 0 795 851 A2	09-17-1997	Tokyo Shibaura		
A. A		EP 0 984 660 A2	03-08-2000	Nippon T&T Corp		
A. A		JP 2000 312 395	11-07-2000	Alpine Electron		
A. A		JP 2001 189 987	07-10-2001	Pioneer Electro		
A. A		WO 02 07151	01-24-2002	ALIPHCOM, Inc.		

Examiner Signature	A. Zad	Date Considered	12/10/05

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant. [1] Applicant's unique citation designation number (optional). [2] See Kinds Codes of USPTO Patent Documents at www.uspto.gov or MPEP 901.04. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

This collection of information is required by 37 CFR 1.97 and 1.98. 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 2 hours 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, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.

If you need assistance in completing the form, call 1-800-PTO-9199 (1-800-786-9199) and select option 2.

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(use as many sheets as necessary)

Sheet 1 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	~~2644~~ 2654
Examiner Name	~~Unknown~~ A. Azad
Attorney Docket Number	ALPH.P009

U.S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	U.S. Patent Document Number	Kind Code[2] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
AA		5,406,622		Silverberg et al.	04-11-1995	
AA		5,517,435		Sugiyama	05-14-1996	

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FOREIGN PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Foreign Patent Document Office[3]	Number[4]	Kind Code[5] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear	T[6]

Examiner Signature	A. Azad	Date Considered	12/10/05

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[1] Unique citation designation number. [2] See attached Kinds of U.S. Patent Documents. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST. 16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(use as many sheets as necessary)

Sheet 2 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	~~2644~~ 2654
Examiner Name	~~Unknown~~ AZAD
Attorney Docket Number	ALPH.P009

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OTHER PRIOR ART – NON PATENT LITERATURE DOCUMENTS

Examiner Initials*	Cite No.[1]	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.), date, page(s), volume-issue number(s), publisher, city and/or country where published.	T[2]
A.A	✓	GREGORY C. BURNETT: "The Physiological Basis of Glottal Electromagnetic Micropower Sensors (GEMS) and Their Use in Defining an Excitation Function for the Human Vocal Tract", Dissertation, University of California at Davis, January, 1999, USA.	
A.A	✓	TODD J. GABLE et al.: "Speaker Verification Using Combined Acoustic and EM Sensor Signal Processing", IEEE Intl. Conf. on Acoustics, Speech & Signal Processing (ICASSP-2001), Salt Lake City, USA, 2001.	
A.A	✓	A. HUSSAIN: "Intelligibility Assessment of a Multi-Band Speech Enhancement Scheme", Proceedings IEEE Intl. Conf. on Acoustics, Speech & Signal Processing (ICASSP-2000), Istanbul, Turkey, June 2000.	

Examiner Signature	A.K. A	Date Considered	12/10/05

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

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Sheet 1 of 2

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Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	~~2644~~ 2654
Examiner Name	~~Unknown~~ AZAD
Attorney Docket Number	ALPH.P009

U.S. PATENT DOCUMENTS

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A.A		5,212,764		Ariyoshi	05-18-1993	
		5,400,409		Linhard	03-21,1995	
		5,414,776		Sims, Jr.	05-09-1995	
		5,539,859		Robbe et al.	07-23-1996	
		5,633,935		Kanamori et al.	05-27-1997	
		5,835,608		Wamaka et al.	11-10-1998	
		5,917,921		Sasaki et al.	06-29-1999	
		6,000,396		Nagata	12-28-1999	
		6,191,724	B1	McEwan	02-20-2001	
		5,986,600		McEwan	11-16-1999	
		5,966,090		McEwan	10-12-1999	
A.A		6,006,175		Holzrichter	12-21-1999	

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[1] Unique citation designation number. [2] See attached Kinds of U.S. Patent Documents. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST. 16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(use as many sheets as necessary)

Sheet 2 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	~~2644~~ 2655
Examiner Name	~~Unknown~~ A-ZAB
Attorney Docket Number	ALPH.P009

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OTHER PRIOR ART – NON PATENT LITERATURE DOCUMENTS			
Examiner Initials*	Cite No.[1]	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.), date, page(s), volume-issue number(s), publisher, city and/or country where published.	T[2]
A.A	✓	Zhao Li et al: "Robust Speech Coding Using Microphone Arrays", Signals Systems and Computers, 1997. Conf. record of 31st Asilomar Conf., Nov. 2-5, 1997, IEEE Comput. Soc. Nov. 2, 1997, USA.	
A.A	✓	L.C. Ng et al.: "Denoising of Human Speech Using Combined Acoustic and EM Sensor Signal Processing", 2000 IEEE Intl Conf on Acoustics Speech and Signal Processing. Proceedings (Cat. No. 00CH37100), Istanbul, Turkey, June 5-9, 2000 XP002186255, ISBN 0-7803-6293-4	
A.A	✓	S. Affes et al.: "A Signal Subspace Tracking Algorithm for Microphone Array Processing of Speech". IEEE Transactions on Speech and Audio Processing, N.Y, USA Vol. 5, No. 5, Sept. 1, 1997. XP000774303, ISBN 1063-6676	

Examiner Signature	A.A	Date Considered	12/10/03

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

[1] Unique citation designation number. [2] Applicant is to place a check mark here if English language Translation is attached.

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Notice of References Cited	Application/Control No. 10/159,770	Applicant(s)/Patent Under Reexamination BURNETT, GREGORY C.	
	Examiner ABUL K. AZAD	Art Unit 2654	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	Classification
*	A	US-5,590,241	12-1996	Park et al.	704/227
*	B	US-6,233,551	05-2001	Cho et al.	704/208
*	C	US-4,653,102	03-1987	Hansen, Per K.	381/92
*	D	US-5,664,052	09-1997	Nishiguchi et al.	704/214
*	E	US-6,006,175	12-1999	Holzrichter, John F.	704/208
*	F	US-4,777,649	10-1988	Carlson et al.	704/233
	G	US-			
	H	US-			
	I	US-			
	J	US-			
	K	US-			
	L	US-			
	M	US-			

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	Classification
	N					
	O					
	P					
	Q					
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	S					
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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.	Applicant(s)/Patent under Reexamination
10/159,770	BURNETT, GREGORY C.
Examiner	**Art Unit**
ABUL K. AZAD	2654

√	Rejected	–	(Through numeral) Cancelled	N	Non-Elected	A	Appeal
=	Allowed	+	Restricted	I	Interference	O	Objected

Claim		Date
Final	Original	12/10/05
	1	√
	2	√
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Claim		Date
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Search Notes	Application/Control No.	Applicant(s)/Patent under Reexamination
	10/159,770	BURNETT, GREGORY C.
	Examiner	Art Unit
	ABUL K. AZAD	2654

SEARCHED			
Class	Subclass	Date	Examiner

INTERFERENCE SEARCHED			
Class	Subclass	Date	Examiner

SEARCH NOTES (INCLUDING SEARCH STRATEGY)		
	DATE	EXMR
EAST Searched (See attached)	12/7/2005	AA

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
S1	430	704/208,214.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 10:34
S2	250	S1 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 13:09
S3	26	("4032711" | "4052568" | "4357491" | "4401849" | "4410763" | "4433435" | "4531228" | "4535473" | "4552996" | "4627091" | "4630304" | "4696041" | "4718097" | "4815136" | "5151940" | "5222147" | "5305422" | "5313531" | "5323337" | "5479560" | "5579431" | "5617508" | "5649055" | "6038532" | "6266633" | "RE32172").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2005/12/07 09:49
S4	2	("4052568").PN.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/12/07 10:26
S5	57	("4052568").URPN.	USPAT	OR	OFF	2005/12/07 10:26
S6	430	704/208,214.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/07 10:51
S7	250	S6 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/07 10:51
S8	80	S7 and microphone	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/07 11:00
S9	2243	voiced and unvoiced	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 09:39

S10	1001	S9 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 09:39
S11	397	S10 and microphones	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 09:40
S12	174	S11 and (background adj noise)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 09:41
S13	113	S12 and vary$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S14	22	S10 and (two adj microphones)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 09:58
S15	2	("5590241").PN.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/12/10 10:34
S16	17	("4184049" | "4335276" | "4468804" | "4490839" | "4653102" | "4747143" | "4852169" | "4916743" | "4952931" | "5083310" | "5148488" | "5251263" | "5260896" | "5293425" | "5319736" | "5353376" | "5396554").PN.	US-PGPUB; USPAT; USOCR	OR	OFF	2005/12/10 10:34
S17	11	("5590241").URPN.	USPAT	OR	OFF	2005/12/10 10:43
S18	2243	voiced and unvoiced	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S19	1001	S18 and threshold	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19

S20	397	S19 and microphones	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S21	174	S20 and (background adj noise)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S22	113	S21 and vary$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S23	86	S22 and gain	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 12:19
S24	73	S18 and (first adj threshold) and (second adj threshold)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 13:10
S25	47	S24 and (background adj noise)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/12/10 13:17
S26	2	("5617508").PN.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	OFF	2005/12/10 13:17

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Atty. Docket No. ALPH.P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:	)	
	)	
Gregory C. Burnett	)	Examiner: To be assigned
	)	
Application No. 10/159,770	)	Art Unit: 2644
	)	
Filed: May 30, 2002	)	
	)	
For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS	)	

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

INFORMATION DISCLOSURE STATEMENT UNDER 37 CFR § 1.97

Sir:

Enclosed is an Information Disclosure Citation Form PTO/SB/08. Also enclosed are copies of all references that are not U.S. Patent references. It is respectfully requested that all cited references be considered and that a copy of the enclosed Form PTO/SB/08 be initialed by the Examiner to indicate such consideration and a copy thereof returned to Applicant.

Pursuant to 37 CFR 1.97(h), the submission of this Information Disclosure Statement is not to be construed as an admission that the information cited in this statement is material to patentability.

This Information Disclosure Statement is being submitted pursuant to 37 CFR 1.97(b)(3) before the mailing of a first office action on the merits.

The Commissioner is hereby authorized to charge any fees which may be required in connection with this submission to Deposit Account No. 501914.

Respectfully submitted,

SHEMWELL GREGORY & COURTNEY LLP

Dated: November 4, 2004

Barbara B. Courtney
Registration No. 42,442

OIPE JC14 NOV 05 2004 PATENT & TRADEMARK OFFICE

PTO/SB/08A (08-03)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(Use as many sheets as necessary)

Sheet 1 of 1

Complete if Known	
Application Number	10/159,770
Filing Date	05/30/2002
First Named Inventor	Gregory C. Burnett
Art Unit	2644
Examiner Name	Not yet assigned
Attorney Docket Number	ALPH.P009

U. S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Document Number Number-Kind Code[2] (if known)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
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		EP 0 637 187 A	02-01-1995	Pan Communicati		
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		JP 2001 189 987	07-10-2001	Pioneer Electro		
		WO 02 07151	01-24-2002	ALIPHCOM, Inc.		

Examiner Signature		Date Considered	

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant. [1] Applicant's unique citation designation number (optional). [2] See Kinds Codes of USPTO Patent Documents at www.uspto.gov or MPEP 901.04. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

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Barbara B. Courtney
(Name of person transmitting correspondence)

November 4, 2004
Date

Serial/Patent No.: 10/159,770 Filing/Issue Date: May 30, 2002 Client: ALIPH, Inc.
Title: Detecting Voiced and Unvoiced Speech using Both Acoustic and Nonacoustic Sensors

Atty. Docket No.: ALPH.P009 Date Mailed: November 4, 2004

The following has been received in the U.S. Patent & Trademark Office on the date stamped hereon:

- ☐ Amendment/Response (pgs.)
- ☐ Preliminary Amendment (pgs.)
- ☐ Application - Utility (pgs.)
- ☐ Application - Rule 1.53(b) Contin. (pgs.)
- ☐ Application - Rule 1.53(b) Divis. (pgs.)
- ☐ Application - Rule 1.53(b) CIP (pgs.)
- ☐ Application - Rule 1.53(d) CPA (pgs.)
- ☐ Application - PCT (pgs.)
- ☐ Application - Provisional (pgs.)
- ☐ Drawings (sheets)
- ☐ Declaration & POA (pgs.)
- ☐ Assignment & Cover Sheet (pgs.)
- ☐ Power of Attorney
- ☐ Nonpublication Request (35 USC 122(b))
- ☑ Other Seven (7) cited references
- ☐ Petition for Extension of Time (month(s))
- ☑ Information Disclosure Statement & PTO/SB/08A
- ☑ Information Disclosure Statement Transmittal
- ☐ Submission of Formal Drawings
- ☐ Notice of Appeal
- ☐ Appeal Brief (pgs. in triplicate)
- ☐ Reply Brief
- ☐ Response to Notice of Missing Parts
- ☐ Utility Patent Application Transmittal
- ☐ Fee Transmittal (in duplicate)
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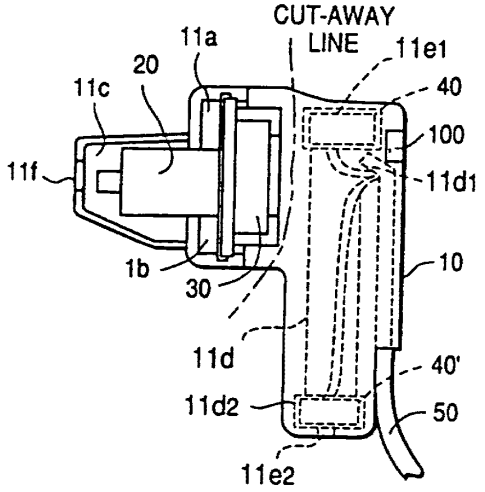
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(54) **Two-way communications earset.**

(57) A one piece two way voice communications earset which is situated in or at the ear of the user and which includes either two unidirectional microphones having their outputs combined or a single bi-directional microphone.

FIG. 4



EP 0 637 187 A1

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BACKGROUND OF THE INVENTION

The present invention relates to an earset for use in two-way voice communications which is worn in or at an ear canal of a user and which is operated hands-free. The earset of the present invention can transmit the user's voice and receive voice signals from another party simultaneously without causing feedback.

Modern wireless technologies have advanced to the extent that portable, wireless and mobile telephones have become small in size and are used widely. Such portable telephones typically include a handset requiring the use of at least one hand for communication.

A disadvantage of the handset type of telephone is that they are inconvenient to use in front of a computer or while driving a car. In such instances, the user must use a hand to hold the handset while typing at the keyboard, which is very difficult to do, or while holding the steering wheel of a car, which is dangerous. Alternatively, the user must hold the telephone handset to their ear with their shoulder, which is very uncomfortable.

In order to solve this drawback, hands-free telephones have been developed. A conventional two-way communications apparatus used with hands-free telephones comprises a headband including a speaker positioned in front of one ear and a boom attached to the headband with a small microphone at one end of the boom. With the boom connected to the headband, the microphone extends in front of the mouth of the user. However, this type of hands-free telephone is not convenient to use because it is cumbersome and because it disturbs the hair of the user.

Another conventional two-way communications system used with hands-free telephones includes a structure which hangs from the ear of the user. This type of communications system includes a speaker positioned in front of the user's ear and a boom having a microphone extending in front of the mouth of the user. Again, such a structure is not convenient for the user because it typically does not hang well at the ear and because it is bulky to carry owing to the boom.

Alternatively, as shown in Fig. 1, yet another conventional two-way communications system includes a receiver section 1 with a speaker 1a which is placed in the user's ear canal. Sound is transmitted to the user via an opening 1b in the back of the speaker 1a. The two way communications system also includes a transmitter section 3 which is located relatively close to the user's mouth. The transmitter section 3 includes a microphone 3a.

In the conventional two-way communications system shown in Fig. 1, however, the transmitter section 3 is attached to the receiver section 1 by a chord 2. The cord 2 also extends past the transmitter section 3 and is connected to a telephone terminal (not shown).

The two-way communications system shown in Fig. 1 does not require hands to operate. However, due to the connection of the transmitter section 3 by the cord 2, the microphone 3a dangles. With such a configuration, the microphone 3a must be omnidirectional causing noise when the microphone 3a rotates at the chord 2. Additionally, with the two-way communications system shown in Fig. 1, the sound intake of the microphone 3a may be blocked by the face or clothing of the user, causing a decrease in the quality of the transmitted voice signals. Moreover, contact between the transmitter section 3 and the clothing of the user tends to cause noise.

Conventional two-way communications systems, such as those shown in Fig. 2 and Fig. 3 attempt to overcome the deficiencies described above with the two-way communications system shown in Fig. 1.

The conventional two-way communications earsets shown in Figs. 2 and 3 integrate the speaker 1a and the microphone 3a into a single case 4. The L-shaped case 4 houses the speaker 1a in one leg and houses the microphone 3a in the other leg. In the embodiment shown in Fig. 2, the speaker 1a has a single opening 4a at a back portion thereof. Alternatively, as shown in Fig. 3, the speaker 1a may have a plurality of holes 4a'. The microphone 3a has an opening 4b through which signals are received. There is a chord 2 coming out from the lower part of the L-shaped case 4 which is connected to a telephone terminal (not shown).

By incorporating the speaker 1a and the microphone 3a in the case 4, conventional two-way communications earsets shown in Figs. 2 and 3 solve many of the problems discussed above related to the communications system shown in Fig. 1. However, the two-way communications earsets of Figs. 2 and 3 cause feedback because sound that leaks out from the opening 4a is fed into the microphone 3a though sound intake hole 4b. Therefore, in the communications earsets shown in Figs. 2 and 3, either the sound volume of the speaker 1a has to be controlled below a certain level or the sensitivity of the microphone 3a has to be kept below a certain level to reduce the effects of feedback.

However, if the sound volume is maintained at a low level, the user may find it difficult to hear voice signals due to the presence of ambient noise. In an effort to reduce the effects of ambient noise, the user may cover his ear with his hand. Unfortunately, this significantly increases the acoustic coupling between the speaker and microphone, causing even more feedback.

In addition, with two-way communications earsets shown in Figs. 2 and 3, voice signals of the user are picked up at the ear, which is about 14 cm away from the user's mouth. Conventional telephone handsets are designed to pick up voice signals directly in front of the user's mouth which are approximately 2.5 cm away from the microphone. At this distance, the level of the voice signal is approximately 94dBSPL. Because the two-way communications earsets shown in Figs. 2 and 3 receive voice signals further away from the mouth than conventional handsets, the signals are approximately 14 dBSPL (sound pressure level) lower than those received by a conventional telephone handset. Because the voice signals are received from further away, they require greater amplification. As a result, the systems shown in Figs. 2 and 3 tend to pick up more ambient noise and room reverberation, both of which make the received voice signals less intelligible.

Accordingly, an object of the present invention is to provide an earset for use in two-way voice communications which is worn in or at an ear canal of the user and which is operated hands-free.

Another object of the present invention is to provide an earset which transmits the user's voice and receives voice signals from another party simultaneously without causing feedback caused by acoustic or mechanical coupling between the speaker and the microphone.

A further object of the present invention is to provide a two-way communications earset which delivers an adequate level of sound to the user without causing feedback, while also delivering an adequate level of sound to a receiver a the other end of the communication system.

A still further object of the present invention is to provide a two-way voice communications earset which is less sensitive to ambient noise at a level almost equal to a telephone handset and to room reverberation so that clear voice communications can be obtained.

Additional objects and advantages of the invention will be set forth in the description which follows, and in part will be obvious from the description, or may be learned by practice of the invention. The objects and advantages of the invention may be realized and obtained by means of the instrumentalities and combinations particularly pointed out in the appended claims.

SUMMARY OF THE INVENTION

To achieve the objects in accordance with the purposes of the present invention, as embodied and described herein, the earset of the present invention comprises a one piece two-way simultaneous voice transmitting and receiving device comprising a receiver for emitting sound primarily to the ear canal of a user and a transmitter, coupled to the receiver, and located outside the ear canal, comprising a first microphone adapted to be located at a first position relative to the mouth of the user and outputting a first signal, a second microphone spaced a distance away from the first microphone and adapted to be located at a second position relative to the mouth of the user further from the mouth of the user than the first position, and outputting a second signal and means for detecting the first and second signals.

The earset of the present invention further comprises a one piece two-way simultaneous voice transmitting and receiving device comprising a receiver, emitting sound primarily to the ear canal of a user, and a transmitter, coupled to the receiver and located outside the ear canal, comprising a microphone with bi-directional directivity having two sound receiving ends, a cavity housing the microphone substantially at a central portion thereof and directed toward the mouth of the user and sound absorbing materials placed with the cavity and abutting the sound receiving ends of the microphone.

BRIEF DESCRIPTION OF THE DRAWINGS

The accompanying drawings, which are incorporated in and constitute a part of the specification, illustrate the presently preferred apparatus of the present invention and, together with the general description given above and the detailed description of the preferred embodiments given below serve to explain the principles of the invention. In the drawings:

Fig. 1 is a sectional view of a conventional earset with a microphone attached to a cord;

Fig. 2 is a side sectional view of a conventional one piece earset adapted to be worn in a user's ear;

Fig. 3 is a side sectional view of another conventional one piece earset adapted to be worn in a user's ear;

Fig. 4 is a sectional view of a two-way communications earset according to a first embodiment of the present invention;

Fig. 5 is a side view of a second embodiment of a two-way communications earset according to the present invention placed in an ear of a user;

Fig. 6 is a sectional view of a two-way communications system according to a third embodiment of the present invention;

Fig. 7 is a side view showing the relative positioning of a sound source relative to the earset shown in Fig. 4;

Fig. 8 is a side view showing the relative positioning of a mouth relative to the earset shown

in Fig. 4;
Fig. 9 is a side view showing the relative positioning of a speaker output relative to the earset shown in Fig. 6;
Fig. 10 is a side view showing the relative positioning of a sound source with respect to an alternative embodiment of the two-piece microphone;
Fig. 11 is a circuit diagram of conventional circuit that produces an electret microphone output;
Fig. 12 is a circuit diagram of circuitry used in the earset of the present invention for subtracting the outputs of two microphones when the microphones are in phase;
Fig. 13 is a circuit diagram of circuitry used in the earset of the present invention for adding the outputs of two microphones when the microphone outputs are out of phase;
Fig. 14 is a circuit diagram of circuitry used in the earset of the present invention for adding the outputs of two microphones when the microphones are mutually out of phase;
Fig. 15 is a side view of a two-way communications earset according to a fourth embodiment of the present invention;
Fig. 16 is a side sectional view of the earset according to the fourth embodiment of the present invention hung on the user's ear;
Fig. 17 is a sectional view of a two-way communications earset according to the fifth embodiment of the present invention;
Fig. 18 is a side view of the earset according to the fifth embodiment inserted in a user's ear canal;
Fig. 19 is a side view of the earset according to the fifth embodiment with an alternative ear insertion shape;
Fig. 20 is a back view of a two-way communications earset according to the sixth embodiment of the present invention;
Fig. 21 is a side sectional view of the earset according to the sixth embodiment hung on an ear of a user;
Fig. 22 is a block diagram of a circuit used in the earset of the present invention for reducing the degree of amplification of sound to less than 70 dBSPL; and
Fig. 23 is a graphical representation of the input-output characteristics of the circuit shown in Fig. 22.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Fig. 4 is a sectional view of a two-way communications earset according to a first embodiment of the present invention. The embodiment shown in Fig. 4 comprises an earset 10 which has an L-shaped configuration. The earset 10 is preferably made of plastic and includes an enlarged portion 11a which is inserted in the ear of the user. The enlarged portion 11a includes an earset component portion 11b and an insertion portion 11c.

The enlarged portion 11a is placed in the user's ear auricle and the insertion portion 11c is inserted into the ear canal. The insertion portion 11c is preferably shaped so that the ear canal of the user will not be completely clogged by the earset 10 and so that the earset 10 will not be so loose as to cause feedback due to leaked sound.

A speaker 20 is housed in the earset component portion 11b and is supported by resilient material 30. The output of the speaker 20 is input to the insertion part 11c and is fed into the ear canal of the user through hole 11f. The speaker 20 can preferably be either a magnetic-type speaker or dynamic-type speaker. As shown in Fig. 1, the speaker 20 is situated in the earset component portion 11b. However, in accordance with the present invention, the speaker 20 can be placed in an alternate location, so long as the output from the speaker 20 is guided into the ear canal of the user.

The earset 10 also includes a cylindrical cavity 11d. At the ends of the cavity 11d are housing spaces 11d1 and 11d2. The housing spaces 11d1 and 11d2 are connected to the outside of the earset 10 through two respective holes 1e1 and 1e2. Two electret pieces 40 and 40' are respectively housed in the two housing spaces 11d1 and 11d2. Each of the electret pieces 40 and 40' is preferably a small and sensitive device.

Additionally, in a preferred embodiment of the present invention, the electret pieces 40 and 40' are preferably back-type electret so that they are immune to vibrations emanating from the earset 10 and from extraneous noise from the wires connecting the elements of the earset 10. Such wires may include the wire 50, which is substantially parallel to the cylindrical cavity 11d, and is led to the outside of the earset 10 and the wires coming from speaker 20 and electret pieces 40 and 40'.

The earset 10 shown in Fig. 4 also preferably includes a combination circuit 100 which combines the outputs of the electret pieces 40 and 40'. The combination circuit is explained in more detail below, but preferably is coupled to the output wires of the electret pieces 40 and 40' and has its output coupled to wire 50. As shown in Fig. 4, the combination circuit 100 is located within the cylindrical portion 11d. However, in accordance with the present invention, the combination circuit 100 may be located at any convenient place in the earset 10, so long as the outputs of the electret pieces 40 and 40' are input thereto.

Fig. 5 shows an earset according to a second embodiment of the present invention. As shown in Fig. 5, the earset has a two-piece construction including a two-way wireless communication device 52 connected to an earset 51. The communication device 52 includes a transmitter/receiver at the front portion of the ear lobe. The communication device 52 also includes an integrated circuit (not shown) that transmits voice signals from electret pieces 40 and 40' to a station and receives voice signals from the station, and a combination circuit 100 that combines the outputs of electret pieces 40 and 40'.

The communication device 52 also includes a battery cell located at the back portion of the ear lobe. The battery cell and the transmitter/receiver remain connected to the ear lobe by a pinching mechanism located therebetween. The earset 51, shown in Fig. 5, is substantially the same as that shown in Fig. 4.

Fig. 6 is a side sectional view of an earset according to a third embodiment of the present invention. The earset 60 includes a cylindrical cavity 61d which preferably has a slightly folded cylindrical shape. With such a shape, the lower electret piece 40' is situated closer the user's mouth than the upper electret piece 40. With such a configuration, the lower microphone 40' picks up more of the user's voice signals that does the upper microphone 40 and therefore the users voice can be amplified to a greater extent than when the earset shown in Fig. 4 is used.

The operation of the earset according to any of the embodiments shown in Figs. 4-6 will be described below, specifically with respect to Fig. 4. The enlarged portion 11a of the earset 10 is inserted into the user's ear auricle and the insertion part 11c is inserted into the ear canal. The shape of the insertion part 11c is formed so that the insertion part 11c will not clog the ear canal. With such a shape, the user can hear sound coming from the outside and therefore finds the earset 10 safe to use.

Additionally, the shape of the insertion part 11c is formed so that the insertion part 11c will not leak much sound. As shown in Fig. 4, the speaker 20 is preferably situated in the earset component portion 11b. However, the speaker 20 can be located in other places within the earset 10 as long as the output of the speaker 20 is output to the user's ear canal.

With the earset 10 situated in the user's ear, the user can hear sound from the speaker 20 that is sent via the wire 50 from a duplex transmitting and receiving terminal (not shown). The output of the speaker 2 is led to the ear canal via a hole 11f of the insertion part 11c.

The voice signal from the user reaches the earset 10 via holes 11e1 and 11e2, and then respectively reaches the electret pieces 40 and 40'. The voice signals are converted into electrical signals by electret pieces 40 and 40' and are sent after being combined by the combination circuit 100 via the wire 50 to the duplex voice transmitting and receiving terminal (not shown).

The earsets according to the embodiments shown in Figs. 4-6 have a higher sensitivity to the voice signal of the user than to noise signals coming from places distant from to the user's mouth. This is because the lower electret piece 40' is located closer to the user's mouth than the upper electret piece 40. Thus, when the combination circuit 100 combines the outputs of the electret pieces 40 and 40', the voice signal of the user is output and noise signals from distant sources are nulled.

More specifically, if the output signals of the electret pieces 40 and 40' are in phase relative to the voice of the wearer, the combination circuit 100 preferably subtracts the output signals. If the output signals of the electret pieces are out of phase relative to the voice of the wearer, the circuit 100 preferably adds the outputs. However, regardless of whether the outputs are in phase or out of phase, in order to achieve an effective cancellation of noise, preferably the electret pieces 40 and 40' are identical in shape and electrical performance characteristics.

The noise cancellation which is achieved by the earset according to the present invention is illustrated with respect to Fig. 7. Fig. 7 is a side view showing the relative positioning of a sound source relative to the earset 10 shown in Fig. 4. This explanation assumes that the two electret pieces 40 and 40' have identical electrical performance characteristics and the circuit 100 subtracts the outputs of the electret pieces 40 and 40'.

Typically, a sound source, having sound pressure P, transmits sound through the air and a microphone a distance L from the sound source receives the sound, producing an electrical output of V volts. The relation of P, V and L is as follows:

$$V = k\,P/L \quad \text{(k is a constant related to a microphone)} \qquad (1)$$

In the earset according to the present invention including electret pieces 40 and 40', two voltages V and V' are produced. Additionally, the upper electret piece 40 is located a distance L1 from the sound source while the lower electret piece is located a distance L2 from the sound source P. According to equation (1) above, the difference between the outputs V and V' respectively from the electret pieces 40 and 40', the distances L1 and L2

from the sound source P and the microphone constants k and k' have the following relationship:

$$Vo = V1 - V2 = (k1\ P/L1) - (k2\ P/L2) \quad (2)$$

Assuming the two electret pieces 40 and 40' are identical in electrical performance such that k1 = k2 = k, the formula (2) can be simplified as follows:

$$Vo = V1 - V2 = L1\ (1 - (L1/L2)) \quad (3)$$

If the distance between the electret pieces 40 and 40' is L3, and L1 and L2 are 10 times larger than L3, formula (3) shows that as L1 and L2 become almost equal, Vo becomes zero. In other words, as the distance to the noise source P increases, the difference between the outputs of the electret pieces 40 and 40' also approaches zero, so that no signal will be output from the earset 10 as a result of such a noise source P. Since L3 is designed to be around 5 cm, noise sources with a distance of 50 cm will not cause an produce an output from the earset.

Fig. 8 shows the distance relationship between the earset 10 and the mouth M of the user. Typically, the distance between the mouth and the entrance to an ear canal is approximately 14 cm. As shown in Fig. 8, the distance between the mouth M and the upper electret piece 40 is designated as L2, while the distance between the lower electret piece 40' and the user's mouth M is designated as L1. Because the distance L3 between electret pieces 40 and 40' is approximately 5 cm, the L1/l2 relationship becomes less than 1 in equation (3). As a result, a significant output is produced by the difference between the outputs of the electret pieces 40 and 40'. This output is primarily due to the voice signals of the user and is not due to ambient noise signals, which are effectively cancelled.

In the third embodiment of the present invention shown in Fig. 6, the lower electret piece 40' is preferably located closer to the user's mouth. With such a configuration, the ratio L1/L2 becomes even smaller because the distance between the user's mouth M and the lower electret piece 40' is decreased. As a result, the output signal Vo is greater than the output signal when using the first embodiment of the present invention shown in Fig. 4.

Using a combination of electret pieces 40 and 40', as shown in the first through third embodiments of the present invention, and the combination circuit 100 also helps to reduce feedback of the earset. As shown in Fig. 9, under this circumstance, the sound source P is the output of the speaker 20. As long as the distances between the speaker 20 and the electret pieces 40 and 40', shown in Fig. 9 as L1 and L2, respectively, are nearly equal, the output Vo, as evidenced from equation (3), becomes small, thus reducing feedback.

The electret pieces 40 and 40' used in the earset according to the present invention can be an omni-directional type, a unidirectional type or bi-directional type. Unidirectional type electret pieces produce more directionality toward the user's mouth than do omni-directional type electret pieces and therefore may be preferable to omni-directional type microphones.

Fig. 10 is a side sectional view of a dual microphone structure using two unidirectional microphones. As shown in Fig. 10, the lower electret piece 104' is directly opposite the upper electret piece 104 in the cylindrical cavity 101. The two electret pieces 104 and 104' are placed back to back at the ends of the cavity 101 so that directionality of the two electret pieces 40 and 40' is opposite. Therefore, when the earset with the unidirectional microphones shown in Fig. 10 is placed in a user's ear, the directionality of the lower electret piece 104' is toward the user's mouth while the directionality of the upper electret piece 104 is in a direction opposite to the mouth. More specifically, the directionality of the electret pieces 104 and 104' together form a figure eight, with a null point in the center.

In operation, the unidirectional electret pieces 104 and 104' require an input at the respective back surfaces 104a and 104a' of the electret pieces 104 and 104'. This input is provided by a hole 102 in the cylindrical cavity 101, Preferably, the hole 102 is placed at the null point of the directionality of the unidirectional microphones 104 and 104', that is, in the center of the cylindrical cavity 102. More specifically, the hole 102 should be placed such that the distances to the respective back surfaces 104a and 104a' of the unidirectional microphones 104 and 104' are equal. As shown in Fig. 10, the distances to the back surfaces 104a and 104a' are Lp1 and Lp2. The distance from the sound source P to the hole 102 is Lp. In a preferred embodiment of the present invention, the distances from the sound source P to the back surfaces 104a and 104a' are equal so that the combination circuit 100, not shown in Fig. 10, can effectively reduce noise.

As well as being an input for the unidirectional microphones 104 and 104', the hole 102 can preferably also be used as an outlet for wires connecting the electret pieces 104 and 104' and the wire 50, shown in Fig. 4.

The combination circuit 100, which combines the outputs from the electret pieces 4 and 4', shown in Figs. 4 and 7-9, 64 and 64', shown in Fig. 6 and 104 and 104', shown in Fig. 10, is explained

below with respect to Figs. 11-14.

Fig. 11 is a circuit diagram of a conventional output circuit of a single electret microphone. As shown in Fig. 11, a positive direct voltage Vcc, a load resister RL and direct current cut-off capacitor C are connected to the microphone output terminal 111. An alternating voltage v appears as the output with respect to the microphone reference terminal 112. With the circuit of Fig. 11, only one output from a single microphone is produced.

However, in accordance with the present invention, the earset can have two microphones, the outputs of which must be combined. Fig. 12 shows a circuit diagram of a first combination circuit 100a to be used with the earset of the present invention. The combination circuit 100a, shown in Fig. 12 is similar to the circuit shown in Fig. 11 in that it includes direct voltage Vcc, load resistances RL1 and RL2, respectively of the microphones 40 and 40' and respective current cut-off capacitances C1 and C2.

The combination circuit 100a shown in Fig. 12 further includes a subtraction circuit S which outputs the difference between the outputs of the microphones 40 and 40', which are input, respectively at inputs 121 and 123. The combination circuit shown in Fig. 12 is used when the microphone elements are in phase. Reference voltages for the microphones 40 and 40' are provided at terminals 122 and 124. The two output signals of the microphones are respectively represented by v1 and v2.

The subtraction circuit S comprises an operational amplifier OP1 and resistors, RA1, RA2, RA3 and RA4. The operational amplifier and the resistors are configured such that the subtraction circuit S has no gain relative to the input signals. As a result, the output of the operational amplifier OP1 is Vo = v1 - v2.

In the combination circuit 100a shown in Fig. 12, the resistance of RA1 is preferably more than 20 times greater than the resistances of both RL1 and RL2. The capacitances of C1 and C2 are such that signals with frequencies higher than 50 Hz are passed. Moreover, if the outputs of the microphones which are input to the combination circuit 100a have equal sensitivities in their electrical performance, RL1 is preferably equal to RL2. However, if the microphones have different sensitivities, the values of the resistances RL1 and RL2 are preferably different so that such difference in the microphone sensitivities can be nulled.

Fig. 13 is a circuit diagram of a combination circuit to be used in an earset according to the present invention when the outputs of the microphones are out of phase. Like reference numerals in the combination circuit 100a shown in Fig. 12 are used in Fig. 13.

With the combination circuit 100b of Fig. 13, operational amplifiers OP2 and OP3 are used in combination with resistors RB1 through PRB5. The resistors are configured such that no gain is achieved with respect to the input signals. Thus, the addition circuit produces Vo = v1 + v2.

It is noted that using the combination circuits 100a and 100b, shown respectively in Figs. 12 and 13, differences in electrical performance of the microphones 40 and 40' can be compensated for by adjusting the values of the resistors. It is noted that in order to allow for continuous adjustments either one of the resistors RL1 and RL2 shown in Figs. 12 and 13 can be replaced by a variable resistor.

For a perfectly matched pair of microphones 40 and 40' that are perfectly in phase, the combination circuit 100c shown in Fig. 14 can be employed. The combination circuit 100c shown in Fig. 14 is an idealized version of that shown in Fig. 12, assuming that the microphone outputs appearing at terminals 121 and 123 are identical in electrical characteristics so that adjustments made via the subtraction circuit S are unnecessary. With the combination circuit 100c, no adjustments for differences in sensitivity of the microphones 40 and 40' can be made and the output Vo is simply the difference between the outputs of the microphones 40 and 40'. Using the combination circuit 100c shown Fig. 14 the number of wires coming out from the earset can be reduced.

Fig. 15 shows a back view of a two-way communications earset according to a fourth embodiment of the present invention. The earset 150, shown in Fig. 15, hangs at the ear of the user, instead of being inserted into the user's ear canal, as in the first through third embodiments of the present invention. The earset 150 includes an ear cover 157 having an arc shaped portion 157a that hangs at the base of the user's ear. Inside the ear cover 157 is a speaker 152, similar to the speaker 20 shown in Fig. 4.

Preferably, the ear cover 157 is constructed of an electrically-conductive plastic, which discharges static electricity that a user may pick up before inserting the earset 150 into their ear. If the ear cover is not constructed of such a material, the user may get a shock upon inserting the earset 150 into his ear.

Fig. 16 is a sectional view of the two-way communications earset according to the fifth embodiment of the present invention shown in Fig. 15 hung from a user's ear. As shown in Fig. 16, outside the ear cover 157 are the electret pieces 40 and 40' situated in cylindrical cavity 168, similar to those shown in Fig. 6. Additionally, there is a hole 168a in the cavity 168, which holds the electret pieces 40 and 40', which corresponds to the

hole 102, shown in Fig. 10. The embodiment shown in Figs. 15 and 16 is easy to wear since it hangs at the ear of the user. Additionally, because the ear cover 157 shields sound from the speaker 152, there is little feedback between the microphones 40 and 40' and the speaker 152.

An earset according to the sixth embodiment of the present invention is shown in Figs. 17 and 18. In the earset shown in Figs, 17 and 18, the two microphones, shown in Figs. 4-10 and 15-16, are replaced by a small single electret microphone 179 with bi-directional directivity. The bi-directional microphone 179 is located in the middle of a cylindrical cavity 171d, which is preferably similar to one of the cylindrical cavities shown in Figs. 4 and 6. Between the sound intake holes 171e1 and 171e2 at either end of the cylindrical cavity 171d is sound absorbing materials 172. Preferably, the sound absorbing materials 172 are fibres or cotton filling which fill the spaces on both sides of the microphone 179.

Preferably, in the embodiment shown in Fig. 17, the total length of the cylindrical cavity 171d in which the microphone 179 is mounted should be less than 10 cm owing to the stability required for wearing the earset 170 in the user's ear. Accordingly, the microphone 179 is preferably mounted less than 5 cm from each end of the cylindrical cavity 171d.

It is noted, however, that the earset 170, shown in Fig. 17, can alternatively have the structure shown in Fig, 4, so long as the two electret microphones 40 and 40' are replaced by a single bi-directional microphone 179. Additionally, as with the embodiments shown in Fig. 4-6, the location of the speaker 20 in Fig. 17 can be changed so long as the output of the speaker 20 is input to the ear canal of the user. The remainder of the earset 170 is the same as that shown in Fig. 6 and like reference numerals identify like elements.

Fig. 18 is a side view of the embodiment shown in Fig. 17 inserted in an ear canal C. As shown in Fig. 18, the insertion part 61c fits snugly in the ear canal C, while the enlarged portion 61a abuts the ear auricle A. Alternatively, the earset 170 can have an ear insertion configuration such as that shown in Fig. 19. Fig. 19 shows a modification of the earset 170 which includes an ear insertion portion 191c and an enlarged portion 191a. The ear insertion portion 191c does not fit snugly in the ear canal, but rather is placed at the entrance of the ear canal C. Additionally, in the modification shown in Fig. 19, the enlarged portion 191a surrounds a portion of the ear auricle A and maintains the earset 170 at the user's ear.

It should also be noted that the earsets according to the first through third embodiments of the present invention can include two bi-directional microphones 179 similar to that shown and described with respect to Figs. 17-19. Using two bi-directional microphones in lieu of two unidirectional microphones 104 and 104' may be preferable because of the increased directivity achieved with bi-directional microphones.

An earset according to a sixth embodiment of the present invention is described with respect to Figs. 20 and 21. The earset 200, shown in Figs. 20 and 21 is shaped similar to the embodiment shown in Figs. 15 and 16, but includes a single bi-directional microphone, as shown in the embodiment shown and described with respect to Figs. 17-19. Like reference numerals from those figures are used in Figs 20 and 21.

As shown in Fig. 20, the earset 200 preferably includes a sound absorbing or shielding material 201 surrounding the speaker 152. Preferably, the sound absorbing or shielding material 201 is made of sponge and is ring-shaped surrounding the speaker 152. The sound absorbing or shielding material 201 prevents sound from leaking out of the space enclosed by the ear canal, the ear auricle and the ear cover 157.

Fig. 18 is a side sectional view of the earset 200 hung on the user's ear. As shown in Fig. 21, the bidirectional microphone 179 is placed within a cylindrical pipe 168. The cylindrical pipe 168 includes openings 202 and 202a which input signals to the bidirectional microphone 179 through the sound absorbing material 172. As with the embodiment shown in Fig. 16, signals are output through wire 50.

In the embodiments shown in Figs. 17-21, using a single bi-directional microphone 179, the leakage of the sound from the speaker 20 is prevented because such leaked sound, if any, travels to the bi-directional microphone 179 at a direction in which the microphone sensitivity is not high. This is because the axis of directionality of the bidirectional microphone 179 is perpendicular to the direction of any leaked sound from the speaker 20. As a result, feedback between the bidirectional microphone 179 and the speaker 20 is less likely to occur, as compared with conventional earsets.

Additionally, the earsets shown in the embodiments of the present invention have a cylindrical cavity 11d, 168, and 171d. Preferably, the length of the cylindrical cavity in each of the embodiments of the present invention is less than 10 cm. In the embodiments shown in Figs. 17-21, preferably the length of the cylindrical cavity below the bi-directional microphone 179 is less than 5 cm and is directed toward the user's mouth. Such a structure effectively shortens the distance from the mouth to the microphone 179, since the sound reaches the opening in the lower end of the cylindrical cavity pipe 1e2 without diffusion. Additionally, the sound

absorbing material 172 within the cylindrical cavity 171d, shown in Figs. 17-19, or 168, shown in Fig. 21, limits feedback between the speaker 20 and the bi-directional microphone 179 because the sound absorbing material 171 absorbs higher frequency signals, which are the major cause of such feedback.

The earsets according to the embodiments shown in Figs. 17-21 are less sensitive to ambient noise, because ambient noise under normal circumstances tends to come from the horizontal plane. However, when placed in a user's ear, the bi-directional microphone 179 has its virtually on a vertical plane. So oriented, ambient noise impinges on the bi-directional microphone 179 from its least sensitive direction.

Additionally, as explained above with respect to the dual microphone embodiments, typically, sources of ambient noise tend to be more than 10 times the distance of that between the user's mouth and the bi-directional microphone 179. As explained in detail with respect to Fig. 7, the result of this distance is that the microphone will not output signals responsive to distant noise sources, but will primarily output voice signals of the user.

It is noted, however, that a small amount of ambient noise may still be heard using the earsets of the present invention. This is particularly true when the user stops talking and only ambient noise signals are picked up. In order to reduce such ambient noise, the circuitry shown in Fig. 22 can be used in combination with any of the earset embodiments of the present invention.

Fig. 22 is a block diagram of a circuit used in any one of the earsets of the present invention for reducing the degree of amplification of signals of less than 70 dBSPL. Typically, people use telephones at a noise level of less than 70 dBSPL while the sound level to be picked up at the ear is approximately 80 dBSPL. An ordinary telephone assumes 94 dBSPL at a distance of 2.5 cm in front of the user's mouth. The circuit shown in Fig. 22 amplifies signals with a sound level of less than 70 dBSPL much less than signals with a sound level of 70 dBSPL or greater. Accordingly, ambient noise signals which typically have a sound level of less than 70dBSPL will be amplified less than the voice signals of the user. Thus, ambient noise signals will be reduced.

The circuitry 220 shown in Fig. 22 includes a signal level detection circuit 2211 which receives an input signal at terminal 221. The input signal at terminal 221 is preferably either the output from one of the combination circuits shown in Figs. 12-14, when a dual microphone earset is used, or the output from the bidirectional microphone 179, when such an embodiment of the earset is used. It is noted that the circuitry 220 can be placed at any convenient location within the earsets of the present invention so long as the input terminal 221 is provided with the output of either the combination circuit or the bi-directional circuit 179 and the output of the circuit 220 is led to the output wire.

The circuitry 220 further includes a logarithmic amplifier 2212, which amplifies the input signal. A non-linear signal generating circuit 2213, is coupled between the signal level detection circuit 2211 and the logarithmic amplifier 2212 and generates a control signal, depending on output signal level from the signal level detection circuit 2211.

The logarithmic amplifier 2212 is coupled to a dividing circuit 2214, which is common in typical telephones and divides the output from the logarithmic amplifier 12. Specifically, the output from the logarithmic amplifier 2212 is output to an amplifier 2215, which feeds signals to the speaker 20. The dividing circuit 2214 also outputs the signal from the logarithmic amplifier 2212 to a line 2216 which is connected to a telephone exchange. It is noted that line 2216 shown in Fig. 22 may be incorporated in line 50 shown in the embodiments of the earset of the present invention.

The circuit shown in Fig. 22 operates such that input at terminal 221 is fed into the signal level detection circuit 2211. The signal level detection circuit 2211 produces a control signal equal to the level of the input sound, which is then fed to the control signal generating circuit 2213. The control signal generating circuit 2213 generates a lower control signal when the input signal is less than 70 dBSPL and generates a higher control signal when the input signal is higher than 70 dBSPL.

This output is shown in Fig. 23, which shows a logarithmic plot of the relationship between output signals of the signal level detection circuit 2211 and the output signals of the non-linear signal generating circuit 2213. In Fig. 23, the output of the signal level detection circuit 2211 is shown on the horizontal axis while the output of the non-linear signal generating circuit 2213 is shown on the vertical axis.

The graph of Fig. 23 shows that the output of the control signal generating circuit 2213 controls the degree of the amplification of the logarithmic amplifier 2212. As a result, when ambient noise is low, the amplification of the amplifier 2212 becomes low, so that a small amount of sound is transmitted through the dividing circuit 2214 to the outside line 2216.

Additionally, there is a small amount of sound leaked to the amplifier 2215, resulting in a small amount of sound being leaked to the speaker 20. A further benefit of the circuit shown in Fig. 22 is that room reverberation is diminished because of the lower amplification of reverberated sound.

It is noted that the circuit shown in Fig. 22 can be applied to embodiments shown in Figs. 4, 5 and 15 as well as to the embodiments shown in Figs. 17-19. In any of the embodiments of the earset according to the present invention, the circuit shown in Fig. 22 preferably eliminates any weakness in the performance of the device owing to the distance between earset and the user's mouth. This is because by using the circuit shown in Fig. 22 in combination with the earset of the present invention, ambient noise is reduced and reverberated voice signals are eliminated.

The above described embodiments of the present invention implicitly refer to public wired telephone systems. However, it is to be understood that the embodiments of the present invention can be applied to any two-way simultaneous voice communication system such as wireless, portable, mobile and private systems.

Other embodiments of the invention will be apparent to those skilled in the art from consideration of the specification and practice of the invention disclosed herein. It is intended that the specification and examples be considered as exemplary only, with the true scope and spirit of the invention being indicated by the following claims.

Claims

1. A one piece two-way simultaneous voice transmitting and receiving device comprising:
   receiver means for emitting sound to the ear canal of a user;
   transmitter means, coupled to the receiver means, and located outside the ear canal, comprising:
   a first microphone for sound pickup adapted to be located at a first position relative to the mouth of the user and outputting a first signal; and
   a second microphone for sound pickup spaced a distance away from the first microphone and adapted to be located at a second position relative to the mouth of the user further from the mouth of the user than the first position, and outputting a second signal; and
   means for detecting the first and second signals.

2. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the first and second microphones are bi-directional microphones.

3. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the first and second microphones are unidirectional microphones.

4. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the means for detecting comprises a combination circuit for subtracting the first and second signals.

5. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the means for detecting comprises a combination circuit for adding the first and second signals.

6. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the device is adapted to fit within the ear canal of the user.

7. The one piece two-way simultaneous voice transmitting and receiving device according to claim 1, wherein the device is adapted to hang at the ear of the user.

8. A one piece two-way simultaneous voice transmitting and receiving device comprising:
   a receiver emitting sound to the ear canal of a user; and
   a transmitter, coupled to the receiver and located outside the ear canal, comprising:
   a microphone for sound pickup and bi-directional directivity having two sound receiving ends;
   a cavity housing the microphone substantially at a central portion thereof and directed toward the mouth of the user; and
   sound absorbing materials placed within the cavity and abutting the sound receiving ends of the microphone.

9. The one piece two-way simultaneous voice transmitting and receiving device according to claim 8, wherein the cavity comprises a sound pipe which extends less than 5 cm from each of the sound receiving ends of the microphone.

10. The one piece two-way simultaneous voice transmitting and receiving device according to claim 8, wherein the device is adapted to fit within the ear canal of the user.

11. The one piece two-way simultaneous voice transmitting and receiving device according to claim 8, wherein the device is adapted to fit hang at the ear of the user.

12. A one piece simultaneous voice transmitting and receiving device comprising:
   a receiver for emitting sound primarily to the ear canal of a user;

a transmitter, coupled to the receiver, and adapted to be located outside the ear canal comprising:

a first microphone for sound pickup adapted to be located at a first position relative to the mouth of the user and outputting a first signal; and

a second microphone for sound pickup spaced from the first microphone and adapted to be located at a second position relative to the mouth of the user further from the mouth of the user than the first position and outputting a second signal;

subtracting means, coupled to the first and second microphones, for subtracting the first and second signals and for outputting a third signal with a degree of amplification; and

reducing means, coupled to the subtracting means, for reducing the degree of amplification of the third signal if a sound level in the area of the ear canal is less than 70 dBSPL to a greater extent than when the sound level is greater than 70 dBSPL.

13. The one piece simultaneous voice transmitting and receiving device according to claim 12, wherein the reducing means comprises:

a signal level detection circuit which receives the third signal from an input terminal and which detects the level of the third signal;

a signal generating circuit, coupled to the signal level detection circuit, for outputting a control signal based on the level detected by the signal level detection circuit;

a logarithmic amplifier, coupled to the signal generating circuit, which receives the third signal and amplifies the third signal based on the control signal output from the signal generating circuit;

a dividing circuit, coupled to the logarithmic amplifier, which divides the amplified third signal;

an amplifier, coupled to the dividing circuit, which amplifies a first output of the dividing circuit and outputs an amplified signal to the receiver; and

an output line, coupled to the dividing circuit, for outputting a second output of the dividing circuit to the transmitter.

14. A one piece two-way simultaneous voice transmitting and receiving device comprising:

a receiver section for emitting sound primarily to the ear canal of a user; and

a transmitter section, coupled to the receiver section, located outside the ear canal comprising:

a microphone for sound pickup with bi-directional directivity having two sound receiving ends and for outputting a signal; and

a cavity housing the microphone substantially at a central portion thereof and directed toward the mouth of the user;

sound absorbing materials placed within the cavity and abutting the sound receiving ends of the microphone; and

reducing means, coupled to the microphone, for reducing the degree of amplification of the signal if a sound level in the area of the ear canal is less than 70 dBSPL to a greater extent than when the sound level is greater than 70 dBSPL.

15. A two-way simultaneous voice transmitting and receiving device comprising:

an ear cover adapted to hang at the auricle of a user's ear;

a speaker attached to an inside portion of the ear cover and facing the entrance of the user's ear canal;

a transmitter attached to an outside portion of the ear cover and having a first microphone for sound pickup adapted to be located at a first position relative to the mouth of the user and outputting a first signal and a second microphone for sound pickup spaced from the first microphone and adapted to be located at a second position relative to the mouth of the user that is further away from the mouth of the user than the first position and outputting a second signal; and

a combination circuit, coupled to the transmitter, which subtracts the first and second signals.

16. A two-way simultaneous voice transmitting and receiving device comprising:

an ear cover adapted to hang at the auricle of a user's ear;

a speaker attached to an inside portion of the ear cover and facing the entrance of the user's ear canal;

a transmitter section adapted to be located outside the user's ear canal, comprising:

a microphone for sound pickup with bi-directional directivity having two sound receiving ends and for outputting a signal; and

a cavity housing the microphone substantially at a central portion thereof and directed toward the mouth of the user; and

sound absorbing materials placed within the cavity and abutting the sound receiving ends of the microphone.

17. A two-way simultaneous voice transmitting and receiving device comprising:

an ear cover adapted to hang at the auricle of a user's ear;

a speaker attached to an inside portion of the ear cover and facing the entrance of the user's ear canal;

a transmitter, coupled to the receiver, and adapted to be located outside the ear canal comprising:

a first microphone for sound pickup adapted to be located at a first position relative to the mouth of the user and outputting a first signal; and

a second microphone for sound pickup spaced from the first microphone and adapted to be located at a second position relative to the mouth of the user further from the mouth of the user than the first position and outputting a second signal;

subtracting means, coupled to the first and second microphones, for subtracting the first and second signals and for outputting a third signal with a degree of amplification; and

reducing means, coupled to the subtracting means, for reducing the degree of amplification of the third signal if a sound level in the area of the ear canal is less than 70 dBSPL to a greater extent than when the sound level is greater than 70 dBSPL.

18. A two-way simultaneous voice transmitting and receiving system comprising:

an ear cover adapted to hang at the auricle of a user's ear;

a speaker attached to an inside portion of the ear cover and facing the entrance of the user's ear canal;

a transmitter section, coupled to the receiver section, located outside the ear canal comprising:

a microphone for sound pickup with bi-directional directivity having two sound receiving ends and for outputting a signal;

a cavity housing the microphone substantially at a central portion thereof and directed toward the mouth of the user; and

sound absorbing materials placed within the cavity and abutting the sound receiving ends of the microphone; and

reducing means, coupled to the microphone, for reducing the degree of amplification of the signal if a sound level in the area of the ear canal is less than 70 dBSPL to a greater extent than when the sound level is greater than 70 dBSPL.

FIG. 1
PRIOR ART

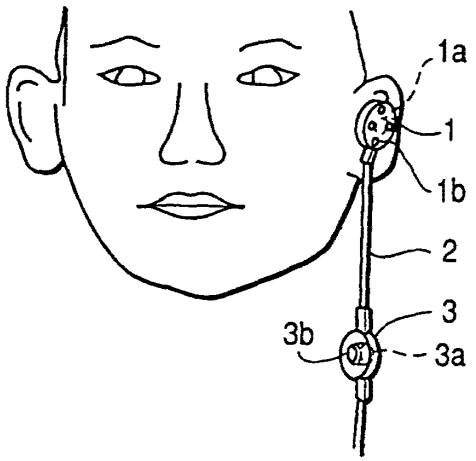


FIG. 2
PRIOR ART

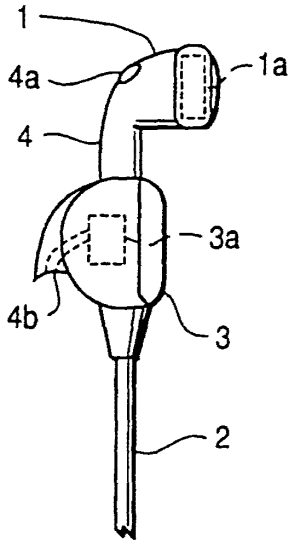


FIG. 3
PRIOR ART

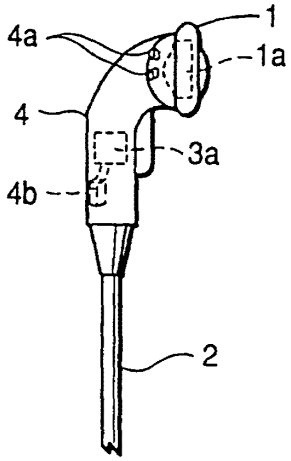


FIG. 4

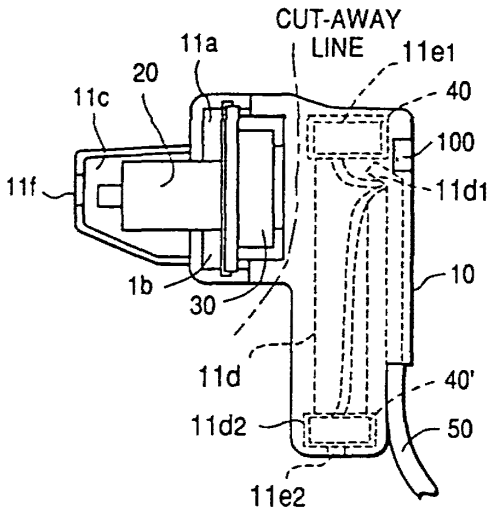


FIG. 5

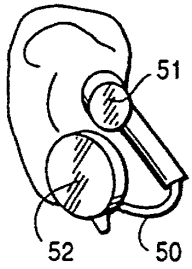


FIG. 6

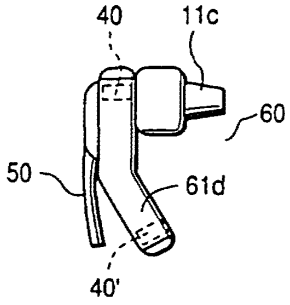


FIG. 7

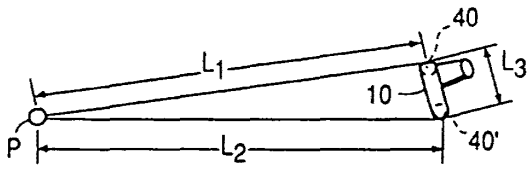


FIG. 8

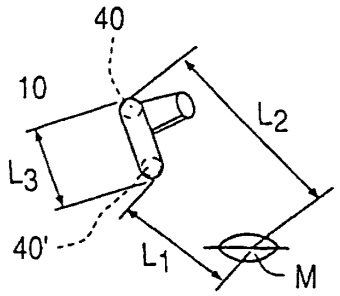


FIG. 9

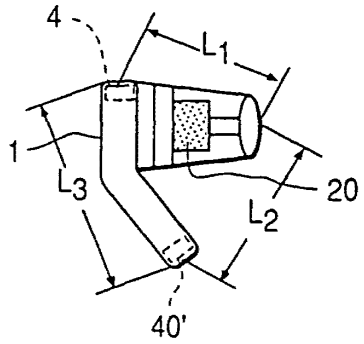


FIG. 10

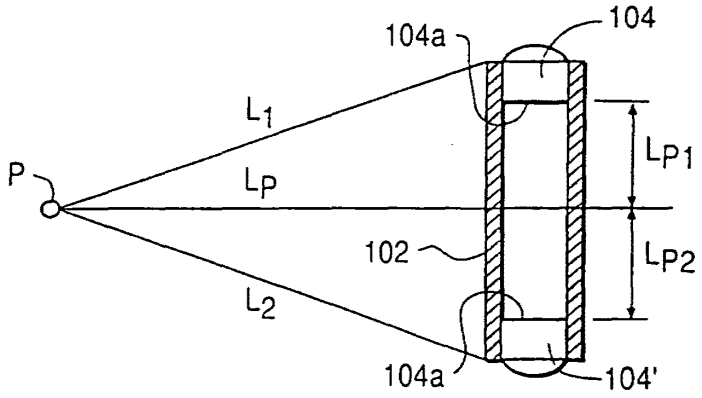


FIG. 11
PRIOR ART

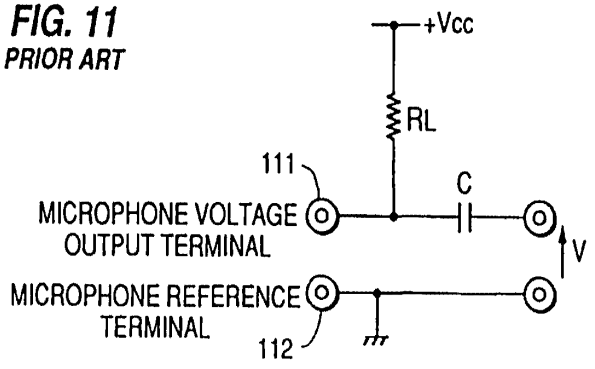


FIG. 12

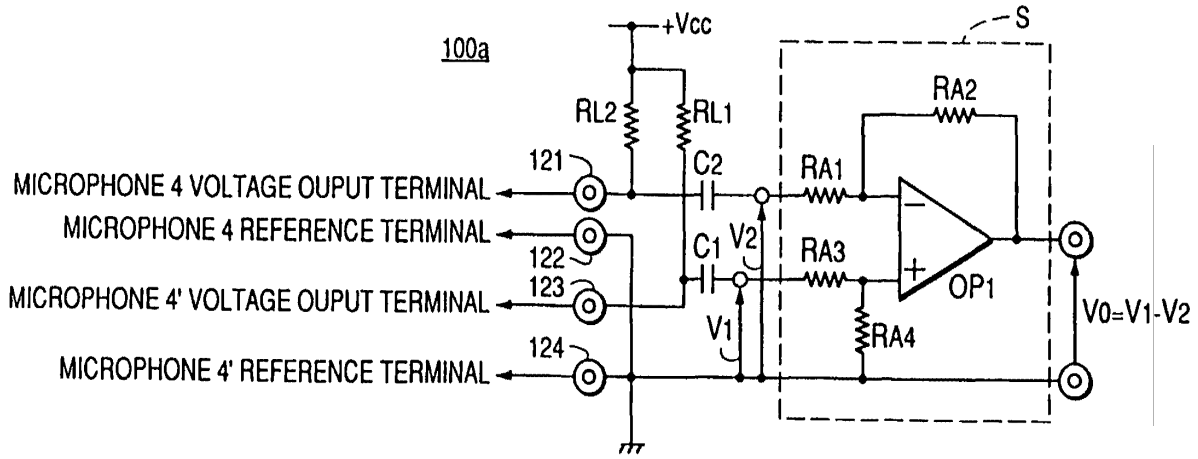


FIG. 13

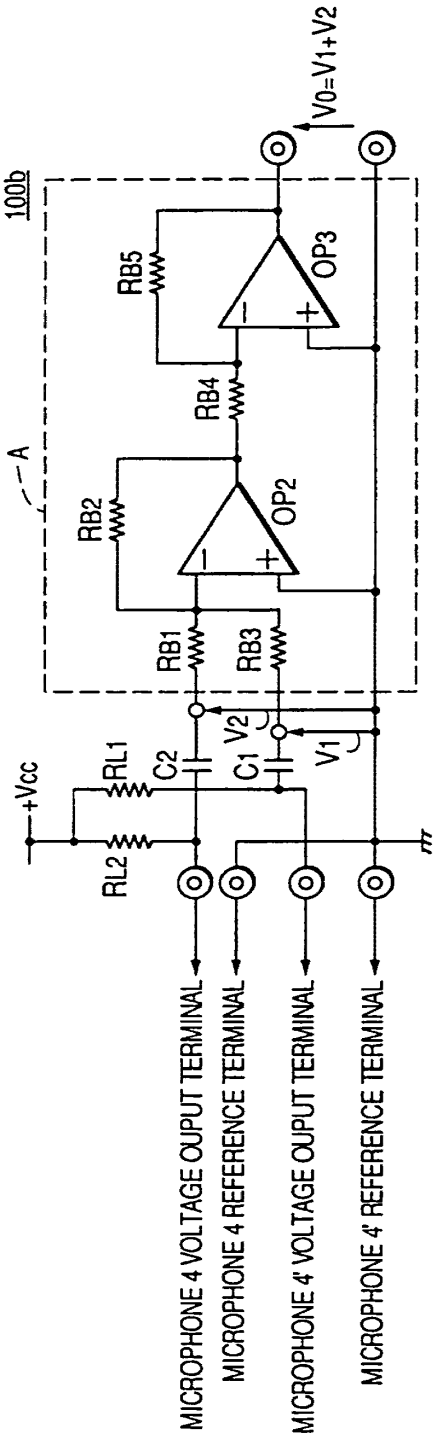

100b
A
V0=V1+V2
OP3
OP2
RB5
RB4
RB2
RB1
RB3
V2
V1
C1
C2
RL1
RL2
+Vcc
MICROPHONE 4 VOLTAGE OUPUT TERMINAL
MICROPHONE 4 REFERENCE TERMINAL
MICROPHONE 4' VOLTAGE OUPUT TERMINAL
MICROPHONE 4' REFERENCE TERMINAL

FIG. 14

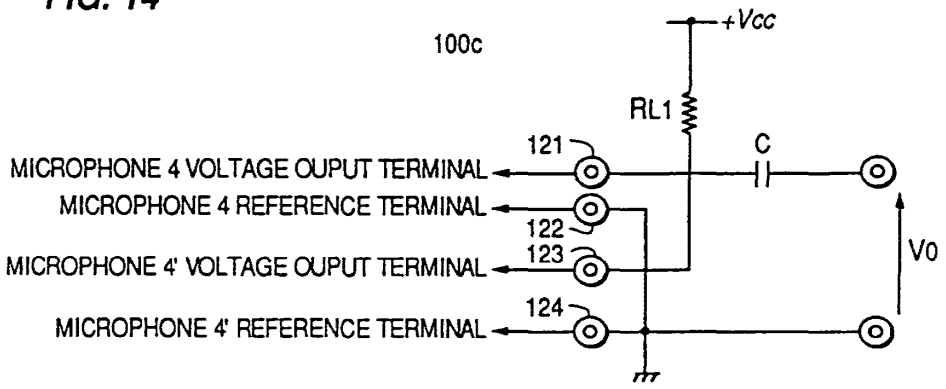


FIG. 15

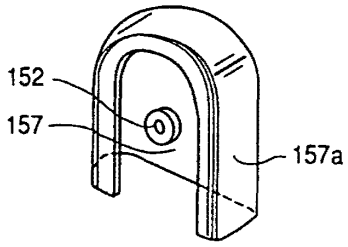


FIG. 16

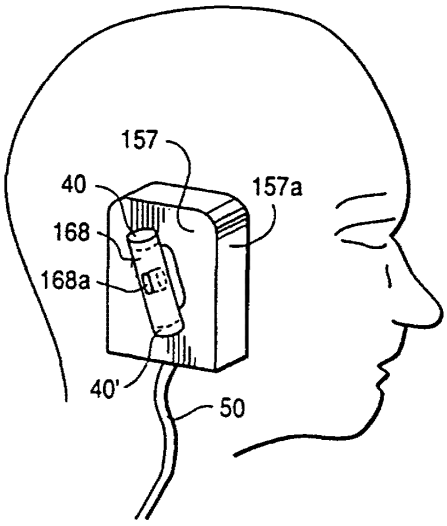


FIG. 17

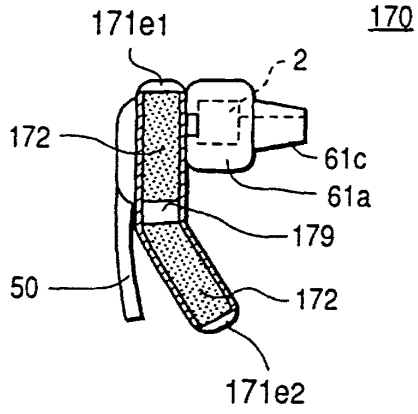


FIG. 18

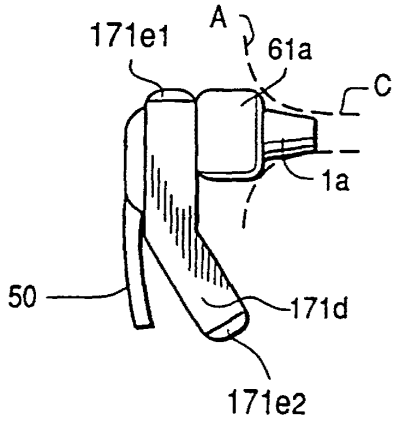


FIG. 19

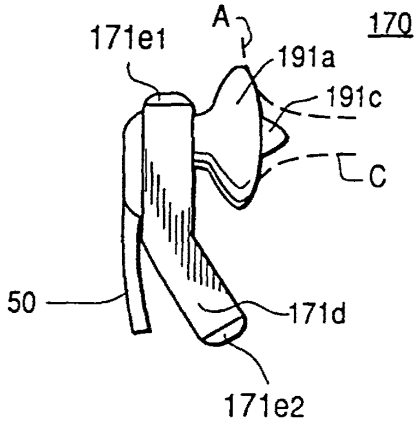


FIG. 21

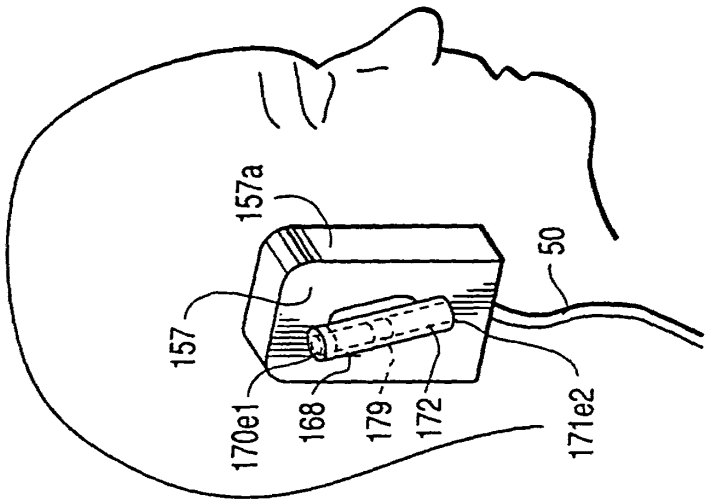


FIG. 20

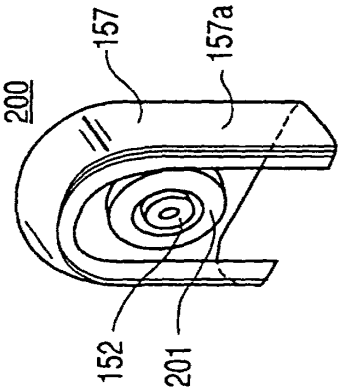


FIG. 22

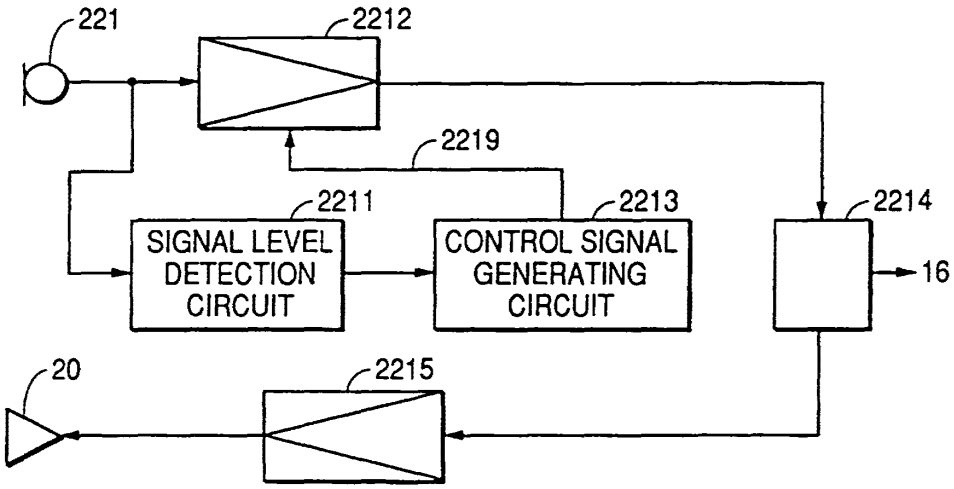
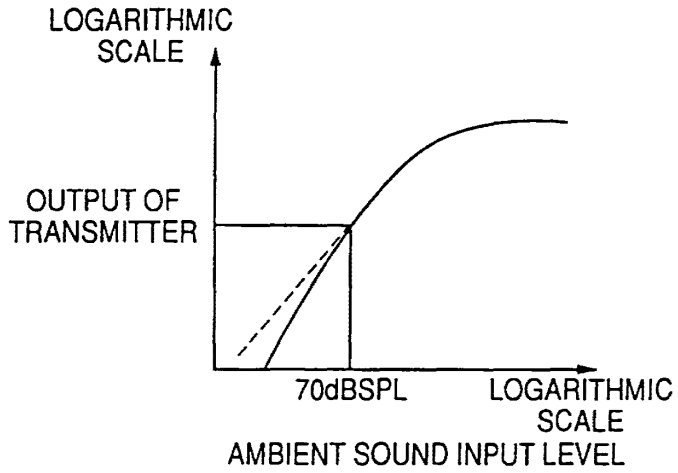

221
2212
2219
2211
2213
2214
SIGNAL LEVEL DETECTION CIRCUIT
CONTROL SIGNAL GENERATING CIRCUIT
16
20
2215

FIG. 23


LOGARITHMIC SCALE
OUTPUT OF TRANSMITTER
70dBSPL
LOGARITHMIC SCALE
AMBIENT SOUND INPUT LEVEL

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 93 11 2064

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
A	US-A-5 210 792 (KAJIHARA) * column 1, line 47-64 * * column 2, line 9-32 * * column 3, line 21 - column 5, line 30 * ---	1,6-8, 10-12, 14-18	H04R1/10
A	EP-A-0 414 264 (SONY) * page 2, line 7-10 * * page 2, line 44 - page 3, line 7 * * page 3, line 36 - page 4, line 17 * * page 5, line 45 - page 6, line 47 * ---	1-5,8,9, 12,14-18	
A	US-A-3 995 124 (GABR) * column 1, line 4-7 * * column 1, line 26-34 * * column 2, line 12 - column 4, line 17 * ---	1-3,8,9, 14,16,18	
A	US-A-5 046 103 (WARNAKA ET AL.) -----	12-14, 17,18	TECHNICAL FIELDS SEARCHED (Int.Cl.6) H04R H04M A61F

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	14 December 1993	Zanti, P

CATEGORY OF CITED DOCUMENTS

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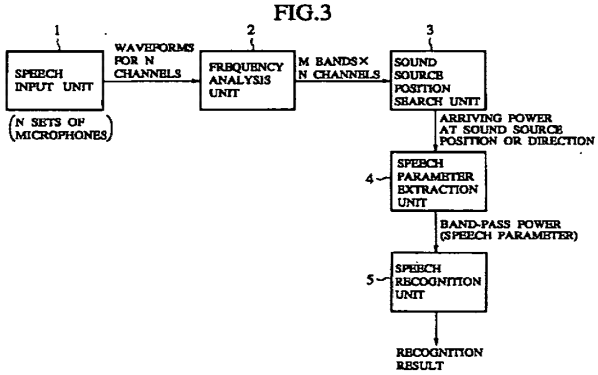
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(54) Method and system for microphone array input type speech recognition

(57) A microphone array input type speech recognition scheme capable of realizing a high precision sound source position or direction estimation by a small amount of calculations, and thereby realizing a high precision speech recognition. A band-pass waveform, which is a waveform for each frequency bandwidth, is obtained from input signals of the microphone array, and a band-pass power of the sound source is directly obtained from the band-pass waveform. Then, the obtained band-pass power is used as the speech parameter. It is also possible to realize the sound source estimation and the band-pass power estimation at high precision while further reducing an amount of calculations, by utilizing a sound source position search processing in which a low resolution position estimation and a high resolution position estimation are combined.

FIG.3



EP 0 795 851 A2

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Description

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The present invention relates to a microphone array input type speech recognition scheme in which speeches uttered by a user are inputted through microphone array and recognized.

DESCRIPTION OF THE BACKGROUND ART

In the speech recognition, a surrounding environment under which the speech input is made can largely affect the recognition performance. In particular, background noises and reflected sounds of the user's speeches can degrade the recognition performance so that they are sources of a serious problem encountered in a use of a speech recognition system. For this reason, in general, a short range microphone designed for use near the mouth of the user such as a headset microphone or a hand microphone has been employed, but it is uncomfortable to wear the headset microphone on a head for any extended period of time, while the hand microphone can limit a freedom of the user as it occupies the user's hands, and there has been a demand for a speech input scheme that can allow more freedom to the user.

A microphone array has been studied as a potential candidate for a speech input scheme that can resolve the conventionally encountered inconvenience described above, and there are some recent reports of its application to the speech recognition system. The microphone array is a set of a plurality of microphones which are arranged at spatially different positions, where noises can be reduced by the synthetic processing of outputs of these microphones.

Fig. 1 shows a configuration of a conventional speech recognition system using a microphone array. This speech recognition system of Fig. 1 comprises a speech input unit 11 having a microphone array formed by a plurality (N sets) of microphones, a sound source direction estimation unit 12, a sound source waveform estimation unit 13, a speech detection unit 14, a speech analysis unit 15, a pattern matching unit 16, and a recognition dictionary 17.

In this configuration of Fig. 1, the speech entered at the microphone array is converted into digital signals for respective microphones by the speech input unit 11, and the speech waveforms of all channels are entered into the sound source direction estimation unit 12.

At the sound source direction estimation unit 12, a sound source position or direction is estimated from time differences among signals from different microphones, using the known delay sum array method or a method based on the cross-correlation function as disclosed in U. Bub, et al.: "Knowing Who to Listen to in Speech Recognition: Visually Guided Beamforming", ICASSP '95, pp. 848-851, 1995.

A case of estimating a direction of the sound source and a case of estimating a position of the sound source respectively correspond to a case in which the sound source is far distanced from the microphone array so that the incident sound waves can be considered as plane waves and a case in which the sound source is relatively close to the microphone array so that the sound waves can be considered as propagating in forms of spherical waves.

Next, the sound source waveform estimation unit 13 focuses the microphone array to the sound source position or direction obtained by the sound source direction estimation unit 12 by using the delay sum array method, and estimates the speech waveform of the target sound source.

Thereafter, similarly as in the usual speech recognition system, the speech analysis is carried out for the obtained speech waveform by the speech analysis unit 15, and the pattern matching using the recognition dictionary 17 is carried out for the obtained analysis parameter, so as to obtain the recognition result. For a method of pattern matching, there are several known methods including the HMM (Hidden Markov Model), the multiple similarity method, and the DP matching, as detailed in Rabiner et al.: "Fundamentals of Speech Recognition", Prentice Hall, for example.

Now, in the speech recognition system, it is custom to input the speech waveform. For this reason, even in the conventional speech recognition system using the microphone array as described above, the sound source position (or the sound source direction) and the speech waveform are obtained by processing the microphone array outputs according to the delay sum array method, due to a need to estimate the speech waveform by a small amount of calculations. The delay sum array method is often utilized because the speech waveform can be obtained by a relatively small amount of calculations, but the delay sum array method is also associated with a problem that the separation power is lowered when a plurality of sound sources are located close to each other.

On the other hand, as a method for estimating the sound source position (or direction), there is a parametric method based on a model as disclosed in S. V. Pillai: "Array Signal Processing", Springer-Verlag, New York, 1989, for example, which is presumably capable of estimating the sound source position at higher precision than the delay sum array method, and which is also capable of obtaining the power spectrum necessary for the speech recognition from the sound source position estimation processing at the same time.

Fig. 2 shows a processing configuration for this conventionally proposed parametric method. In the configuration of Fig. 2, signals from a plurality of microphones are entered at a speech input unit 21, and the frequency analysis based

on the FFT (Fast Fourier Transform) is carried out at a frequency analysis unit 22. Then, the sound source position estimation processing is carried out for each frequency component at a power estimation unit 23, and the final sound source position estimation result is obtained by synthesizing the estimation results for all the frequencies at a sound source direction judgement unit 24.

Here, the sound source position estimation processing is a processing for estimating a power at each direction or position while minutely changing a direction or position over a range in which the sound source can possibly be located, so that a very large amount of calculations are required. In particular, in a case of assuming the propagation of sound waves in forms of spherical waves, it is going to estimate a position of the sound source rather than an arriving direction of the sound waves, so that two- or three-dimensional scanning is necessary and consequently an enormous amount of calculations are required.

Moreover, in the conventionally proposed parametric method described above, it is necessary to carry out this scanning processing for each frequency component obtained by the fast Fourier transform of the speech, so that it is difficult to reduce a required amount of calculations.

SUMMARY OF THE INVENTION

It is therefore an object of the present invention to provide a method and a system for microphone array input type speech recognition capable of realizing a high precision sound source position or direction estimation by a small amount of calculations, and thereby realizing a high precision speech recognition. This object is achieved by obtaining a band-pass waveform, which is a waveform for each frequency bandwidth, from input signals of the microphone array, and directly obtaining a band-pass power of the sound source from the band-pass waveform. Then, the obtained band-pass power can be used as the speech parameter.

It is another object of the present invention to provide a method and a system for microphone array input type speech recognition capable of realizing the sound source estimation and the band-pass power estimation at high precision while further reducing an amount of calculations. This object is achieved by utilizing a sound source position search processing in which a low resolution position estimation and a high resolution position estimation are combined.

According to one aspect of the present invention there is provided a microphone array input type speech recognition system, comprising: a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; a speech parameter extraction unit for extracting a speech parameter for speech recognition, from the band-pass power distribution for each frequency bandwidth calculated by the sound source position search unit, according to the sound source position or direction estimated by the sound source position search unit; and a speech recognition unit for obtaining a speech recognition result by matching the speech parameter extracted by the speech parameter extraction unit with a recognition dictionary.

According to another aspect of the present invention there is provided a microphone array input type speech analysis system, comprising: a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; and a speech parameter extraction unit for extracting a speech parameter from the band-pass power distribution for each frequency bandwidth estimated by the sound source position search unit, according to the sound source position or direction estimated by the sound source position search unit.

According to another aspect of the present invention there is provided a microphone array input type speech analysis system, comprising: a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; and a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution.

According to another aspect of the present invention there is provided a microphone array input type speech rec-

ognition method, comprising the steps of: inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; extracting a speech parameter for speech recognition, from the band-pass power distribution for each frequency bandwidth calculated by the calculating step, according to the sound source position or direction estimated by the calculating step; and obtaining a speech recognition result by matching the speech parameter extracted by the extracting step with a recognition dictionary.

According to another aspect of the present invention there is provided a microphone array input type speech analysis method, comprising the steps of: inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; and extracting a speech parameter from the band-pass power distribution for each frequency bandwidth calculated by the calculating step, according to the sound source position or direction estimated by the calculating step.

According to another aspect of the present invention there is provided a microphone array input type speech analysis method, comprising the steps of: inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones; analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; and calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution.

Other features and advantages of the present invention will become apparent from the following description taken in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram of a conventional microphone array input type speech recognition system.

Fig. 2 is a block diagram of a processing configuration for a conventionally proposed parametric method for estimating the sound source position or direction.

Fig. 3 is a block diagram of a microphone array input type speech recognition system according to the first embodiment of the present invention.

Fig. 4 is a diagram of a filter function to be used in a sound source position search unit in the system of Fig. 3.

Figs. 5A and 5B are diagrams showing a relationship between a sound source position and microphone positions in the system of Fig. 3, for a case of direction estimation and for a case of position estimation, respectively.

Fig. 6 is a diagram for explaining a peak detection from a sound source power distribution after the synthesizing processing in the system of Fig. 3.

Fig. 7 is a block diagram of one exemplary configuration for a speech recognition unit in the system of Fig. 3.

Fig. 8 is a block diagram of another exemplary configuration for a speech recognition unit in the system of Fig. 3.

Fig. 9 is a flow chart for the overall processing in the system of Fig. 3.

Fig. 10 is a diagram for explaining a read out waveform data for one frame used in a calculation of a correlation matrix at a sound source position search unit in the system of Fig. 3.

Figs. 11A and 11B are diagrams showing a relationship between a resolution and an increment value for search in a sound source position estimation in the system of Fig. 3.

Fig. 12 is a block diagram of a sound source position search unit in the system of Fig. 3 according to the second embodiment of the present invention.

Fig. 13 is a flow chart for the processing of the sound source position search unit of Fig. 12.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now to Fig. 3 to Fig. 10, the first embodiment of a method and a system for microphone array input type speech recognition according to the present invention will be described in detail.

Fig. 3 shows a basic configuration of a microphone array input type speech recognition system in this first embodiment. This speech recognition system of Fig. 1 comprises a speech input unit 1, a frequency analysis unit 2, a sound

source position search unit 3, a speech parameter extraction unit 4, and a speech recognition unit 5.

The speech input unit 1 has a microphone array (not shown) formed by N sets (8 sets, for example) of microphones, and converts speeches entered from the microphone array into digital signals.

The frequency analysis unit 2 analyzes the input speech for each microphone (channel) entered at the speech input unit 1 by a band-pass filter bank (a group of band-pass filters), and obtains a band-pass waveform which is a waveform for each frequency bandwidth.

The sound source position search unit 3 estimates a power arriving from each position or direction for each bandwidth according to the band-pass waveform for each frequency bandwidth obtained for each channel by the frequency analysis unit 2, as a sound source position judgement information, and identifies a sound source direction or position by synthesizing the obtained sound source position judgement information for a plurality of frequency bandwidths.

The speech parameter extraction unit 4 extracts the band-pass power of the speech signals arrived from the sound source direction or position identified by the sound source position search unit 3, as a speech parameter, according to the sound source position judgement information obtained at the sound source position search unit 3.

The speech recognition unit 5 carries out the speech recognition by matching the speech parameter extracted by the speech parameter extraction unit 4 with a recognition dictionary.

Now, the outline of the overall operation in the speech recognition system of Fig. 3 will be described.

First, the speeches entered at N (=8) sets of microphones are AD converted at the sampling frequency such as 12 KHz for example, microphone channel by microphone channel, at the speech input unit 1. Then, the frequency analysis is carried out at the frequency analysis unit 2, to obtain band-pass waveforms for a plurality of bands (bandwidths) corresponding to microphones. Here, a number M of bands used in the analysis is assumed to be equal to 16. The transmission bandwidths of the band-pass filters are to be determined as those required at the speech recognition unit 5. A manner of constructing the band-pass filters is well known and can be found in Rabiner et al.: "Fundamentals of Speech Recognition", Prentice Hall, for example.

Next, at the sound source position search unit 3, an arriving power in each bandwidth is estimated for each position or direction, according to the band-pass waveforms for N (=8) channels in each bandwidth outputted by the frequency analysis unit 2, as the sound source position judgement information. This processing is repeated for M (=16) times. This calculation of the sound source position judgement information is a calculation of an arriving power while sequentially displacing an assumed sound source position or direction, so as to obtain a distribution of arriving powers over a range in which the sound source can be located.

Thereafter, the sound wave arriving direction or the sound source position is estimated by synthesizing the above described power distribution obtained for each of M frequency bandwidths. Here, a position or direction with a large value in the power distribution is to be estimated as that of the source source.

In addition, at the speech parameter extraction unit 4, the sound source power (band-pass power) at the sound wave arriving direction or the sound source position is extracted, from the sound source position judgement information estimated for each bandwidth at the sound source position search unit 3, as the speech parameter. This speech parameter is then given to the speech recognition unit 5, where the speech recognition result is obtained and outputted.

As described, in this first embodiment, the sound source position is determined according to the estimated power distribution for each frequency bandwidth to be used in the speech recognition, and the speech parameter is obtained according to the determined sound source position, so that even when the sound source position is unknown, it is possible to realize the speech recognition by directly obtaining the speech parameter at high precision with a small amount of calculations.

Note that, when the sound source position is known, it suffices to obtain the arriving power value by limiting a power distribution calculation range to a single known sound source position or direction, and the configuration of Fig. 3 is also applicable to this case without any change. This simplified operation is effective when it is possible to assume that a user makes a speech input by approaching to a specific location.

Next, the detailed operation for obtaining the power distribution from a plurality of band-pass waveforms at the sound source position search unit 3 will be described.

At the sound source position search unit 3, in order to obtain a power at each direction or position from a plurality (M sets) of band-pass waveforms, the calculation of the minimum variance method is carried out. The minimum variance method is well known and described in Haykin: "Adaptive Filter Theory", Prentice Hall, for example.

In this first embodiment, at a time of the sound source power estimation by the minimum variance method, in order to deal with signals having a certain bandwidth which is not a narrow bandwidth, a filter function as indicated in Fig. 4 is realized by calculation, where the band-pass waveforms of the same frequency bandwidth obtained for N sets of microphones (i = 1 to N) by the frequency analysis unit 2 are added by an adder 32 after passing through transversal filters 31-1 to 31-N with multiple delay line taps corresponding to N sets of microphones (i = 1 to N). Here, the filter coefficients w_{11} to w_{1J}, • • • • • • • • , w_{N1} to w_{NJ} of the filters 31-1 to 31-N are switchably set up for each bandwidth, so as to realize the filter function for all bandwidths.

In the configuration of Fig. 2, a number of taps in the filter is denoted as J, and a filter coefficient of the i-th microphone (microphone No. i) is denoted as w_{ij} ($1 \leq i$ d N, $1 \leq j \leq J$). Here, J is equal to 10, for example, but this setting may

be be changed depending on a width of the bandwidth.

The filter output y in this configuration of Fig. 2 can be expressed as follows.

First, by denoting the band-pass waveform of the k-th frequency bandwidth ($1 \leq k \leq M$) obtained from the waveform for the i-th microphone as $x_{ik}(n)$, and by arranging band-pass waveform sample sequences: $x_{ik} = (x_{ik}(n-J+1), x_{ik}(n-J+2), \cdots\cdots\cdots, x_{ik}(n-1), x_{ik}(n))$ from J samples past of a certain time n up to this time n, for all N sets of microphones, it is possible to obtain a vector given by the following equation (1).

$$x_k = (x_{1k}, x_{2k}, \cdots\cdots\cdots, x_{Nk})^T \tag{1}$$

Also, by arranging the filter coefficients w_{ij}, it is possible to obtain a vector given by the following equation (2).

$$\begin{aligned} w_k = (&w_{11}, w_{12}, \cdots\cdots\cdots, w_{1J}, w_{21}, w_{22}, \cdots\cdots\cdots, \\ &w_{2J}, \cdots\cdots\cdots, w_{N1}, w_{N2}, \cdots\cdots\cdots, w_{NJ})^T \end{aligned} \tag{2}$$

Using the above equations (1) and (2), the filter output y can be expressed as:

$$y = w_k^* x_k \tag{3}$$

where * denotes a complex conjugate of a vector. In this expression, x_k is usually called a snap-shot.

Now, denoting the expectation value as E[ ], the expectation value of the filter output power y^2 is expressed as :

$$E[y^2] = E[w_k^* x_k x_k^* w_k] = w_k^* R_k w_k \tag{4}$$

where $R_k = E[x_k x_k^*]$ is a correlation matrix of x. Then, the estimation vector according to the minimum variance method is obtained by minimizing this expectation value $E[y^2]$ under the constraint conditions that a response of the microphone array for a target direction or position is to be maintained constant.

These constraint conditions can be expressed as:

$$w_k^* A = g \tag{5}$$

where g is a column vector with constant values in a size equal to a number L of constraint conditions. For example, this g can be [1, 1, ·····, 1]. Also, A is a matrix formed by direction control vectors a_m for different frequencies as column vectors. This matrix A can be expressed as:

$$A = [a_1, a_2, \cdots\cdots\cdots, a_L] \tag{6}$$

and each direction control vector a_m (m = 1, 2, ·····, L) can be expressed as:

$$a_m = (1, a_2 e^{-j\omega m \tau 2}, \cdots\cdots\cdots, a_N e^{-j\omega m \tau N}) \tag{7}$$

where $\tau_2, \cdots\cdots\cdots, \tau_N$ are propagation time differences of the incident sound wave for the second to N-th microphones with reference to the first microphone, respectively. Note that the propagation time difference τ_1 of the incident sound wave for the first microphone is set equal to zero. Also, ω_m is an angular frequency, and $a_2, \cdots\cdots\cdots, a_N$ are amplitude ratio of the incident sound wave for the second to N-th microphone with reference to the first microphone, respectively. Here, L is set equal to 10, for example, and ω_m is set to be $\omega_m = ((\omega_a - \omega_b)/(L-1)) \cdot m + \omega_b$, where ω_a is an upper limit angular frequency of the bandwidth and ω_b is a lower limit angular frequency of the bandwidth.

When the problem of minimization under the constraints given by the equations (4) and (5) is solved by the Lagrange's method of indeterminate coefficients, the filter coefficient w_k for minimizing the arriving power from any direction or position other than the sound source direction θ or the sound source position θ can be given by the following equation (8).

$$w_k = R_k^{-1} A (A^* R_k^{-1} A)^{-1} g \tag{8}$$

Using this filter coefficient w_k, the arriving power (arriving band-pass power, sound source power) $P_k(\theta)$ for the k-th bandwidth from the sound source θ can be calculated as the following equation (9).

$$P_k(\theta) = g^* (A^* R_k^{-1} A)^{-1} g \tag{9}$$

In a case of the sound source position estimation, θ is taken as a vector for expressing the coordinates.

Now, with reference to Figs. 5A and 5B, a manner of obtaining the propagation time difference and the amplitude for each microphone will be described. Here, the explanation is given on two-dimensional plane for the sake of simplicity, but the extension to the three-dimensional space should be obvious.

First, as shown in Fig. 5A, the coordinates of the first microphone (No. 1) are denoted as (x_1, y_1) and the coordinates of the i-th microphone (No. i) are denoted as (x_i, y_i). Then, for a case of the plane waves, when the sound waves are incident from a direction θ, the propagation time difference τ_i of the incident sound waves at the i-th microphone and the first microphone is given by:

$$\tau_i(\theta) = ((x_i - x_1)^2 + (y_i - y_1)^2)^{1/2} \cos(\theta - \tan^{-1}((y_i - y_1)/(x_i - x_1))) \quad (10)$$

and the amplitude can be assumed as:

$$a_1 = a_2 = \cdots\cdots\cdots = a_N = 1 \quad (11)$$

On the other hand, in a case of a point sound source, as shown in Fig. 5B, when an assumed sound source position θ is located at (x_s, y_s), the propagation time difference τ_i and the amplitude a_i can be given by:

$$\tau_i = (((x_i - x_s)^2 + (y_i - y_s)^2)^{1/2} - ((x_i - x_s)^2 + (y_i - y_s)^2)^{1/2})/c \quad (12)$$

and

$$a_i = ((x_i - x_s)^2 + (y_i - y_s)^2)^{1/2} / ((x_i - x_s)^2 + (y_i - y_s)^2)^{1/2} \quad (13)$$

where c is the speed of sound.

$P_k(\theta)$ given by the above equation (9) becomes large when θ coincides with the arriving direction or the sound source position, or small when they do not coincide. For this reason, by calculating $P_k(\theta)$ for each direction or position, the arriving direction or the sound source position can be estimated as a position of the peak.

To this end, in a case of obtaining the sound source direction, the sound source position search unit 3 calculates $P_k(\theta)$ while sequentially changing θ gradually, 1° by 1° for example. Also, in a case of obtaining the sound source position, the sound source position search unit 3 calculates $P_k(\theta)$ for lattice points at 2 cm interval for example, within the search range. The increment value for θ may be changed to any appropriate value depending on factors such as a wavelength and a distance to an assumed sound source position.

Next, at the sound source position search unit 3, $P_k(\theta)$ of the equation (9) obtained for all bandwidths are synthesized to estimate the sound source position or the sound source direction.

Here, the synthesizing can be realized by multiplying a weight W_k to the arriving power distribution $P_k(\theta)$ for each bandwidth, and taking a sum for all the frequency bandwidths from k = 1 to k = M, that is:

$$P(\theta)_{total} = \Sigma W_k P_k(\theta) \quad (14)$$

and estimating the sound source from a peak on the distribution after this synthesizing processing (the total sound source power distribution).

Here, all the weights W_k may be set equal to 1, or the weight for the frequency of a noise source with a known frequency characteristic such as a power source frequency may be set small so as to reduce the influence of the noise.

The detection of the sound source is carried out according to a size of a peak in $P(\theta)_{total}$ as described above, and a single largest peak can be detected as the sound source. Alternatively, as shown in Fig. 6, by setting a prescribed threshold with reference to an average value of portions other than peak portions on the synthesized (total) sound source power distribution, such as 5 dB, and all peaks above this threshold may be detected as the sound sources, while not detecting any sound source at all when there is no peak above this threshold.

In this manner, the arriving power distribution $P_k(\theta)$ for each bandwidth given by the equation (9) is used in Judging whether the sound source exists at an assumed sound source position which is set to be each direction or position determined with reference to positions of a plurality of microphones, so that this arriving power distribution $P_k(\theta)$ will be referred to as the sound source position judgement information.

Next, the speech parameter extraction unit 4 can extract the power of the k-th frequency bandwidth of the sound source, from the already obtained arriving power distribution $P_k(\theta)$ for each bandwidth, according to the sound source direction or position obtained by the sound source position search unit 3. Consequently, by extracting the power for all the bandwidths from k = 1 to k = M, it is possible to obtain the band-pass power to be used as the speech parameter.

The band-pass power of the sound source obtained in this manner is sent from the speech parameter extraction unit 4 to the speech recognition unit 5, and used in the speech recognition processing.

As shown in Fig. 7, the speech recognition unit 5 comprises a speech power calculation unit 501, a speech detection unit 502, a pattern matching unit 503, and a recognition dictionary 504.

In this speech recognition unit 5, the speech power is calculated from the speech parameter (band-pass power) extracted by the speech parameter extraction unit 4, and a speech section is detection by the speech detection unit 502 according to the calculated speech power. Then, for the speech parameter in the detected speech section, the pattern matching with the recognition dictionary 504 is carried out by the pattern matching unit 503.

Note that, as shown in Fig. 8, the speech recognition unit 5 may be formed by a pattern matching unit 511, a recognition dictionary 512, and a speech detection unit 513, so as to carry out the word spotting scheme in which the continuous pattern matching for the speech parameter is carried out and a section with the largest matching score is determined as a speech section.

The total speech power can be obtained by summing powers for all the bandwidths extracted by the speech parameter extraction unit 4, so that it is possible to use the known speech section detection method based on the speech power as disclosed in L. F. Lamel et al.: "An Improved Endpoint Detector for Isolated Word Recognition", IEEE Transactions on Acoustics, Speech, and Signal Processing, Vol.ASSP-29, No. 4, pp. 777-785, August 1981. The above processing is to be carried out for each frame of the input speech waveform data so as to recognize the speech continuously.

In this first embodiment, using the configuration of the microphone array input type speech recognition system as described above, the speech recognition is realized by directly obtaining the band-pass power which is the speech parameter. Consequently, it is possible to use the minimum variance method which is a sound source direction or sound source position estimation method with a high precision and a small amount of calculations.

Now, the flow of processing as described above will be summarized with reference to the flow chart of Fig. 9.

First, prior to the start of the processing, the initial setting is made for factors such as whether a direction estimation is to be used or a position estimation is to be used, a range of sound source search to be used, and an increment value to be used in the search (step S1). In an example of Fig. 9, a direction estimation is to be used, a searching range is from -90° to +90°, and an increment value for search is 1°.

Next, the speeches entered at N sets of microphones are A/D converted at the sampling frequency of 12 KHz for example, for all N channels in parallel, by the speech input unit 1. The obtained waveform data are then stored in a buffer (not shown) of the speech input unit 1 (step S2). Normally, this step S2 is carried out continuously in real time, regardless of the other processing.

Next, the waveform data for each channel is read out from the buffer for one frame size, 256 points for example, and applied to the band-pass filter bank at the frequency analysis unit 2, so as to extract the band-pass waveform for each frequency bandwidth k (k = 1 to M), where M = 16 in this example (step S3). Here, the calculations for the band-pass filters may be carried out independently for each microphone in parallel, or sequentially for each microphone serially.

Next, at the source source position search unit 3, using the band-pass waveform data for N channels obtained by the frequency analysis unit 2 at the step S3, the correlation matrix R_k for each frequency bandwidth k is obtained (step S4). Here, as shown in Fig. 10, the calculation of the correlation matrix R_k is realized by obtaining the correlation matrix R_k as a time average of the auto-correlation matrices of samples for 20 frames taken at an interval of 2 samples from a frame data of 256 samples (points) for example.

In addition, at the step S4, using this correlation matrix R_k, the arriving power distribution $Pk(\theta) = g^* (A^* R_k^{-1} A)^{-1} g$ is obtained as the sound source position judgement information for each assumed position or direction. This calculation is carried out over the entire space to be searched through, so as to obtain the spatial distribution of the arriving powers. As for the bandwidths, when M = 16, the calculation is carried out from k = 1 to k = 16.

Next, at the sound source position search unit 3, the arriving power distributions $P_k(\theta)$ for different frequency bandwidths are summed over the entire frequency bandwidths, for each θ, so as to obtain the total sound source power distribution $P(\theta)_{total}$. Then, the largest peak is extracted from this $P(\theta)_{total}$ and identified as the sound source position θ_ϕ (step S5).

Next, at the speech parameter extraction unit 4, a value on the arriving power distribution (sound source position judgement information distribution) $P_k(\theta)$ for each frequency bandwidth obtained by the sound source position search unit 3 at the sound source position θ_ϕ is extracted, and this is repeated for all the frequency bandwidths for each sound source, so as to obtain the speech parameter $P_k(\theta_\phi)$ (step S6).

In addition, at the step S6, the powers for different bandwidths k of the speech parameter $P_k(\theta)$ are summed to obtain the power for the entire speech bandwidth at the speech power calculation unit 501 of the speech recognition unit 5.

Next, using the power for the entire speech bandwidth obtained at the step S6, the speech section is detected by the speech detection unit 502 of the speech recognition unit 5 (step S7).

Then, whether the end of the speech section is detected by the speech detection unit 502 or not is judged (step S8), and if not, the processing returns to the step S2 to carry out the frequency analysis for the next waveform data frame.

On the other hand, when the end of the speech section is detected, a matching of the speech parameter in the detected speech section with the recognition dictionary 504 is carried out, and the obtained recognition result is outputted (step S9). Then, the processing returns to the step S2 to carry out the frequency analysis for the next waveform data frame.

Thereafter, the above processing is repeated so as to carry out the speech parameter estimation and the speech recognition continuously.

Note that the processing described above can be carried out at high speed by adopting the pipeline processing using a plurality of processors (as many processors as a number of microphones, for example) which are operated in parallel.

Referring now to Fig. 11A to Fig. 13, the second embodiment of a method and a system for microphone array input type speech recognition according to the present invention will be described in detail.

This second embodiment is directed to a scheme for further reducing an amount of calculations in the sound source position estimation, by changing the constraint conditions used in the spectrum estimation so as to control the resolution, and changing the search density according to the resolution so as to reduce an amount of calculation in the sound source search. In this second embodiment, the basic configuration of the speech recognition system is the same as that of the first embodiment, so that Fig. 3 will be also referred in the following description.

In the first embodiment described above, the constraint conditions used in the spectrum estimation based on the minimum variance method are that a response of the microphone array for one direction or position is to be maintained constant. In this case, the resolution of the estimation is sufficiently high, so that the peak is found by the dense search in which the arriving power is obtained while changing θ 1° by 1° for example within the sound source search range.

When the resolution is so high as in this case, as shown in Fig. 11A, there is a possibility for failing to detect an accurate apex of the peak when the search is carried out not so densely, so that an amount of calculations for the search cannot be reduced easily.

In contrast, when the resolution of the sound source position estimation processing can be lowered, as shown in Fig. 11B, a possibility for overlooking the peak position can be made low even when the search is carried out coarsely, so that an amount of calculations can be reduced. In this case, however, it may not be possible to separate the closely located sound sources, or the estimation precision may be lowered, as much as the resolution is lowered.

For this reason, this second embodiment adopts a scheme in which the lower resolution search is carried out first and then the high resolution search is carried out only in a vicinity of the peak, so as to realize the high precision sound source position estimation with a small amount of calculations. This scheme will now be described in detail.

The resolution at a time of the sound source position estimation can be controlled by requiring that responses of the microphone array for a plurality of directions or positions are to be simultaneously maintained constant, instead of just requiring a response of the microphone array for one direction or position to be maintained constant as in the constraint conditions of the equation (5).

For example, using two time delays $\tau_1(\theta_1)$ and $\tau_2(\theta_2)$ for two angles θ_1 and θ_2 (according to the equation (10)), it is possible to use two direction control vectors $a_m(\theta_1)$ and $a_m(\theta_2)$ (m = 1, 2, ·····, L) given by the following equations (15) and (16).

$$a_m(\theta_1) = (1, a_2 e^{-j\omega m\tau 2(\theta 1)}, \cdots\cdots\cdots, a_N e^{-j\omega m\tau N(\theta 1)}) \quad (15)$$

$$a_m(\theta_2) = (1, a_2 e^{-j\omega m\tau 2(02)}, \cdots\cdots\cdots, a_N e^{-j\omega m\tau (02)}) \quad (16)$$

Then, using these two direction control vectors $a_m(\theta_1)$ and $a_m(\theta_2)$, it is possible to set:

$$A = [a_1(\theta_1), a_2(\theta_1), \cdots\cdots\cdots, a_L(\theta_1), a_1(\theta_2), a_2(\theta_2), \cdots\cdots\cdots, a_L(\theta_2)] \quad (17)$$

so as to make responses of the microphone array to two directions simultaneously.

Here, when θ_1 and θ_2 are set to close values, such as $\theta_2 = \theta_1 + 1°$ for example, it is equivalent to a case of making a response of the microphone array for a single direction with a width between θ_1 and θ_2, so that it is equivalent to the lowering of the resolution. Note here that a number of directions for which the responses of the microphone array are to be made simultaneously is not necessarily limited to two.

When the resolution is lowered, the search can be coarser compared with a case of high resolution, so that an amount of calculations can be reduced.

Then, after the search using the low resolution sound source position estimation processing described above, the high resolution search as described in the first embodiment can be carried out only in a vicinity of the peak detected by the first search, so that the high precision sound source position estimation can be realized with a reduced amount of calculations overall.

Fig. 12 shows a configuration of the sound source position search unit 3 in this second embodiment for realizing the above described sound source position estimation processing.

In this configuration of Fig. 12, the sound source position search unit 3 comprises a low resolution sound source position search unit 301 and a high resolution sound source position search unit 302. The low resolution sound source position search unit 301 coarsely estimates the arriving power distribution in terms of positions or directions by using the low resolution spectrum estimation. The high resolution sound source position search unit 302 densely estimates the arriving power distribution by using the high resolution spectrum estimation only in a vicinity of the position or direction obtained by the low resolution sound source position search unit 301.

Now, the flow of processing in this sound source position search unit 3 in the configuration of Fig. 12 will be described with reference to the flow chart of Fig. 13.

First, using the inputs of the band-pass waveforms corresponding to the microphone, the correlation matrix R_k is calculated (step S11). Here, a method for obtaining this correlation matrix R_k is the same as in the first embodiment.

Next, using the obtained correlation matrix R_k, the low resolution sound source position search is carried out (step S12). At this point, the increment value θ_d for search is set to be a relatively large value, such as 5° for example, so as to carry out the search coarsely over the entire search range. Also, in order to lower the resolution, a matrix as expressed by the equation (17) which has two direction control vectors $a_m(\theta_1)$ and $a_m(\theta_2)$ for two directions or positions as expressed by the equations (15) and (16) as column vectors is used instead of a matrix A in the equation (9). In Fig. 13, this matrix is denoted as B in order to distinguish it from a matrix A of the equation (6). The search is carried out for each bandwidth.

Next, the low resolution arriving power distributions for different bandwidths are synthesized, and the sound source position $\theta_\emptyset$ is obtained from a peak therein (step S13).

Next, in a vicinity of the sound source position obtained at the step S13, the high resolution sound source position search is carried out. Here, the setting of the search range is set to be ±10° of the sound source position obtained at the step S13, for example. At this point, the equation to be used for the arriving power estimation (arriving power distribution) is the same as the equation (9), and the increment value is set to a smaller value such as 1° for example (step S14).

Next, the high resolution arriving power distributions for different bandwidths obtained at the step S14 are synthesized, and the sound source position $\theta_\emptyset'$ is obtained from a peak therein (step S15).

At the speech parameter extraction unit 4, the power (speech parameter) of the sound source is extracted from the arriving power distribution obtained by the high resolution sound source position search at the high resolution sound source position search unit 302 of the sound source position search unit 3.

As described, in this second embodiment, by using the sound source position search processing in which the low resolution sound source position estimation and the high resolution sound source position estimation are combined, it is possible to estimate the sound source position and its band-pass power while reducing an amount of calculations considerably.

As described, according to the present invention, a band-pass waveform which is a waveform for each frequency bandwidth is obtained from input signals of the microphone array, and a band-pass power of the sound source is directly obtained from the band-pass waveform, so that it is possible to realize a high precision sound source position or direction estimation by a small amount of calculations. Moreover, the obtained band-pass power can be used as the speech parameter so that it is possible to realize a high precision speech recognition.

In the speech recognition system of the present invention, the input signals of the microphone array entered by the speech input unit are frequency analyzed by the frequency analysis unit, so as to obtain the band-pass waveform which is a waveform for each frequency bandwidth. This band-pass waveform is obtained by using the band-pass filter band (a group of band-pass filters), instead of using the frequency analysis based on FFT as in the conventional speech recognition system. Then, the band-pass power of the sound source is directly obtained from the obtained band-pass waveform by the sound source position search unit.

Here, in order to handle signals within some bandwidth collectively, a filter configuration (filter function) having a plurality of delay line taps for each microphone channel is used and the sound source power is obtained as a sum of the filter outputs for all channels, while using the minimum variance method which is a known high precision spectrum estimation method.

The sound source power estimation processing using the minimum variance method is also used in the conventionally proposed parametric method described above, but a use of only one delay line tap has been assumed conventionally, so that it has been impossible to obtain the bandwidth power collectively.

In contrast, in the speech recognition system of the present invention, a filter configuration with a plurality of delay line taps is used so that the power in each direction or position is obtained for each frequency bandwidth necessary for the speech recognition, rather than obtaining the power in each direction or position for each frequency, and therefore the obtained power can be directly used for the speech recognition while a required amount of calculations can be reduced.

For example, in a case of using the conventional FFT with 512 points, it has been necessary to repeatedly obtain the power in each direction for each of 256 components, but in the present invention, when a number of bands in the band-pass filter bank is set to 16 for example, it suffices to estimate the power in each direction for 16 times. In addition,

this power (band-pass power) can be estimated at higher precision compared with the conventional case of using the delay sum array processing, so that it is possible to realize the high precision speech recognition.

In the sound source position search unit, the synthesizing of the band-pass power distributions for a plurality of frequency bandwidths can be realized by multiplying respective weights to the band-pass power distributions of different bandwidths, and taking a sum of the weighted band-pass power distributions for all the frequency bandwidths. Here, all the weights may be set equal to 1, or the weight for the frequency of a noise source with a known frequency characteristic such as a power source frequency may be set small so as to reduce the influence of the noise.

Also, in the sound source position search unit, the estimation of the sound source position or direction from the synthesized power distribution for each position or direction can be realized by detecting a peak in the synthesized power distribution and setting a position or direction corresponding to the detected peak as the sound osurce position or direction.

Furthermore, according to the present invention, by using a sound source position search processing in which a low resolution position estimation and a high resolution position estimation are combined, it is possible to realize the sound source estimation and the band-pass power estimation at high precision while further reducing an amount of calculations.

In this case, the coarse search carried out by the sound source position search unit for the purpose of reducing an amount of calculations is the low resolution search so that it is possible to malke it highly unlikely to overlook the sound source position or direction (a peak position in the power distribution).

Although there is a possibility for being unable to separate closely located sound sources or lowering the estimation precision by the low resolution search alone, the high resolution search is also carried out only in a vicinity of the sound source position or direction obtained by the low resolution search (in a vicinity of the detected peak), so that it is possible to realize the high precision sound source position estimation with a further reduced amount of calculations.

It is to be noted that, in the above description, the speech recognition system having a speech recognition unit has been described, but the speech parameter extraction technique according to the present invention can be utilized separately from the speech recognition device.

Namely, it is also possible to provide a microphone array input type speech analysis system for analyzing the input signals of the microphone array and extracting the speech parameter, which is formed by the speech input unit, the frequency analysis unit, the sound source position search unit, and the speech parameter extraction unit substantially as described above, in which the band-pass waveform which is a waveform for each frequency bandwidth is obtained from input signals of the microphone array, a band-pass power of the sound source is directly obtained from the band-pass waveform, and the obtained band-pass power is used as the speech parameter.

Similarly, it is also possible to provide a microphone array input type speech analysis system for analyzing the input signals of the microphone array and estimating the sound source position or direction, which is formed by the speech input unit, the frequency analysis unit, and the sound source position search unit substantially as described above, in which the band-pass waveform which is a waveform for each frequency bandwidth is obtained from input signals of the microphone array, a band-pass power of the sound source is directly obtained from the band-pass waveform, and the sound source position or direction is estimated according to a synthesized band-pass power distribution.

Such a microphone array input type speech analysis system according to the present invention is utilizable not only in the speech recognition but also in the other speech related processing such as a speaker recognition.

It is also to be noted that, besides those already mentioned above, many modifications and variations of the above embodiments may be made without departing from the novel and advantageous features of the present invention. Accordingly, all such modifications and variations are intended to be included within the scope of the appended claims.

Claims

1. A microphone array input type speech recognition system, comprising:

   a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones;

   a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth;

   a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution;

   a speech parameter extraction unit for extracting a speech parameter for speech recognition, from the band-pass power distribution for each frequency bandwidth calculated by the sound source position search unit, according to the sound source position or direction estimated by the sound source position search unit; and

a speech recognition unit for obtaining a speech recognition result by matching the speech parameter extracted by the speech parameter extraction unit with a recognition dictionary.

2. The system of claim 1, wherein the sound source position search unit includes:

a low resolution sound source position estimation unit for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and
a high resolution sound source position estimation unit for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation unit, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant, wherein the speech parameter extraction unit extracts the speech parameter for speech recognition according to the accurate sound source position or direction.

3. The system of claim 1, wherein the frequency analysis unit obtains the band-pass waveforms for each channel by using a band-pass filter bank.

4. The system of claim 1, wherein the sound source position search unit calculates the band-pass power distribution for each frequency bandwidth, by calculating a band-pass power for each frequency bandwidth, in each one of a plurality of assumed sound source positions or directions within a prescribed search range.

5. The system of claim 1, wherein the sound source position search unit calculates the band-pass power distribution for each frequency bandwidth by using a filter function configuration having a plurality of delay line taps for each channel.

6. The system of claim 1, wherein the sound source position search unit calculates the band-pass power distribution for each frequency bandwidth by using a minimum variance method for minimizing an output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant.

7. The system of claim 1, wherein the speech parameter extraction unit extracts the band-pass power distribution for each frequency bandwidth calculated by the sound source position search unit for the sound source position or direction estimated by the sound source position search unit directly as the speech parameter.

8. The system of claim 1, wherein the sound source position search unit synthesizes the calculated band-pass power distributions for a plurality of frequency bandwidths by weighting the calculated band-pass power distributions with respective weights, and summing weighted band-pass power distributions.

9. The system of claim 1, wherein the sound source position search unit estimates the sound source position or direction by detecting a peak in the synthesized band-pass power distribution and setting a position or direction corresponding to a detected peak as the sound source position or direction.

10. A microphone array input type speech analysis system, comprising:

a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones;
a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth;
a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; and
a speech parameter extraction unit for extracting a speech parameter from the band-pass power distribution for each frequency bandwidth estimated by the sound source position search unit, according to the sound source position or direction estimated by the sound source position search unit.

11. The system of claim 10, wherein the sound source position search unit includes:

a low resolution sound source position estimation unit for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and

a high resolution sound source position estimation unit for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation unit, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant, wherein the speech parameter extraction unit extracts the speech parameter according to the accurate sound source position or direction.

12. A microphone array input type speech analysis system, comprising:

a speech input unit for inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones;

a frequency analysis unit for analyzing an input speech of each channel inputted by the speech input unit, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; and

a sound source position search unit for calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the frequency analysis unit, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution.

13. The system of claim 12, wherein the sound source position search unit includes:

a low resolution sound source position estimation unit for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and

a high resolution sound source position estimation unit for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation unit, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant.

14. A microphone array input type speech recognition method, comprising the steps of:

inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones:

analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth;

calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution;

extracting a speech parameter for speech recognition, from the band-pass power distribution for each frequency bandwidth calculated by the calculating step, according to the sound source position or direction estimated by the calculating step; and

obtaining a speech recognition result by matching the speech parameter extracted by the extracting step with a recognition dictionary.

15. The method of claim 14, wherein the calculating step includes the steps of:

a low resolution sound source position estimation step for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and

a high resolution sound source position estimation step for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation step, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant, wherein the extracting step extracts the speech parameter for speech recognition according to the accurate sound source position or direction.

16. The method of claim 14, wherein the analyzing step obtains the band-pass waveforms for each channel by using a band-pass filter bank.

17. The method of claim 14, wherein the calculating step calculates the band-pass power distribution for each frequency bandwidth, by calculating a band-pass power for each frequency bandwidth, in each one of a plurality of assumed sound source positions or directions within a prescribed search range.

18. The method of claim 14, wherein the calculating step calculates the band-pass power distribution for each frequency bandwidth by using a filter function configuration having a plurality of delay line taps for each channel.

19. The method of claim 14, wherein the calculating step calculates the band-pass power distribution for each frequency bandwidth by using a minimum variance method for minimizing an output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant.

20. The method of claim 14, wherein the extracting step extracts the band-pass power distribution for each frequency bandwidth calculated by the calculating step for the sound source position or direction estimated by the calculating step directly as the speech parameter.

21. The method of claim 14, wherein the calculating step synthesizes the calculated band-pass power distributions for a plurality of frequency bandwidths by weighting the calculated band-pass power distributions with respective weights, and summing weighted band-pass power distributions.

22. The method of claim 14, wherein the calculating step estimates the sound source position or direction by detecting a peak in the synthesized band-pass power distribution and setting a position or direction corresponding to a detected peak as the sound source position or direction.

23. A microphone array input type speech analysis method, comprising the steps of:

inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones;
analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth;
calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution; and
extracting a speech parameter from the band-pass power distribution for each frequency bandwidth calculated by the calculating step, according to the sound source position or direction estimated by the calculating step.

24. The method of claim 23, wherein the calculating step includes the steps of:

a low resolution sound source position estimation step for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and
a high resolution sound source position estimation step for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation step, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant, wherein the extracting step extracts the speech parameter according to the accurate sound source position or direction.

25. A microphone array input type speech analysis method, comprising the steps of:

inputting speeches in a plurality of channels using a microphone array formed by a plurality of microphones;
analyzing an input speech of each channel inputted by the inputting step, and obtaining band-pass waveforms for each channel, each band-pass waveform being a waveform for each frequency bandwidth; and
calculating a band-pass power distribution for each frequency bandwidth from the band-pass waveforms for each frequency bandwidth obtained by the analyzing step, synthesizing calculated band-pass power distributions for a plurality of frequency bandwidths, and estimating a sound source position or direction from a synthesized band-pass power distribution.

26. The method of claim 25, wherein the calculating step includes the steps of:

a low resolution sound source position estimation step for estimating a rough sound source position or direction, by minimizing an output power of the microphone array under constraints that responses of the microphone array for a plurality of directions or positions are to be maintained constant; and
a high resolution sound source position estimation step for estimating an accurate sound source position or direction in a vicinity of the rough sound source position or direction estimated by the low resolution sound source position estimation step, by minimizing the output power of the microphone array under constraints that a response of the microphone array for one direction or position is to be maintained constant.

FIG.1
PRIOR ART

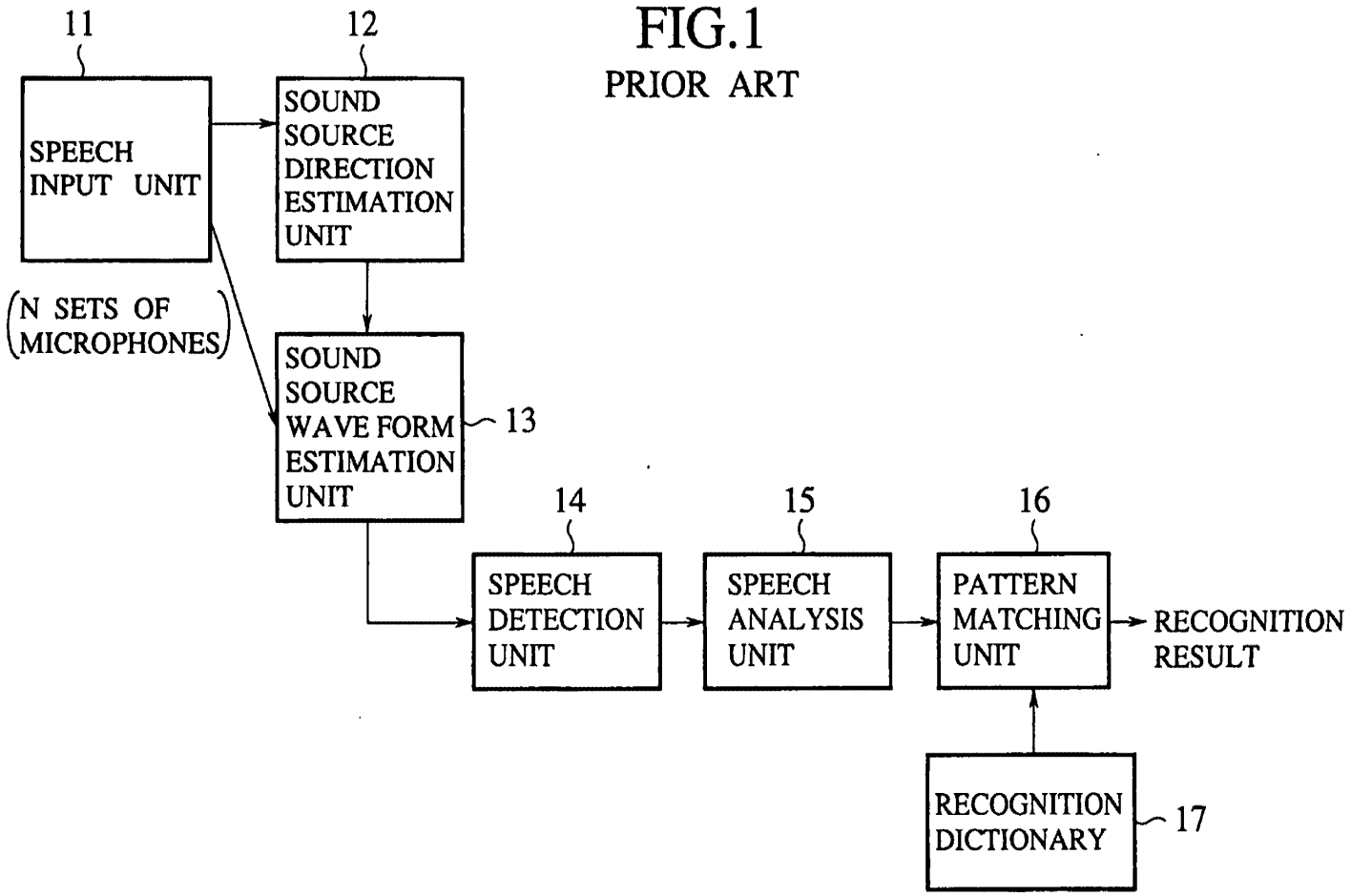


FIG.2
PRIOR ART

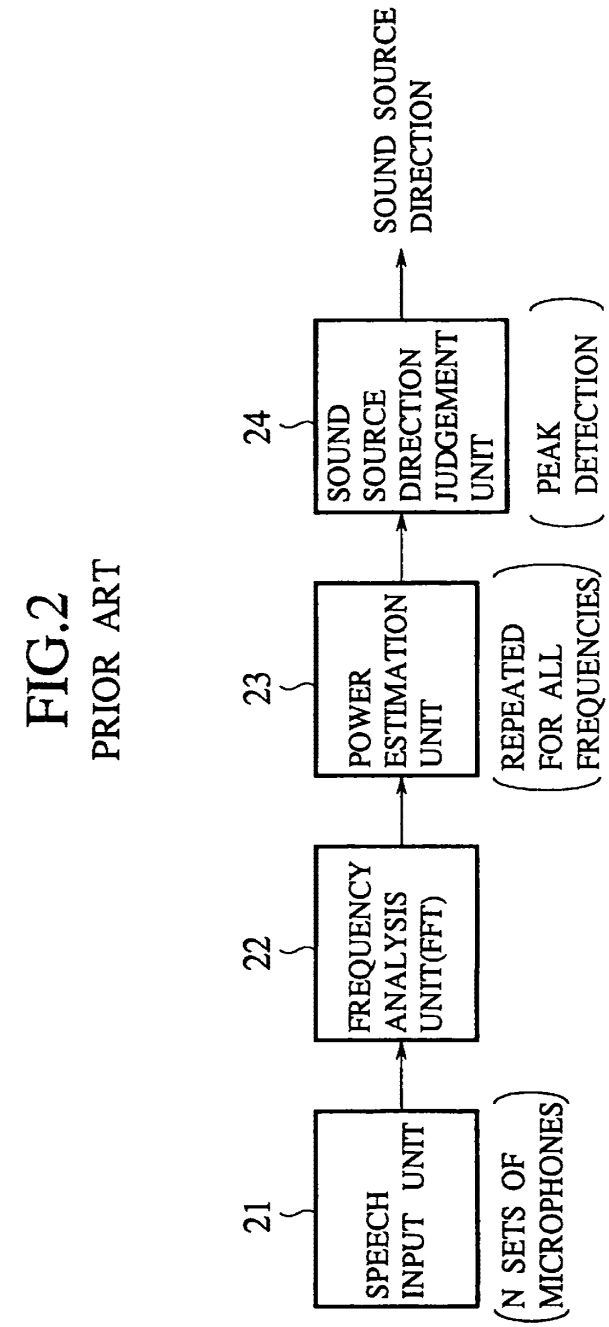


FIG.3

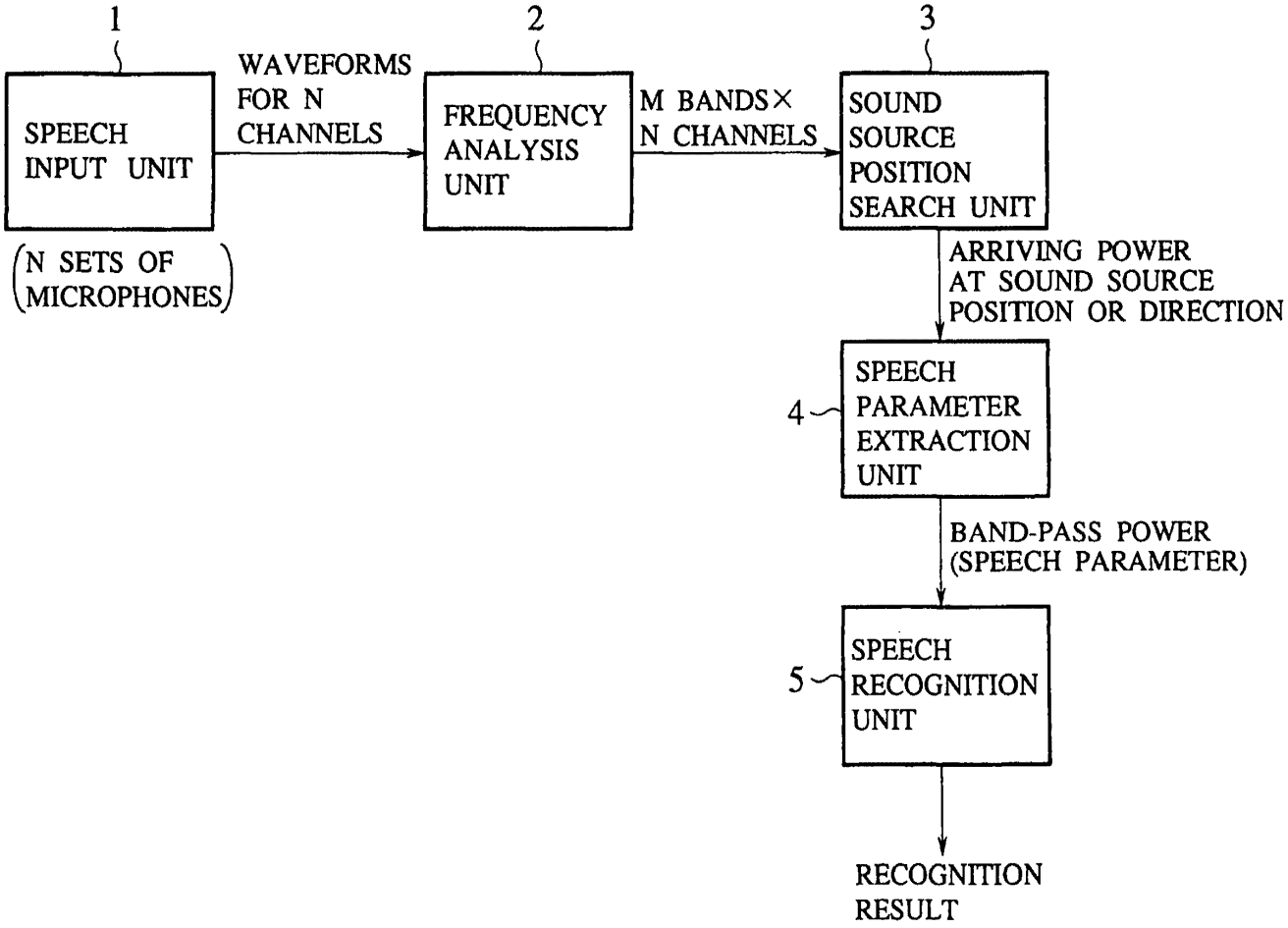

1
SPEECH INPUT UNIT
(N SETS OF MICROPHONES)
WAVEFORMS FOR N CHANNELS
2
FREQUENCY ANALYSIS UNIT
M BANDS × N CHANNELS
3
SOUND SOURCE POSITION SEARCH UNIT
ARRIVING POWER AT SOUND SOURCE POSITION OR DIRECTION
4
SPEECH PARAMETER EXTRACTION UNIT
BAND-PASS POWER (SPEECH PARAMETER)
5
SPEECH RECOGNITION UNIT
RECOGNITION RESULT

FIG.4

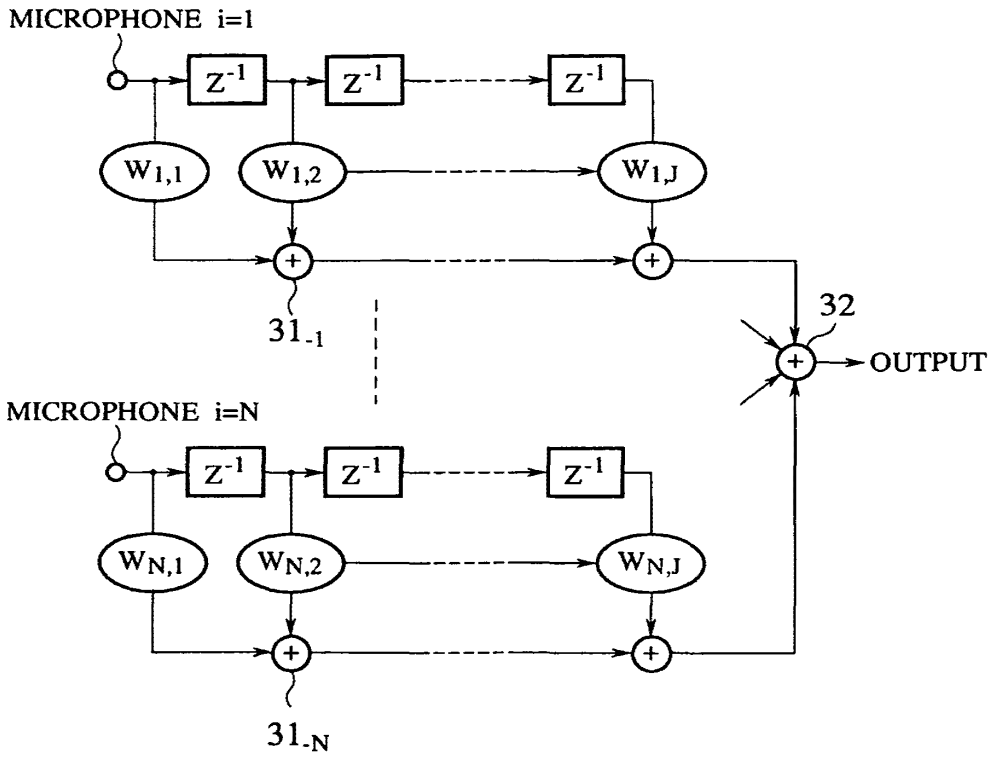

MICROPHONE i=1
Z-1
Z-1
Z-1
W1,1
W1,2
W1,J
31-1
32
OUTPUT
MICROPHONE i=N
Z-1
Z-1
Z-1
WN,1
WN,2
WN,J
31-N

FIG.5A

DIRECTION ESTIMATION

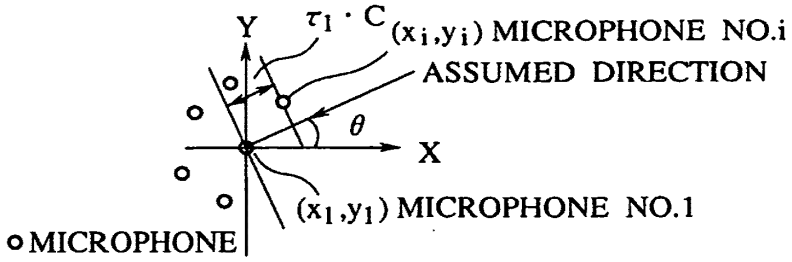


FIG.5B

POSITION ESTIMATION

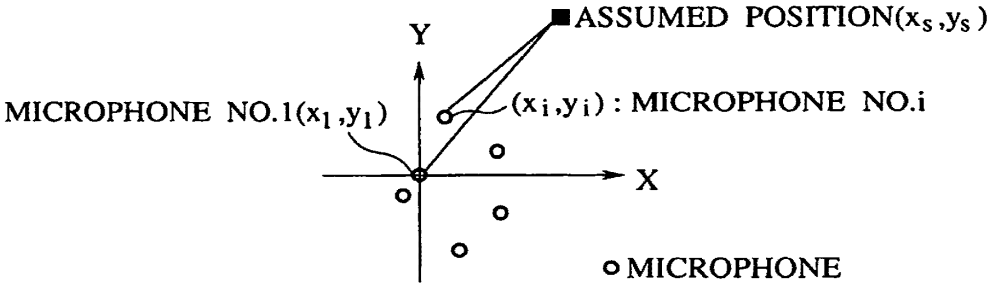


FIG.6

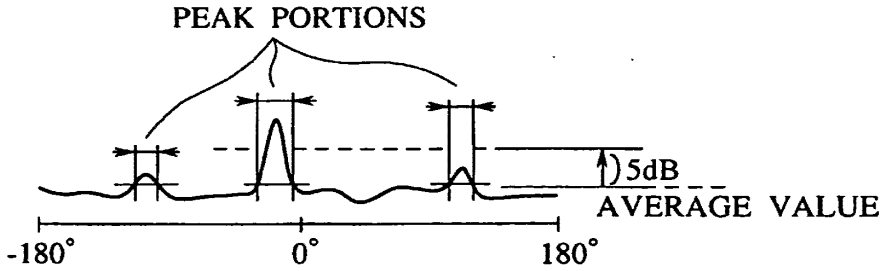
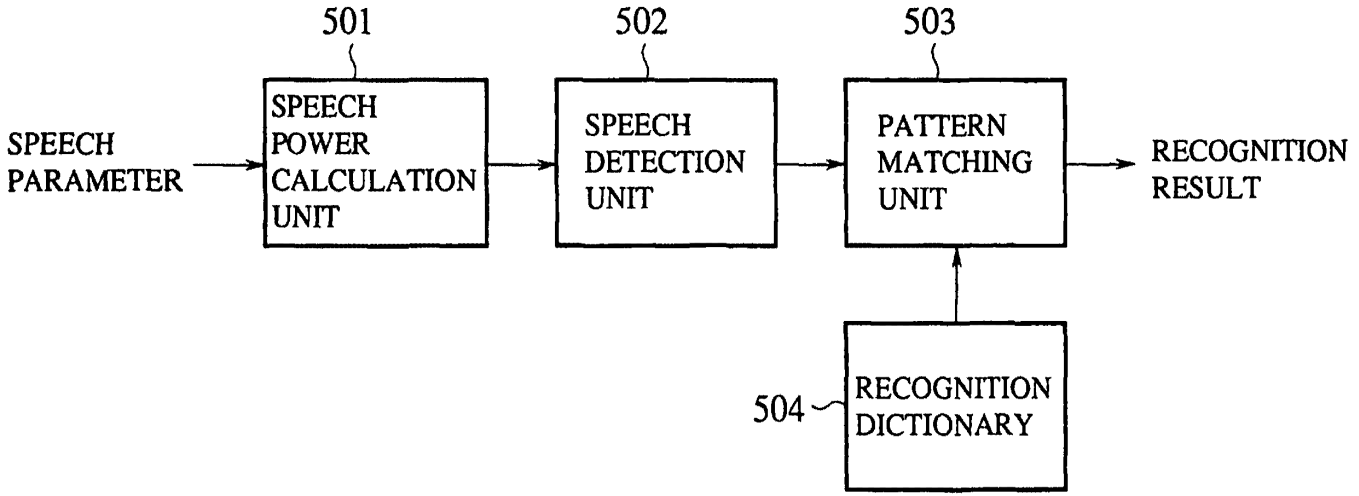


FIG.7



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FIG.8

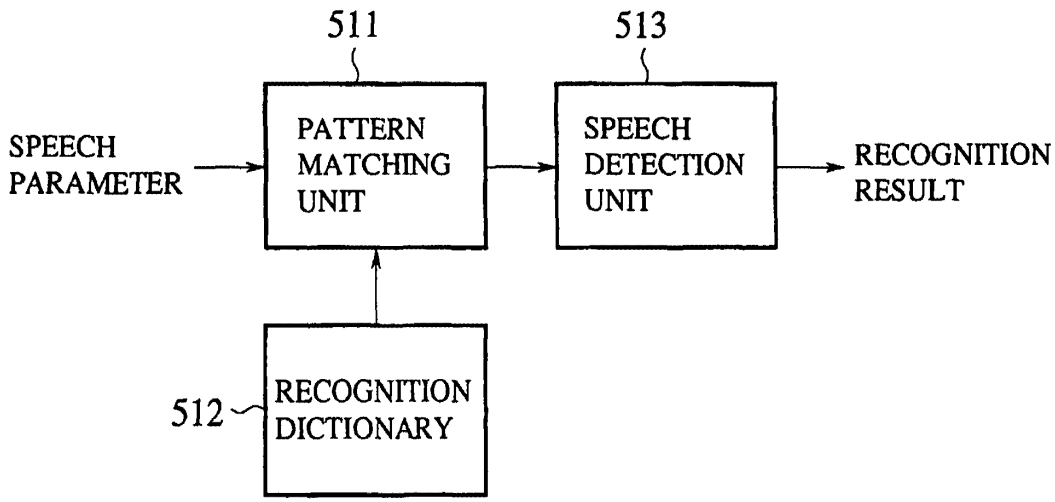

511
513
SPEECH PARAMETER
PATTERN MATCHING UNIT
SPEECH DETECTION UNIT
RECOGNITION RESULT
512
RECOGNITION DICTIONARY

FIG.9

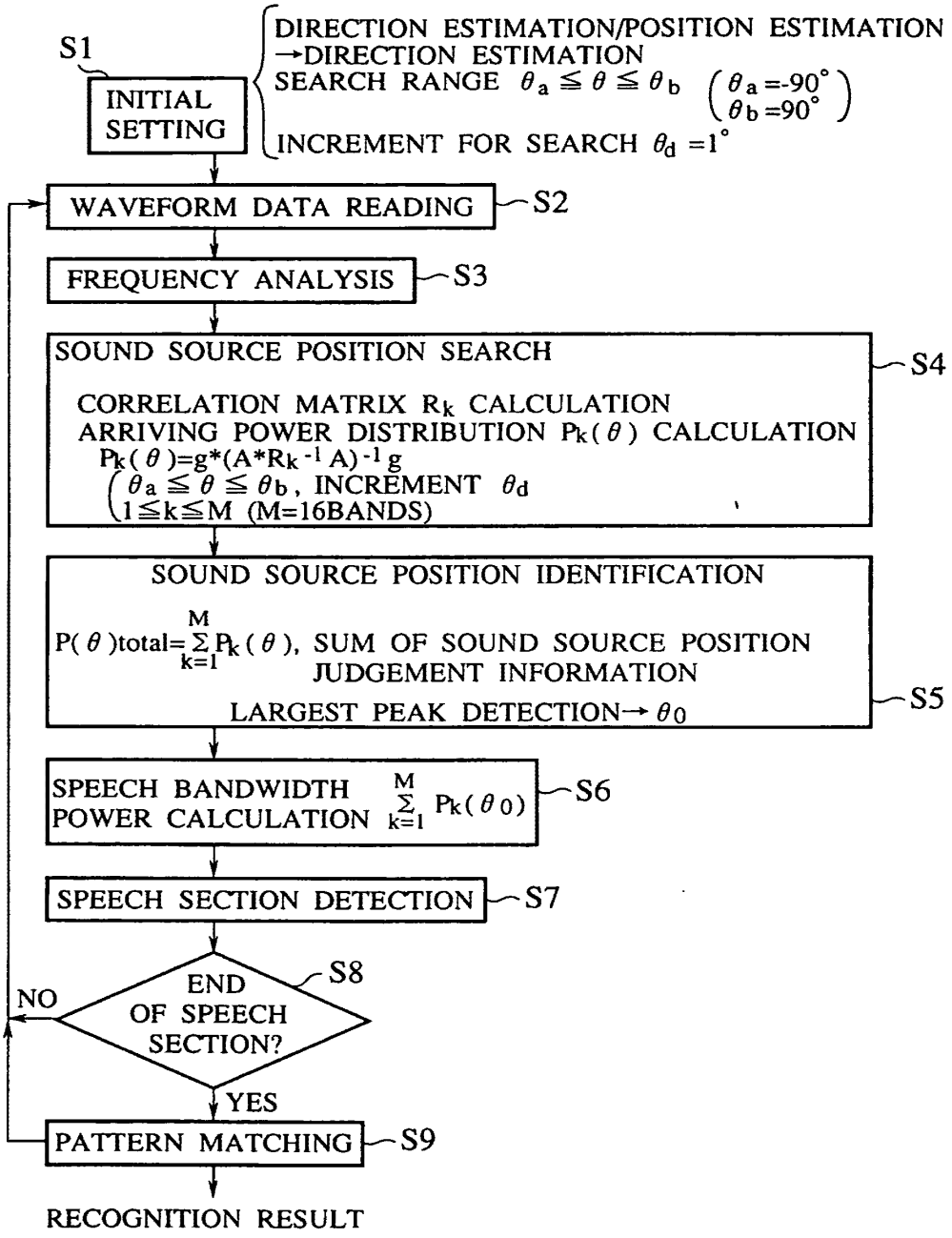

S1
INITIAL SETTING
DIRECTION ESTIMATION/POSITION ESTIMATION
→DIRECTION ESTIMATION
SEARCH RANGE θa ≦ θ ≦ θb (θa=-90° θb=90°)
INCREMENT FOR SEARCH θd =1°
WAVEFORM DATA READING
S2
FREQUENCY ANALYSIS
S3
SOUND SOURCE POSITION SEARCH
CORRELATION MATRIX Rk CALCULATION
ARRIVING POWER DISTRIBUTION Pk(θ) CALCULATION
Pk(θ)=g*(A*Rk-1 A)-1 g
(θa ≦ θ ≦ θb, INCREMENT θd
1≦k≦M (M=16BANDS)
S4
SOUND SOURCE POSITION IDENTIFICATION
P(θ)total=Σ Pk(θ), SUM OF SOUND SOURCE POSITION JUDGEMENT INFORMATION
LARGEST PEAK DETECTION→θ0
S5
SPEECH BANDWIDTH POWER CALCULATION Σ Pk(θ0)
S6
SPEECH SECTION DETECTION
S7
END OF SPEECH SECTION?
S8
NO
YES
PATTERN MATCHING
S9
RECOGNITION RESULT

FIG.10

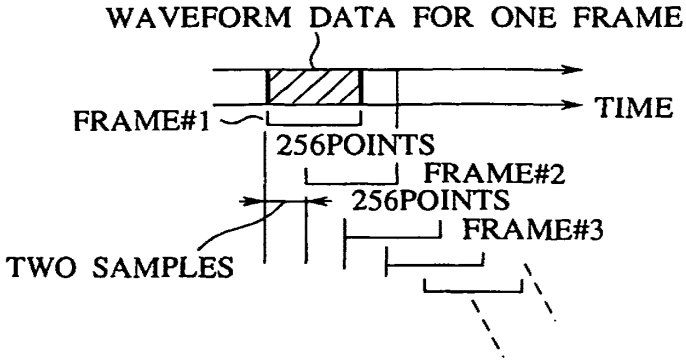


FIG.11A

HIGH RESOLUTION, COARSE SEARCH

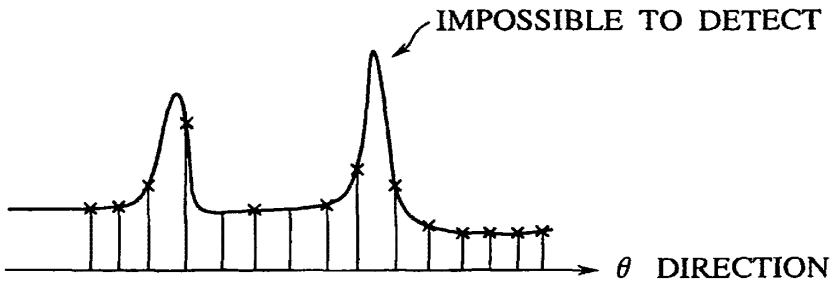


FIG.11B

LOW RESOLUTION, COARSE SEARCH

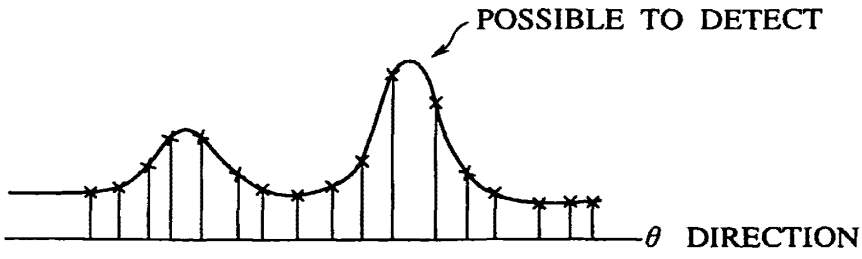


FIG.12

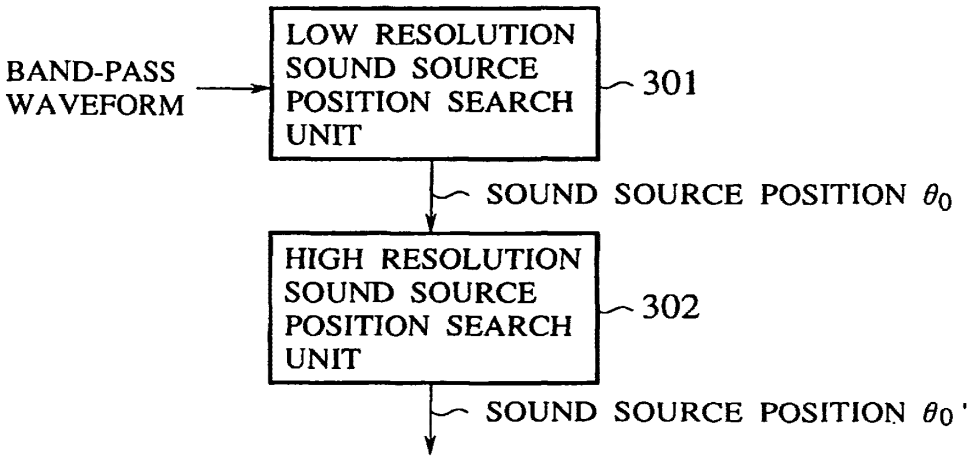
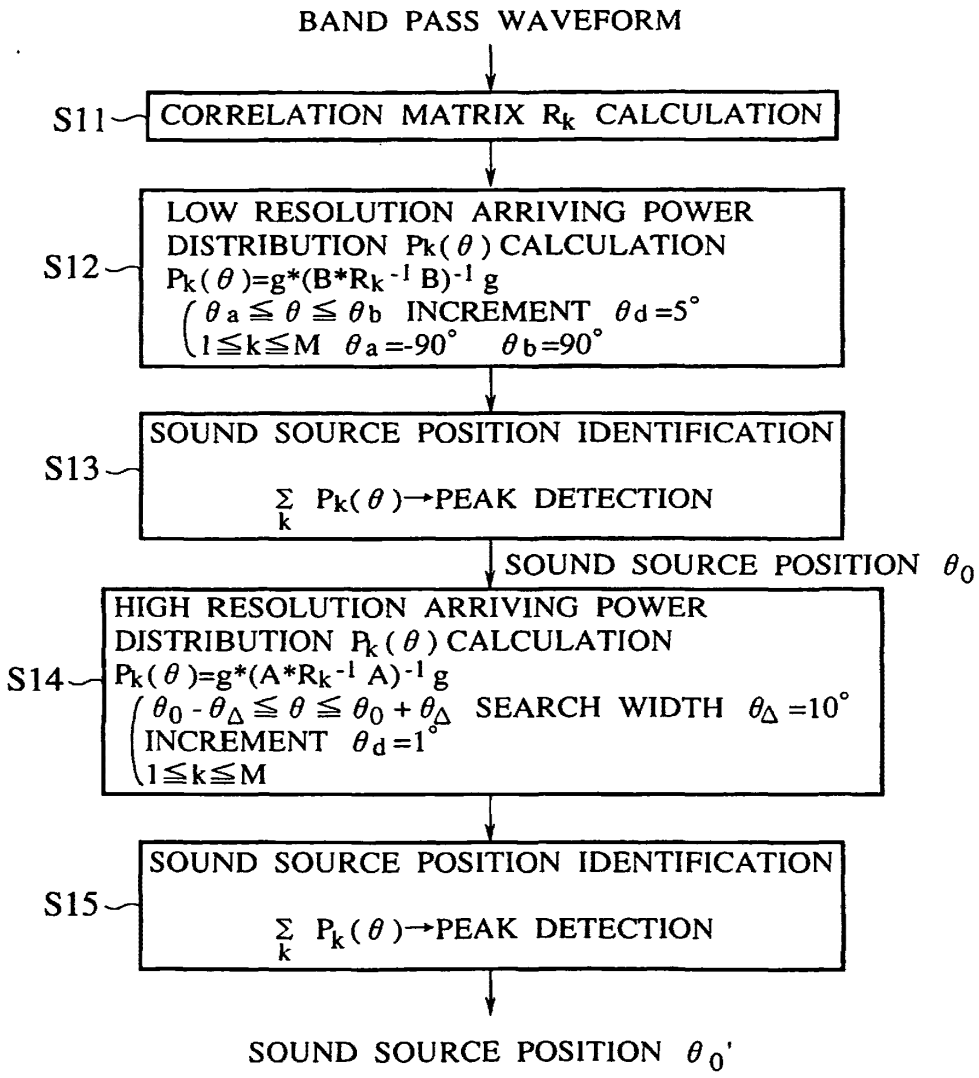

BAND-PASS WAVEFORM
LOW RESOLUTION SOUND SOURCE POSITION SEARCH UNIT
301
SOUND SOURCE POSITION θ_0
HIGH RESOLUTION SOUND SOURCE POSITION SEARCH UNIT
302
SOUND SOURCE POSITION θ_0'

FIG.13


BAND PASS WAVEFORM
S11
CORRELATION MATRIX R_k CALCULATION
S12
LOW RESOLUTION ARRIVING POWER DISTRIBUTION $P_k(\theta)$ CALCULATION
$P_k(\theta)=g^*(B^*R_k^{-1}B)^{-1}g$
$\theta_a \leqq \theta \leqq \theta_b$ INCREMENT $\theta_d=5°$
$1 \leqq k \leqq M$ $\theta_a=-90°$ $\theta_b=90°$
S13
SOUND SOURCE POSITION IDENTIFICATION
$\sum_k P_k(\theta) \rightarrow$PEAK DETECTION
SOUND SOURCE POSITION θ_0
S14
HIGH RESOLUTION ARRIVING POWER DISTRIBUTION $P_k(\theta)$ CALCULATION
$P_k(\theta)=g^*(A^*R_k^{-1}A)^{-1}g$
$\theta_0-\theta_\Delta \leqq \theta \leqq \theta_0+\theta_\Delta$ SEARCH WIDTH $\theta_\Delta=10°$
INCREMENT $\theta_d=1°$
$1 \leqq k \leqq M$
S15
SOUND SOURCE POSITION IDENTIFICATION
$\sum_k P_k(\theta) \rightarrow$PEAK DETECTION
SOUND SOURCE POSITION θ_0'

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

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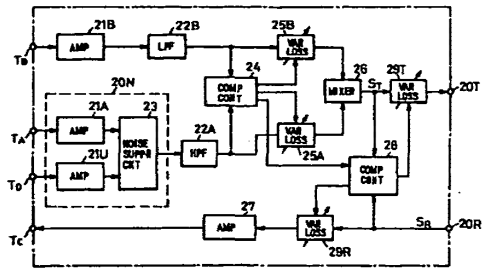
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Remarks:
This application was filed on 30 - 11 - 1999 as a divisional application to the application mentioned under INID code 62.

(54) Transmitter-receiver having ear-piece type acoustic transducer part

(57) A transmitter-receiver comprising: acoustic transducing means composed of a bone-conducted sound pickup microphone (14) for picking up a bone-conducted sound, a directional microphone (15) for picking up an air-conducted sound, an omnidirectional microphone (16) for detecting noise, and a receiver (17) for transducing a received speech signal to a received speech sound; a low-pass filter (22B) which permits the passage therethrough of those low-frequency components in the output from said bone-conducted sound pickup microphone which are lower than a predetermined cutoff frequency; a noise suppressing part (20N) which combines the outputs from said directional microphone (15) and said omnidirectional microphone (16) to suppress a noise component; a high-pass filter (22A) which permits the passage therethrough of those high-frequency components in the output from said noise suppressing part which are higher than said cutoff frequency; a combining circuit (26) which combines the outputs from said low-pass filter and said high-pass filter and outputs a speech sending signal; and means (27) for supplying said received speech signal to said receiver.

FIG. 2



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Description

TECHNICAL FIELD

[0001] The present invention relates to a transmitter-receiver which comprises an ear-piece type acoustic transducing part having a microphone and a receiver formed as a unitary structure and a transmitting-receiving circuit connected to the acoustic transducing part and which permits hands-free communications. More particularly, the invention pertains to a transmitter-receiver which has an air-conducted sound pickup microphone and a bone-conducted sound pickup.

BACKGROUND OF THE INVENTION

[0002] Conventionally, this kind of transmitter-receiver employs, as its ear-piece or ear-set type acoustic transducing part, means which picks up vibrations of the skull caused from talking sound by an acceleration pickup set in the auditory canal (which means will hereinafter be referred to also as a bone-conducted sound pickup microphone and the speech sending signal picked up by this means will hereinafter be referred to as a "bone-conducted sound signal"), or (2) means which guides a speech or talking sound as vibrations of air by a sound pickup tube extending to the vicinity of the mouth and picks up the sound by a microphone set on an ear (which means will hereinafter be referred to also as an air-conducted sound pickup microphone and the speech sending signal picked up by this means will hereinafter be referred to as an "air-conducted sound signal").

[0003] Such a conventional transmitter-receiver of the type which sends speech through utilization of bone conduction is advantageous in that it can be used even in a high-noise environment and permits hands-free communications. However, this transmitter-receiver is not suited to ordinary communications because of its disadvantages that the clarity of articulation of the transmitted speech is so low that the listener cannot easily identify the talker, that the clarity of articulation of the transmitted speech greatly varies from person to person or according to the way of setting the acoustic transducing part on an ear, and that an abnormal sound as by the friction of cords is also picked up. On the other hand, the transmitter-receiver of the type utilizing air conduction is more excellent in clarity than the above but has defects that it is inconvenient to handle when the sound pickup tube is long and that the speech sending signal is readily affected by ambient noise when the tube is short.

[0004] The air-conducted sound pickup microphone picks up sounds having propagated through the air, and hence has a feature that the tone quality of the picked-up speech signals is relatively good but is easily affected by ambient noise. The bone-conducted sound pickup microphone picks up a talker's vocal sound transmitted through the skull into the ear set, and hence has a feature that the tone quality of the picked-up speech signal is relatively low because of large attenuation of components above 1 to 2 kHz but that the speech signal is relatively free from the influence of ambient noise. As a transmitter-receiver assembly for sending excellent speech (acoustic) signals through utilization of the merits of such air-conducted sound pickup microphone and bone-conducted sound pickup microphone, there is disclosed in Japanese Utility Model Registration Application Laid-Open No. 206393/89 a device according to the prior art portion of claim 1 that mixes the speech signal picked up by the air-conducted sound pickup microphone and the speech signal picked up by the bone-conducted sound pickup microphone.

[0005] According to this device, the speech signals from the bone conduction type microphone and the air conduction type microphone are both applied to a low-pass filter and a high-pass filter which have a cutoff frequency of 1 to 2 kHz, then fed to variable attenuators and combined by a mixer into a speech sending signal. With this configuration, low-frequency noises in the output from the air conduction type microphone which are lower than the cutoff frequency are removed, and it is possible to remove or cancel components higher than the cutoff frequency in the noise which the bone conduction type microphone is likely to pick up, such as frictional noise by the friction between a cord extending from the ear set and the human body or clothing, or wind noise by the wind blowing against the ear set. Moreover, in a high-noise environment, the SN ratio of the speech sending signal can be improved by decreasing the attenuation of the bone-conducted sound signal from the low-pass filter and increasing the attenuation of the air-conducted sound signal from the high-pass filter through manual control.

[0006] With this configuration, however, when the level of noise from the air-conducted sound pickup microphone is high, the frequency components higher than the cutoff frequency need to be appreciably attenuated for the purpose of attenuating the noise, and consequently, the speech sending signal is substantially composed only of the bone-conducted sound signal components, and hence is extremely low in tone quality. Moreover, the attenuation control by the variable attenuator is manually effected by an ear set user and the user does not monitor the speech sending signal; hence, it is almost impossible to set the attenuation to the optimum value under circumstances where the amount of noise varies. Furthermore, it is cumbersome to manually control the ratio of combining the speech signal from the air-conducted sound pickup microphone and the speech signal from the bone-conducted sound pickup microphone.

SUMMARY OF THE INVENTION

[0007] It is therefore an object of the present invention to provide a transmitter-receiver which automatically processes the speech sending signal in accordance with use environments (such as the tone quality and the amount of sound) to send speech of the best tone quality.

[0008] This object is achieved with transmitter-receiver as claimed in claim 1. Preferred embodiments are subject-matter of the dependent claims.

[0009] According to the present invention, it is possible to efficiently cancel the noise component in the air-conducted sound by the noise component from the omnidirectionnal microphone and to effectively prevent howling which results from the coupling the speech sending signal and the received speech signal.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010]

Fig. 1 is a sectional view illustrating the configuration of an acoustic transducing part for use in a first embodiment of the present invention;

Fig. 2 is a block diagram illustrating the construction of a transmitting-receiving circuit connected to the acoustic transducing part in Fig. 1;

Fig. 3 is a diagram for explaining the characteristics of a directional microphone and an omnidirectional microphone;

Fig. 4 is a table for explaining control operations of a comparison/control circuit 24 shown in Fig. 2;

Fig. 5 is a block diagram illustrating a transmitter-receiver according to a second embodiment of the present invention;

Fig. 6 is a graph showing the relationship between the tone quality of an air-conducted sound signal and the ambient noise level, and the relationship between the tone quality of a bone-conducted sound signal and the ambient noise level;

Fig. 7 is a graph showing the relationship of the ambient noise level to the level ratio between the bone-conducted sound signal and the air-conducted sound signal in the listening or silent state;

Fig. 8 is a graph showing the relationship of the ambient noise level to the level ratio between the bone-conducted sound signal and the air-conducted sound signal in the talking or double-talking state;

Fig. 9 is a table for explaining operating states of the Fig. 5 embodiment;

Fig. 10A is a blocked diagram showing the construction of a signal mixing circuit which is used as a substitute for each of signal select circuits 33_1 to 33_n in the Fig. 5 embodiment;

Fig. 10B is a graph showing the mixing operation of the circuit shown in Fig. 10A;

Fig. 11 is a block diagram illustrating a modified form of the Fig. 5 embodiment; and

Fig. 12 is a block diagram showing the comparison/control circuit 32 in Fig. 5 or 11 constructed as an analog circuit.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0011] In Fig. 1 there is schematically illustrated the configuration of an ear-piece type acoustic transducing part 10 for use in an embodiment of the present invention. Reference numeral 11 denotes a case of the ear-piece type acoustic transducing part 10 wherein various acoustic transducers described later are housed, 12 a lug or protrusion for insertion into the auditory canal 50, and 13 a sound pickup tube for picking up air-conducted sounds. The sound pickup tube 13 is designed so that it faces the user's mouth when the lug 12 is put in the auditory canal 50; that is, it is adapted to pick up sounds only in a particular direction. The lug 12 and the sound pickup tube 13 are formed as a unitary structure with the case 11.

[0012] Reference numeral 14 denotes an acceleration pickup (hereinafter also referred to as a bone-conducted sound microphone) for picking up bone-conducted sounds, and 15 a directional microphone for picking up air-conducted sounds (i.e. an air-conducted sound pickup microphone), which has such directional characteristics that its sensitivity is high in the direction of the user's mouth (i.e. in the direction of the sound pickup tube 13). The directional microphone 15 has its directivity defined by the combining of sound pressure levels of a sound picked up from the front of the microphone 15 and a sound picked up from behind through a guide hole 11. Accordingly, the directivity could also be obtained even if the sound pickup tube 13 is removed to expose the front of the directional microphone 15 in the surface of the case 11.

[0013] Reference numeral 16 denotes an omnidirectional microphone for detecting noise, which has a sound pickup aperture or opening in the direction opposite to the directional microphone 15. Reference numeral 17 denotes an elec-

tro-acoustic transducer (hereinafter referred to as a receiver) for transducing a received speech signal into a sound, and 18 lead wires for interconnecting the acoustic transducing part 10 and a transmitting-receiving circuit 20 described later; the transmitting-receiving circuit 20 has its terminals T_A, T_B, T_C and T_D connected via the lead wires 18 to the directional microphone 15, the bone-conducted pickup sound microphone 14, the receiver 17 and the omnidirectional microphone 16, respectively.

[0014] In Fig. 2 there is shown in block form the configuration of the transmitting-receiving circuit 20 which is connected to the acoustic transducing part 10 exemplified in Fig. 1. In Fig. 2 terminals T_A, T_B, T_C and T_D are connected to those T_A, T_B, T_C and T_D in Fig. 1, respectively.

[0015] Reference numeral 21B denotes an amplifier for amplifying a bone-conducted sound signal from the microphone 14, and 21A an amplifier for amplifying an air-conducted sound signal from the directional, microphone 15. The gains of the amplifiers 21B and 21A are preset so that their output speech signal levels during a no-noise period are of about the same order at the inputs of a comparison/control circuit 24 described later. Reference numeral 21U denotes an amplifier which amplifies a noise signal from the noise detecting omnidirectional microphone 16 and whose gain is preset so that its noise output during a silent period becomes substantially the same as the noise output level of the amplifier 21A in a noise suppressor circuit 23 described later. The amplifiers 21A and 21B and the noise suppressor circuits 23 constitute a noise suppressing part 20N. The noise suppressor circuit 23 substantially cancels the noise signal by adding together the outputs from the amplifiers 21A and 21U after putting them 180° out of phase to each other.

[0016] Reference numeral 22B denotes a low-pass filter (LPF), which may preferably be one that approximates characteristics inverse to the frequency characteristics of the microphone 14 used; but it may be a simple low-pass filter of a characteristic such that it cuts the high-frequency components of the output signal from the amplifier 21B but passes therethrough the low-frequency components, and its cutoff frequency is selected within the range of 1 to 2 kHz. Reference numeral 22A denotes a high-pass filter (HPF), which may preferably be one that approximates characteristics inverse to the frequency characteristics of the directional microphone 15; but it may be a simple high-pass filter of a characteristic such that it cuts the low-frequency components of the output signal from the noise suppressor circuit 23 and passes therethrough the high-frequency components, and its cutoff frequency is selected within the range of 1 to 2 kHz.

[0017] The directional microphone 15 and the omnidirectional microphone 16 bear such a relationship of sensitivity characteristic that the former has a high sensitivity within a narrow azimuth angle but the latter substantially the same in all directions as indicated by ideal sensitivity characteristics 15S and 16S in Fig. 3, respectively. Then, assuming that the ambient noise level is the same in any directions and at any positions, and letting the total amount of noise energy per unit time applied to the omnidirectional microphone 16 from all directions be represented by the surface area N_U of a sphere with a radius r, the noise energy per unit time applied to the directional microphone 15 is represented by an area N_A defined by the spreading angle of its directional characteristic on the surface of the sphere. Hence, their energy ratio N_A/N_U takes a value sufficiently smaller than one. Now, assume that the amounts of speech energy S_A and S_U applied to the directional microphone 15 and the omnidirectional microphone 16 take the same value S, and let the gains of the amplifiers 21A and 21U be represented by G_A and G_U, respectively. By setting that a value G_AN_A is nearly equal to a value G_UN_U, noise is substantially canceled by the noise suppressor circuit 23 but the speech signal level at the output of the noise suppressor circuit 23 becomes $G_AS\text{-}G_US=G_AS(1\text{-}N_A/N_U)$, since the energy ratio N_A/N_U is sufficiently smaller than one, the speech level is nearly equal to G_AS--this indicates that a speech signal in the air-conducted sound signal can effectively extracted therefrom ideally. The noise suppressing effect that could be achieved by the directional microphone 15, the omnidirectional microphone 16 and the noise suppressing part 20N actually used was typically in the range of 3 to 10 dB.

[0018] In Fig. 2 the bone-conducted sound signal and the air-conducted sound signal, which have their frequency characteristics equalized by the low-pass filter 22B and the high-pass filter 22A, respectively, are applied to the comparison/control circuit 24, wherein their levels V_B and V_A are compared with predetermined reference levels V_{RB} and V_{RA}, respectively. Based on the results of comparison, the comparison/control circuit 24 controls losses L_B and L_A of variable loss circuits 25B and 25A, thereby controlling the levels of the bone- and air-conducted sound signals. A mixer circuit 26 mixes the bone-conducted sound signal and the air-conducted sound signal having passed through the variable loss circuits 25B and 25A. The thus mixed signal is provided as a speech sending signal S_T to a speech sending signal output terminal 20T via a variable loss circuit 29T. A comparison/control circuit 28 compares the level of a speech receiving signal S_R and the level of the speech sending signal S_T with predetermined reference levels V_{RR} and V_{RT}, respectively, and, based on the results of comparison, controls the losses of variable loss circuits 29T and 29R, thereby controlling the levels of the speech sending signal and the speech receiving signal to suppress an echo or howling. The speech receiving signal from the variable loss circuit 29R is amplified by an amplifier 27 to an appropriate level and then applied to the receiver 17 via the terminal T_C.

[0019] Fig. 4 is a table for explaining the control operations of the comparison/control circuit 24 in Fig. 2. The comparison/control circuit 24 compares the output level V_B of the low-pass filter 226 and the output level V_A of the high-pass filter 22A with the predetermined reference levels V_{RB} and V_{RA}, respectively, and determines if the bone- and air-con-

ducted sound signals are present (white circles) or absent (crosses), depending upon whether the output levels are higher or lower than the reference levels. In Fig. 4, state 1 indicates a state in which the bone-conducted sound signal (the output from the low-pass filter 23B) and the air-conducted sound signal (the output from the high-pass filter 23A), both frequency-equalized, are present at the same time, that is, a speech sending or talking state. State 2 indicates a state in which the bone-conducted sound signal is present but the air-conducted sound signal is absent, that is, a state in which the microphone 14 is picking up abnormal sounds such as wind noise of the case 11 and frictional sounds by the lead wires 18 and the human body or clothing. State 3 indicates a state in which the air-conducted sound signal is present but the bone-conducted sound signal is absent, that is, a state in which no speech signal is being sent and the noise component of the ambient sound picked up by the directional microphone 15 which has not been canceled by the noise suppressor circuit 23 is being outputted. State 4 indicates a state in which neither of the bone-and air-conducted sound signals is present, that is, a state in which no speech signal is being sent and no noise is present. The control operations described in the right-hand columns of the Fig. 4 table show the operations which the comparison/control circuit 24 performs with respect to the variable loss circuits 25B and 25A in accordance with the above-mentioned states 1 to 4, respectively.

[0020] Next, a description will be given of the operation of this embodiment of the above construction. When a user of this transmitter-receiver utters a vocal sound with the ear-piece type acoustic transducing part 10 of Fig. 1 put on his or her ear, the vibration of the skull as well as aerial vibration are created by the vibration of the vocal chords. The vibration of the skull is picked up as a bone-conducted sound signal by the microphone 14, from which the signal is provided via the terminal T_B to the amplifier 21B. The aerial vibration of the speech is picked up by the directional microphone 15, from which the signal is provided as an air-conducted sound signal to the amplifier 21A via the terminal T_A.

[0021] In general, as compared with the air-conducted sound, the bone-conducted sound has many low-frequency components, makes less contribution to articulation and contains, in smaller quantity, high-frequency components which are important for the expression of consonants. On the other hand, abnormal sounds such as wind noise by the wind blowing against the case 11 and frictional sound between the cords (lead wires) 18 and the human body or clothing are present in lower and higher frequency bands than the cutoff frequencies of the filters 22A and 22B. Such wind noise and frictional sounds constitute contributing factors to the lack of articulation of the speech sending sound by the bone conduction and the formation of abnormal sounds. On the other hand, "speech" passes through the sound pickup tube 13 and is picked up as an air-conducted sound signal by the directional microphone 15, from which it is applied to the amplifier 21A via the terminal T_A. The air-conducted sound by a talker's speech is a human voice itself, and hence contains frequency components spanning low and high frequency bands.

[0022] In this embodiment, as described in the afore-mentioned Japanese Utility Model Registration Application Laid-Open Gazette, the high-frequency components of the bone-conducted sound from the amplifier 21B are removed by the low-pass filter 228 to extract the low-frequency components alone and this bone-conducted sound signal having the high-frequency components thus cut out therefrom is mixed with an air-conducted sound signal having cut out therefrom the low-frequency components by the high-pass filter 22A. By this, a speech sending signal is generated which has compensated for the degradation of the articulation which would be caused by the lack of the high-frequency components when the speech sending signal is composed only of the bone-conducted sound signal. Besides, according to the present invention, the processing for the generation of such a speech sending signal is automatically controlled to be optimal in accordance with each of the states shown in Fig. 4, by which it is possible to generate a speech sending signal of the best tone quality on the basis of time-varying ambient noise and the speech transmitting-receiving state.

[0023] The noise levels at the directional microphone 15 and the omnidirectional microphone 16 can be regarded as about the same level as referred to previously; but, because of a difference in their directional sensitivity characteristic, the directional microphone 15 picks up a smaller amount of noise energy than does the omnidirectional microphone 16, and hence provides a higher SN ratio. Since the gains G_A and G_U of the amplifiers 21A and 21U are predetermined so that their output noise levels become nearly equal to each other as mentioned previously, the gain G_A of the amplifier 21A is kept sufficiently larger than the gain G_U of the amplifier 21U. Hence, the user's speech signal is amplified by the amplifier 21A with the large gain G_A and takes a level higher than the noise signal level.

[0024] The comparison/control circuit 24 compares, at regular time intervals (1 sec, for instance), the outputs from the low-pass filter 22B (for the bone-conducted sound) and the high-pass filter 22A (for the air-conducted sound) with the reference levels V_{RB} and V_{RA}, respectively, to perform such control operations as shown in Fig. 4. At first, the characteristic of the transmitter-receiver of the present invention immediately after its assembling is adjusted (or initialized) by setting the losses L_B and L_A of the variable loss circuits 25B and 25A to initial values L_{BO} and L_{AO} so that the level of the air-conducted sound signal to be input into the mixer 26 is higher than the level of the bone-conducted sound signal by 3 to 10 dB when no noise is present (State 4 in Fig. 4). The reason for this is that it is preferable in terms of articulation that the air-conducted sound be larger than the bone-conducted one under circumstances where no noise is present.

[0025] Next, a description will be given of the actual state of use in which the levels of the bone-and air-conducted sound signals vary every moment.

(a) When the output (the bone-conducted sound signal) from the low-pass filter 22B is not present (State 3 or 4 in Fig. 4):

The comparison/control circuit 24 compares the output level V_A of the high-pass filter 22A with the reference level V_{RA}. When the output from the high-pass filter 22A is smaller than the reference level V_{RA} (State 4), the comparison/control circuit 24 decides that noise is not present or small and that no talks are being carried out and sets the losses of the variable loss circuits 25B and 25A to the afore-mentioned initial values L_{BO} and L_{AO}, respectively. When this state changes to the talking state (State 1), a mixture of the bone-conducted sound signal composed of law-frequency components and the air-conducted sound signal composed of high-frequency components is provided as the speech sending signal S_T at the output of the mixer circuit 26.

Next, when the output level V_B of the low-pass filter 22B is smaller than the reference level V_{RB} and the output level V_A of the high-pass filter 22A is larger than the reference level V_{RA} (State 3), the comparison/control circuit 24 decides that no talks are being carried out and that ambient noise is large. In this instance, the comparison/control circuit 24 applies a control signal C_A to the variable loss circuit 25A to set its loss L_A to a value larger than the initial value L_{AO} in proportion to the difference between the output level V_A of the high-pass filter 22A and the reference level value V_{RA} as expressed by such an equation as follows:

$$L_A = \lceil (V_A - V_{RA}) / V_M \rceil K + L_{AO} \quad (1)$$

where K is a predetermined constant and the notation $\lceil x \rceil$ represents the smallest integer greater than x. Alternatively, it is possible to increase the loss L_A by a constant K on a stepwise basis each time the level difference (V_A-V_{RA}) increases by a constant V_M, as expressed by the following equation.

$$L_A = K (V_A - V_{RA}) + L_{AO} \quad (2)$$

When the output from the low-pass filter 22B becomes larger than the reference level V_{RB}, that is, when this State 3 changes to the talking state (State 1), the losses of the variable loss circuits 25A and 25B are not changed but are kept at the values set in the immediately preceding State 3. By this, the bone-conducted sound signal composed of low-frequency components and the air-conducted sound signal of the same level as or lower than the level of the bone-conducted sound signal and composed of high-frequency components are mixed by the mixer circuit 26 into the speech sending signal S_T. In this case, it is also possible to hold the loss of the variable loss circuit 25A unchanged and control the loss of the variable loss circuit 25B so that the mixed output level of the mixer circuit 26 takes a predetermined value.

(b) When the output (the bone-conducted sound signal) level V_B of the low-pass filter 22B is larger than the reference level V_{RB} (State 1 or 2 in Fig. 4):

The comparison/control circuit 24 checks the output level V_A of the high-pass filter 22A and, if it is smaller than the reference level V_{RA} (State 2), determines that no talks are being carried out and that the microphone 14 is picking up abnormal sounds. In such an instance, the comparison/control circuit 24 applies a control signal C_B to the variable loss circuit 25B to set its loss L_B to a value greater than the initial value L_{BO} in proportion to the difference between the output level V_B of the low-pass filter 22B and the reference level V_{RA}, as expressed by the following equation.

$$L_B = K (V_B - V_{RB}) + L_{BO} \quad (3)$$

Alternatively, as is the case with the above, the loss L_B may be controlled as expressed by the following equation.

$$L_B = \lceil (V_B - V_{RB}) / V_M \rceil K + L_{BO} \quad (4)$$

When the output level V_A of the high-pass filter 22A becomes larger than the reference level V_{RA}, that is, when this State 2 changes to the talking state (State 1), the losses of the variable loss circuits 25A and 25B are held unchanged, and hence are kept at the values set in the immediately preceding State 2. An air-conducted sound signal composed of high-frequency components and a bone-conducted sound signal of a level set in accordance with the output level V_B of the low-pass filter 22B and composed of low-frequency components are mixed together by the mixer circuit 26. In this instance, it is also possible to hold the loss of the variable loss circuit 25B unchanged and control the loss of the variable loss circuit 25A so that the output level of the mixer circuit 26 may assume the

afore-mentioned predetermined fixed value.

[0026] Next, when the output level V_A of the high-pass filter 22A is larger than the reference level V_{RA} (State 1), the comparison/control circuit 24 decides that the state is the talking state, and causes the variable loss circuits 25B and 25A to hold losses set in the state immediately preceding State 1. As a result, bone- and air-conducted sound signals of levels controlled in accordance with the losses held unchanged are mixed by the mixer circuit 26, which provides the speech sending signal S_T.

[0027] Incidentally, the variable loss circuits 29T and 29R and the comparison/control circuit 28 are provided to suppress the generation of an echo and howling which result from the coupling of the speech sending system and the speech receiving system. The ear-piece type acoustic transducing part 10 has the following two primary contributing factors to the coupling which leads to the generation of howling. First, when the transmitter-receiver assembly is applied to a telephone set, a two-wire/four-wire junction at a telephone station allows the speech sending signal to sneak as an electrical echo into the speech receiving system from the two-wire/four-wire junction, providing the coupling (sidetone) between the two systems. Second, a speech receiving signal is picked up by the bone-conducted sound pickup microphone 14 or directional microphone 15 as a mechanical vibration from the receiver 17 via the case 11--this also provides the coupling between the two systems. Such phenomena also occur in a loudspeaking telephone system which allows its user to communicate through a microphone and a loudspeaker without the need of holding a handset. In this instance, however, the cause of the sneaking of the received sound into the speech sending system is not the mechanical vibration but the acoustic coupling between the microphone and the speaker through the air.

[0028] This problem could be solved by known techniques such as a method for the suppression of howling in the loudspeaking telephone system. The configuration by the comparison/control circuit 28 and the variable loss circuits 29T and 29R is an example of such a prior art. The comparison/control circuit 28 monitors the output level V_T of the mixer circuit 26 and the signal level V_R at a received speech input terminal 20R and, when the speech receiving signal level V_R is larger than a predetermined level V_{RR} and the output level V_T of the mixer circuit 26 is smaller than a predetermined level V_{RT}, the circuit 28 decides that the transmitter-receiver is in the speech receiving state, and sets a predetermined loss L_T in the variable loss circuit 29T, reducing the coupling of the speech receiving signal to the speech sending system. When the output level V_T of the mixer circuit 26 is larger than the predetermined level V_{RT} and the input level V_R at the speech receiving signal input terminal 20R is lower than the predetermined level V_{RR}, the comparison/control circuit 28 decides that the transmitter-receiver is in the talking state, and sets a predetermined loss L_R in the variable loss circuit 29R, suppressing the sidetone from the speech receiving system. When the output level V_T of the mixer circuit 26 and the input level V_R at the speech receiving signal input terminal 20R are higher than the predetermined levels V_{RT} and V_{RR}, respectively, the comparison/control circuit 28 decides that the transmitter-receiver is in a double-talk state, and sets in the variable loss circuits 29T and 29R losses one-half those of the above-mentioned predetermined values L_T and L_R, respectively. In this way, speech with great clarity can be sent to the other party in accordance with the severity of ambient noise and the presence or absence of abnormal noise.

[0029] According to the first embodiment described above, a mixture of the bone-conducted sound signal composed principally of low-frequency components and the air-conducted sound signal composed principally of high-frequency components is used as the speech signal that is sent to the other party. Moreover, the ratio of mixture of the two signals is automatically varied with the magnitude of ambient noise and the abnormal sound picked up by the microphone 14. This permits the implementation of a transmitter-receiver which can be used in a high-noise environment, obviates such defects of the prior art as low clarity or articulation and discomfort by abnormal sound, and allows hands-free communications.

[0030] In the embodiment depicted in Figs. 1 and 2, the comparison/control circuit 24 and the variable loss circuits 25A and 25B may be dispensed with, and even in such a case, the noise level can be appreciably suppressed by the operations of the directional microphone 15, the omnidirectional microphone 16 and the amplifiers 21A and 21B and the noise suppressor circuit 23 which form the noise suppressing part 20N; hence, it is possible to obtain a transmitter-receiver of higher speech quality than in the past. Alternatively, the omnidirectional microphone 16, the amplifier 21U and the noise suppressor circuit 23 may be omitted, and in this case, too, the processing for the generation of the optimum speech sending signal can automatically be performed by the operations of the comparison/control circuit 24, the variable loss circuits 25A and 25B and the mixer circuits 26 in accordance with the states of signals involved.

[0031] Next, a detailed description will be given, with reference to Figs. 5 through 9, of a second embodiment of the transmitter-receiver according to the present invention.

[0032] Fig. 5 illustrates in block form the transmitter-receiver according to the second embodiment of the invention. The bone-conducted sound pickup microphone 14, the directional microphone 15 and the receiver 17 are provided in such an ear-piece type acoustic transducing part 10 as depicted in Fig. 1. In this embodiment, the air-conducted sound signal from the directional microphone (the air-conducted sound pickup microphone 15 and the bone-conducted sound signal from the bone-conducted sound pickup microphone 14 are fed to an air-conducted sound dividing circuit 31A and a bone-conducted sound dividing circuit 31B via the amplifiers 21A and 21B of the transmitting-receiving circuit 20,

respectively. As is the case with Fig. 2, the gains of the amplifiers 21A and 21B are preset so that input air-and bone-conducted sound signals of a vocal sound uttered in a no-noise environment may have about the same level. The air-conducted sound dividing circuit 31A divides the air-conducted sound signal from the directional microphone 15 into first through n-th frequency bands and applies the divided signals to a comparison/control circuit 32 and signal select circuits 33_1 through 33_n. The bone-conducted sound dividing circuit 31B divides the bone-conducted sound signal from the bone-conducted sound pickup microphone 14 into first through n-th frequency bands and applies the divided signals to the comparison/control circuit 32 and the signal select circuits 33_1 through 33_n. In the present invention, the air- and bone-conducted sound signals need not always be divided (i.e. n = 1), but when divided into frequency bands, they are divided, for example, every one or one-third octave, or into high and low bands, or high, intermediate and low bands.

[0033] A received signal dividing circuit 31R divides the received signal S_R from an external line circuit via the input terminal 20R into first through n-th frequency bands and applies the divided signal to the comparison/control circuit 32. In this embodiment, the comparison/control circuit 32 is such one that converts each input signal into a digital signal by an A/D converter (not shown), and performs such comparison and control operations by a CPU (not shown) as described below. That is, the comparison/control circuit 32 calculates an estimated value of the ambient noise level for each frequency band on the basis of the air-conducted sound signals of the respective bands from the air-conducted sound dividing circuit 31A, the bone-conducted sound signals of the respective bands from the bone-conducted sound dividing circuit 31B and the received signals of the respective bands from the received signal dividing circuit 31R. The comparison/control circuit 32 compares the estimated values of the ambient noise levels with a predetermined threshold value (i.e. a reference value for selection) N_{th} and generates control signals C1 to Cn for the respective bands on the basis of the results of comparison. The control signals C1 to Cn thus produced are applied to the signal select circuits 33_1 to 33_n, respectively. The signal select circuits 33_1 to 33_n respond to the control signals C1 to Cn to select the air-conducted sound signals input from the air-conducted sound dividing circuit 31A or the bone-conducted sound signals from the bone-conducted sound signal dividing circuit 31B, which are provided to a signal combining circuit 34. The signal combining circuit 34 combines the input speech signals of the respective frequency bands, taking into account the balance between the respective frequency bands, and provides the combines signal to the speech transmitting output terminal 20T. The output terminal 20T is a terminal which is connected to an external line circuit.

[0034] Fig. 6 is a graph showing, by the solid lines 3A and 3B, a standard or normal relationship between the tone quality (evaluated in terms of the SN ratio or subjective evaluation) of the air-conducted sound signal picked up by the directional microphone 15 and the ambient noise level and a standard or normal relationship between the tone quality of the bone-conducted sound signal picked up by the bone-conducted sound pickup microphone and the ambient noise level. The ordinate represents the tone quality of the sound signals (the SN ratio in the circuit, for instance) and the abscissa the noise level. As indicated by the solid line 3A, the tone quality of the air-conducted sound signal picked up by the directional microphone 15 is greatly affected by the ambient noise level; the tone quality is seriously degraded when the ambient noise level is high. On the other hand, as indicated by the solid line 3B, the tone quality of the bone-conducted sound signal picked up by the bone-conducted sound pickup microphone 14 is relatively free from the influence of the ambient noise level; degradation of the tone quality by the high noise level is relatively small. Hence, the speech sending signal S_T of good tone quality can be generated by setting the noise level at the intersection of the two solid lines 3A and 3B as the threshold value N_{th} and by selecting either one of the air-conducted sound signal picked up by the directional microphone 15 and the bone-conducted sound signal picked up by the bone-conducted sound pickup microphone, depending upon whether the ambient noise level is higher or lower than the threshold value N_{th}. It was experimentally found that the threshold value N_{th} is substantially in the range of 60 to 80 dBA. The characteristics indicated by the solid lines 3A and 3B in Fig. 6 are standard; the characteristics vary within the ranges defined by the broken lines 3A' and 3B' in dependence upon the characteristics of the microphones 14 and 15, the preset gains of the amplifiers 21A and 21B and the frequency characteristics of the input speech signals, but they remain in parallel to the solid lines 3A and 3B, respectively. The solid lines 3A and 3B are substantially straight.

[0035] The relationship between the tone quality of the air-conducted sound signal by the directional micropohone 15 and the ambient noise level and the relationship between the tone quality of the bone-conducted sound signal by the bone-conducted sound pickup microphone 14 and the ambient noise level differ with the respective frequency bands. For this reason, according to this embodiment, the sound signals are each divided into respective frequency bands and either one of the air- and bone-conducted sound signals is selected depending upon whether the measured ambient noise level is higher or lower than a threshold value set for each frequency band--this provides improved tone quality of the speech sending signal.

[0036] To switch between the air- and bone-conducted sound signals in accordance with the ambient noise level, it is necessary to calculate an estimated value of the ambient noise level. Fig. 7 is a graph showing, by the solid line 4BA, a standard relationship of the ambient noise level (on the abscissa) to the level ratio (on the ordinate) between an ambient noise signal picked up by the directional microphone 15 and an ambient noise signal by the bone-conducted sound pickup microphone 14 in the listening or speech receiving or silent duration. Fig. 8 is a graph showing, by the solid line 5BA, a standard relationship of the ambient noise level to the level ratio between a signal (the air-conducted sound sig-

nal plus the ambient noise signal) picked up by the directional microphone 15 and a signal (the bone-conducted sound signal plus the ambient noise signal) by the bane-conducted sound pickup microphone 15 in the talking or double-talking duration. As shown in Figs. 7 and 8, the characteristic in the listening or silent duration and the characteristic in the talking or double-talking duration differ from each other. Hence, the level V_A of the air-conducted sound signal from the directional microphone 15, the level V_B of the bone-conducted sound signal from the bone-conducted sound pickup microphone 15 and the level V_R of the received signal from the amplifier 27 are compared with the reference level values V_{RA}, V_{RB} and V_{RR}, respectively, to determine if the transmitter-receiver is in the listening (or silent) state or in the talking (or double-talking) state. Next, the level ratio V_B/V_A between the bone-conducted sound signal and the air-conducted sound signals picked up by the microphones 14 and 15 in the listening or silent state is calculated, and the noise level at that time is estimated from the level ratio through utilization of the straight line 4BA in Fig. 7. Depending upon whether the estimated noise level is higher or lower than the threshold value N_{th} in Fig. 6, the signal select circuits 33_1 to 33_n each select the bone-conducted sound signal or air-conducted sound signal. Similarly, the level ratio V_B/V_A between the bone-conducted sound signal and the air-conducted sound signal in the talking or double-talking duration is calculated, then the noise level at that time is estimated from the straight line 5BA in Fig. 8, and the bone-conducted sound signal or air-conducted sound signal is similarly selected depending upon whether the estimated noise level is above or below the threshold value N_{th}.

[0037] Next, the operation of the transmitter-receiver will be described. Incidentally, let is be assumed that there are prestored in a memory 32M of the comparison/control circuit 32 the reference level values V_{RA}, V_{RB} and V_{RR}, the threshold value N_{th} and the level ratio vs. noise level relationships shown in Figs.7 and 8. Since the speech signals and the received signals divided into the first through n-th frequency bands are subjected to exactly the same processing until they are input into the signal combining circuit 34, the processing in only one frequency band will be described using reference numerals with no suffixes indicating the band.

[0038] The comparison/control circuit 32 compares, at regular time intervals (of one second, for example), the levels V_A, V_B and V_R of the air-conducted sound signal, the bone-conducted sound signal and the received signal input from the air-conducted sound dividing circuit 31A, the bone-conducted sound dividing circuit 31B and the received signal dividing circuit 31R with the predetermined reference level values V_{RA}, V_{RB} and V_{RR}, respectively. When the level V_R of the received signal S_R is higher than the predetermined value V_{RR} and the level V_A of the air-conducted sound signal picked up by the directional microphone 15 and the level V_B of the bone-conducted sound signal by the bone-conducted sound pickup microphone 14 are smaller than the predetermined values V_{RA} and V_{RB}, respectively, the comparison/control circuit 32 determines that this state is the listening state shown in the table of Fig. 9. When the level V_R of the received signal level V_R is smaller than the predetermined value V_{RR} and the levels V_A and V_B of the air-conducted sound signal and the bone-conducted sound signal are both smaller than the predetermined values V_{RA} and V_{RB}, the circuit 32 determines that this state is the silent state. In these two states the comparison/control circuit 32 calculates the level ratio V_B/V_A between the air-conducted sound signal from the air-conducted sound dividing circuit 31A and the bone-conducted sound signal from the bone-conducted sound dividing circuit 31B. Based on the value of this level ration, the comparison/control circuit 32 refers to the relationship of Fig. 7 stored in the memory 32M to obtain an estimated value of the corresponding ambient noise level. When the estimated value of the ambient noise level is smaller than the threshold value N_{th} shown in Fig. 6, the comparison/control circuit 32 supplies the signal select circuit 33 with a control signal C instructing it to select and output the air-conducted sound signal input from the air-conducted sound dividing circuit 31A. When the estimated value of the ambient noise level is greater than the threshold value N_{th}, the comparison/control circuit 32 applied th control signal C to the signal select circuit 33 to instruct it to select and output the bone-conducted sound signal input from the bone-conducted sound dividing circuit 31B.

[0039] On the other hand, when the received signal level V_R is smaller than the reference level value V_{RR} and the levels V_A and V_B of the air-conducted sound signal by the directional microphone 15 and the bone-conducted sound signal by the bone-conducted sound pickup microphone 14 are larger then the predetermined reference level values V_{RA} and V_{RB}, the comparison/control circuit 32 determines that this state is the talking state shown in the table of Fig. 9. When the received signal level V_R is larger than the reference level value V_{RR} and the levels V_A and V_B of the air-conducted sound signal and the bone-conducted sound signal are larger than the predetermined reference level values V_{RA} and V_{RB}, the comparison/control circuit 32 determines that this state is the double-talking state. In these two states the comparison/control circuit 32 calculates the level ratio V_B/V_A between the bone-conducted sound signal and the air-conducted sound signal and estimates the ambient noise level N through utilization of the relationship of Fig. 8 stored in the memory 32M.

[0040] When the thus estimated value of the ambient noise level N is smaller than the threshold value N_{th} shown in Fig. 6, the comparison/control circuit 32 applies the control signal C to the signal select circuit 33 to cause it to select and output the air-conducted sound signal input from the air-conducted sound dividing circuit 31A. When the estimated value N of the ambient noise level is greater than the threshold value N_{th}, the circuit 32 applies the control signal C to the signal select circuit 33 to cause it to select and output the bone-conducted sound signal input from the bone-conducted sound dividing circuit 31B.

[0041] The comparison/control circuit 32 has, in the memory 32M for each of the first through n-th frequency bands, the predetermined threshold value N_{th} shown in Fig. 6 and the level ratio vs. noise level relationships representing the straight characteristic lines 4BA and 5BA shown in Figs. 7 and 8. The comparison/control circuit 32 performs the same processing as mentioned above and applies the resulting control signals C1 to Cn to the signal select circuits 33_1 to 33_n. The signal combining circuit 34 combines the speech signals from the signal select circuits 33_1 to 33_n, taking into account the balance between the respective frequency bands.

[0042] While in the above the embodiments have been described to estimate and compare the noise level with the threshold value and control the signal select circuits 33_1 to 33_n accordingly in any state described in the table of Fig. 9, it is also possible to employ a scheme that estimates the noise level only in the silent or listening state and uses the thus estimated noise level to effect control in the talking state and the double-taking state. In such an instance, the characteristic data of Fig. 8 need not be stored in the memory 32M. In contrast to this, the estimation of the noise level may be made only in the talking or double-talking state, in which case the estimated noise level is used for control in the talking or double-talking state. In this instance, the characteristic data of Fig. 7 is not needed.

[0043] Incidentally, the double-talking duration and the silent duration are shorter than the talking or listening duration. Advantage may also be taken of this to effect control in the double-talking state and in the silent state by use of the ambient noise level estimated prior to these states.

[0044] When the level of the bone-conducted sound signal picked up by the bone-conducted sound pickup microphone 14 is abnormally high, it can be considered that noise is made by the friction of cords or the like; hence, it is effective to select the air-conducted sound signal picked up by the directional microphone 15.

[0045] In the case where the estimated noise level N is compared with the threshold value N_{th} for each frequency band and the air-conducted sound signal picked up by the directional microphone 15 is switched to the bone-conducted sound signal by the bone-conducted sound pickup microphone 14 on the basis of the result of comparison as described previously with reference to the Fig. 5 embodiment, the timbre of the speech being sent may sometimes undergo an abrupt change, making the speech unnatural. To solve this problem, an area N_W of a fixed width as indicated by N^- and N^+ is provided about the threshold value N_{th} of the ambient noise level shown in Fig. 6; when the estimated noise level N is within the area N_W, the air-conducted sound signal from the directional microphone 15 and the bone-conducted sound signal from the bone-conducted sound pickup microphone 14 are mixed in a ratio corresponding to the noise level, and when the estimated noise level N is larger than the area N_W, the bone-conducted sound signal is selected, and when the estimated noise level is smaller than the area N_W, the air-conducted sound signal is selected. By this, it is possible to reduce the abrupt change in the timbre prior to or subsequent to the switching operation.

[0046] The modification of the Fig. 5 embodiment for such signal processing can be effected by using, for example, a signal mixer circuit 33 depicted in Fig. 10A in place of each of the signal select circuits 33_1 to 33_n. In this example, the corresponding air-conducted sound signal and bone-conducted sound signal of each frequency band are applied to variable loss circuits 33A and 33B, respectively, wherein they are given losses L_A and L_B set by control signals C_A and C_B from the comparison/control circuit 32. The both signals are mixed in a mixer 33C and the mixed signal is applied to the signal combining circuit 34 in Fig. 5.

[0047] The losses L_A and L_B for the air-conducted sound signal and the bone-conducted sound signal in the area N_W need only to be determined as shown in Fig. 10B, for instance. For the brevity's sake, setting $N_{th} = (N^+ + N^-)/2$, the area width to $D = N^+ - N^-$, the minimum values L_{AO} and L_{BO} of the losses L_A and L_B to 0 dB, respectively, and their maximum values L_{AMAX} and L_{BMAX} to the same L_{MAX} dB, the loss L_A in the area N_W can be expressed, for example, by the following equation.

$$L_A = \frac{1}{2}(L_{AMAX} - L_{AO}) + \frac{L_{AMAX} - L_{AO}}{N^+ - N^-}(N + N_{th}) + L_{AO} \qquad (5)$$
$$= L_{MAX}\left\{\frac{1}{2} + \frac{N - N_{th}}{D}\right\}$$

[0048] Similarly, the loss L_B can be expressed by the following equation.

$$L_B = L_{MAX}\left\{\frac{1}{2} + \frac{N - N_{th}}{D}\right\} \qquad (6)$$

[0049] The value of the maximum loss L_{MAX} is selected in the range of between 20 and 40 dB, and the width D of the area N_W is set to about 20 dB, for instance. When the estimated noise level N is larger than the area N_W, the bone-con-

ducted sound signal is given any loss (L_B = 0) and is applied intact to the mixer 33C. On the other hand, the air-conducted sound signal is not given the loss L_{MAX} but instead the variable loss circuit 33A is opened to cut off the signal. Similarly, when the estimated noise level N is smaller than the area N_W, the air-conducted sound signal is not given any loss (L_A = 0) and is fed intact to the mixer 33C, whereas the bone-conducted sound signal is cut off by opening the variable loss circuit 33B. The comparison/control circuit 32 determines the losses L_A and L_B for each band as described and sets the losses in the variable loss circuits 33A and 33B by the control signals C_A and C_B.

[0050] With such signal processing as described above, it is possible to provide smooth timbre variations of the speech being sent when the air-conducted sound signal is switched to the bone-conducted sound signal or vice versa. Moreover, if the levels of the air-conducted sound signal and the bone-conducted sound signal input into the variable loss circuits 33A and 33B are nearly equal to each other, the output level of the mixer 33C is held substantially constant before and after the switching between the air- and bone-conducted sound signals and the output level in the area N_W is also held substantially constant, ensuring smooth signal switching. Incidentally, the signal select processing by the signal select circuits 33_1 to 33_n, in Fig. 5 corresponds to the case where the width D of the area N_W is set to zero in the processing in the modified embodiment depicted in Figs. 10A and 10B. Hence, it can be said, in a broad sense, that the signal select circuits 33_1 to 33_n, also contribute to the mixing of signals on the basis of the estimated noise level.

[0051] In the above, when the estimation of the ambient noise level may be rough, it can be estimated by using average values of the characteristics shown in Figs. 7 and 8. In this instance, the received signal dividing circuit 31R can be dispensed with. When the estimation of the ambient noise level may be rough, it can also be estimated by using only the speech signal from the directional microphone 14.

[0052] Fig. 11 illustrates in block form a modified form of the Fig. 5 embodiment, in which as is the case with the first embodiment of Figs. 1 and 2, the omnidirectional microphone 16, the amplifier 21U and the noise suppressing circuit 23 are provided in association with the direction microphone 15 and the output from the noise suppressing circuit 23 is fed as an air-conducted sound signal to the air-conducted sound dividing circuit 31A. This embodiment is identical in construction with the Fig. 5 embodiment except the above. In this embodiment, when the transmitter-receiver is in the silent or listening state, a switch 35 is opened and only the air-conducted sound signal provided via the amplifier 21U from the omnidirectional microphone 16 is applied to the noise suppressing circuit 23, from which it is fed intact to the air-conducted sound dividing circuit 31A, and the air-conducted sound signals divided into respective frequency bands are applied to the comparison/control circuit 32. As in the Fig. 5 embodiment, the comparison/control circuit 32 estimates the ambient noise levels through utilization of the relationships shown in Fig. 7 and, based on the estimated levels, generate the control signals C1 to Cn for signal selection (or mixing use in the case of using the Fig. 10A circuit configuration), which are applied to the signal select circuits 33_1 to 33_n (or the signal mixing circuit 36). After this, the switch 35 is turned ON to pass therethrough the air-conducted sound signal from the directional microphone 15 to the noise suppressing circuit 23, in which its noise components are suppressed, and then the air-conducted sound signal is fed to the air-conducted sound dividing circuit 31A. This is followed by the speech sending signal processing by the same signal selection or mixing as described previously with respect to Fig. 5.

[0053] Although in the embodiments of Figs. 5 and 11 the comparison/control circuit 32 has been described to convert the signals input thereto to digital signals and generate the control signals C1 to Cn on the basis of the level ratio-noise level relationships stored in the memory 32M, the comparison/control circuit 32 may also be formed as an analog circuit, for example, as depicted in Fig. 12. In Fig. 12 there is shown in block form only a circuit portion corresponding to one of the divided subbands. A pair of corresponding subband signals from the air-conducted sound signal dividing circuit 31A and the bone-conducted sound signal dividing circuit 31B are both applied to a level ratio circuit 32A and a comparison/logic state circuit 32E. The level ratio circuit 32A calculates the level ratio L_B/L_A between the bone- and air-conducted sound signals in an analog fashion and supplies level converter circuits 32B and 32C with a signal of a level corresponding to the calculated level ratio.

[0054] The level converter circuit 32B performs a level conversion based on the relationship shown in Fig. 7. That is, when supplied with the level ratio V_B/V_A, the level converter circuit 32B outputs an estimated noise level N corresponding thereto and provides it to a select circuit 32D. Similarly, the level converter circuit 32C performs a level conversion based on the relationship shown in Fig. 8. That is, when supplied with the level ratio V_B/V_A, the level converter circuit 32C outputs an estimated noise level corresponding thereto and provides it to the select circuit 32D. On the other hand, the comparison/state logic circuit 32E compares the levels of the corresponding air- and bone-conducted sound signals of the same subband and the level of the received speech signal with the reference levels V_{RA}, V_{RB} and V_{RR}, respectively, to make a check to see if these signals are present. Based on the results of these checks, the comparison/state logic circuit 32E applies a select control signal to the select circuit 32D to cause it to select the output from the level converter circuit 32B in the case of State 1 or 2 shown in the table of Fig. 9 and the output from the level converter circuit 32C in the case of State 3 or 4.

[0055] The select circuit 32D supplies a comparator circuit 32F with the estimated noise level N selected in response to the select control signal. The comparator circuit 32F compares the estimated noise level N with the threshold level N_{th} and provides the result of the comparison, as a control signal C for the subband concerned, to the corresponding

one of the signal select circuits 311 to 31n in Fig. 5 or 11. In this instance, it is also possible to make a check to determine if the estimated noise level N is within the area N_W or high or lower than it as described previously with respect to Fig. 10B, instead of comparing the estimated noise level N with the threshold value N_{th}; if the estimated noise level N is within the area N_W, the control signals C_A and C_B corresponding to the difference between the estimated noise level N and the threshold level N_{th}, as is the case with Eqs. (5) and (6), are applied to the signal mixing circuit of the Fig. 10A configuration to cause it to mix the air-conducted sound signal and the bone-conducted sound signal; when the estimated noise level N is higher than the area N_W, the bone-conducted sound signal is selected and when the estimated noise level N is lower than the area N_W, the air-conducted sound signal is selected.

[0056] As described above, according to the transmitter-receiver of the embodiment shown in each of Figs. 5 and 11 the air-conducted sound signal picked up by the directional microphone and the bone-conducted sound signal by the bone-conducted sound pickup microphone are used to estimate the ambient noise level and, on the basis of the magnitude of the estimated noise level, either one of the air-conducted sound signal and the bone-conducted sound signal is selected or both of the signals are mixed together, whereby a speech sending signal of the best tone quality can be generated. Thus, the communication device of the present invention is able to transmit speech sending signals of excellent tone quality, precisely reflecting the severity and amount of ambient noise regardless of whether the device is in the talking or listening state.

[0057] While in the first and second embodiments the transmitting-receiving circuit 20 is described to be provided outside the case 11 of the ear-piece type acoustic transducing part 10 and connected thereto via the cord 18, it is evident that the transmitting-receiving circuit 20 may be provided in the case 11 of the acoustic transducing part 10.

[0058] It will be apparent that many modifications and variations may be effected without departing from the scope of the novel concepts of the present invention.

Claims

1. A transmitter-receiver comprising:

   acoustic transducing means composed of a bone-conducted sound pickup microphone (14) for picking up a bone-conducted sound, a directional microphone (15) for picking up an air-conducted sound, an omnidirectional microphone (16) for detecting noise, and a receiver (17) for transducing a received speech signal to a received speech sound;
   a low-pass filter (22B) which permits the passage therethrough of those low-frequency components in the output from said bone-conducted sound pickup microphone which are lower than a predetermined cutoff frequency;
   a high-pass filter (22A) which permits the passage therethrough of those high-frequency components in the outputs from said directional microphone (15) and said omnidirectional microphone (16) which are higher than said cutoff frequency;
   a combining circuit (26) which combines the outputs from said low-pass filter and said high-pass filter and outputs a speech sending signal; and
   means (27) for supplying said received speech signal to said receiver;
   characterized in that
   a noise suppressing part (20N) which combines the outputs from said directional microphone (15) and said omnidirectional microphone (16) to suppress a noise component, wherein said high-pass filter (22A) passes the high-frequency components in the output from said noise suppressing part to said combining circuit (26).

2. The transmitter-receiver of claim 1 , which further comprises: first and second variable loss circuits (29T, 29R) connected to the output side of said combining circuit and the input side of said received speech signal supplying means, for controlling the levels of said speech sending signal and said received speech signal, respectively; and a comparison/control circuit (28) which compares the level of said speech sending signal fed to said first variable loss circuit (29T) and the level of said received speech signal fed to said second variable loss circuit (29R) with predetermined first and second reference levels, respectively, and on the basis of the results of comparison, controls the losses that are set in said first and second variable loss circuits (29T, 29R).

3. The transmitter-receiver of claim 1 or 2, wherein said noise suppressing part (20N) comprises: a first amplifier (21A) for amplifying said air-conducted sound signal; a second amplifier (21U) for amplifying said noise components detected by said omnidirectional microphone (16); and a noise suppressor circuit (23) which adds together the outputs from said first and second amplifiers in a 180° out-of-phase relation to each other to generate an air-conducted sound signal with said noise components suppressed and applies it to said high-pass filter (22A).

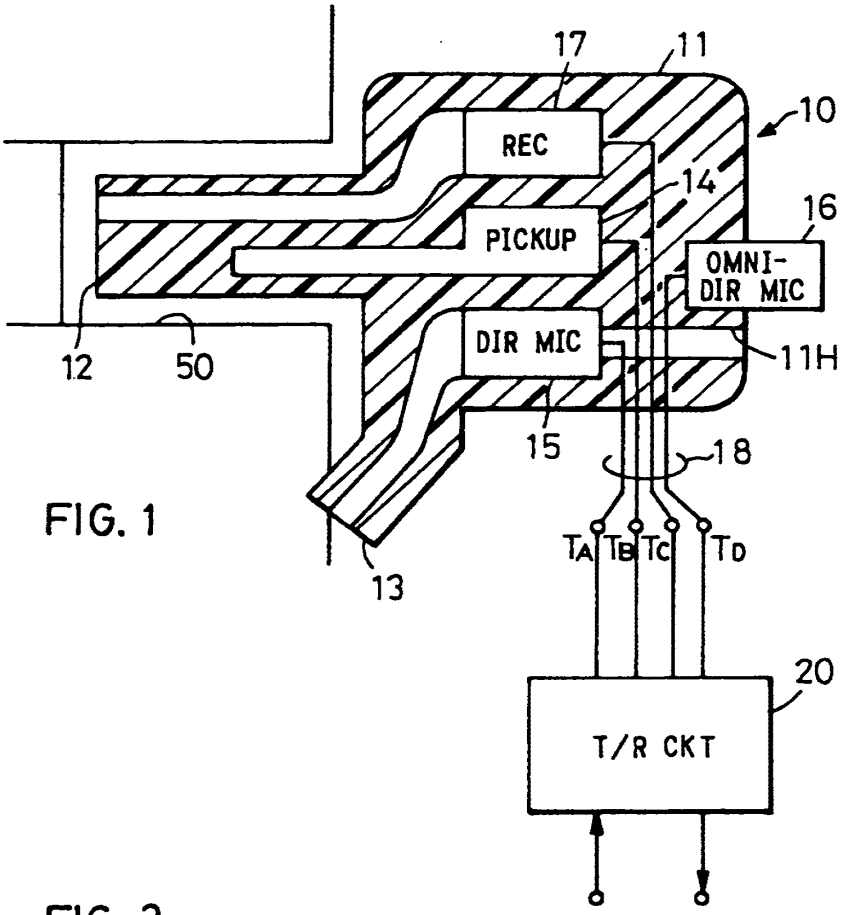

17
11
10
REC
14
16
PICKUP
OMNI-
DIR MIC
12
50
DIR MIC
11H
15
18
TA
TB
TC
TD
13
20
T/R CKT

FIG. 1

FIG. 3

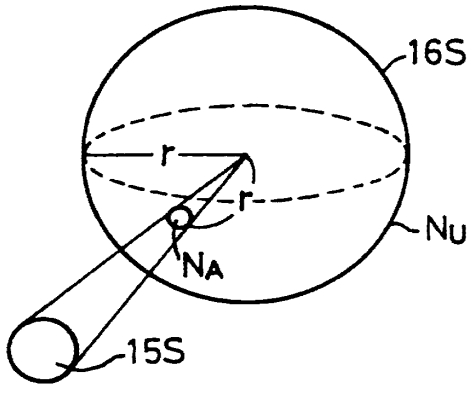

16S
r
r
Nu
NA
15S

FIG. 2

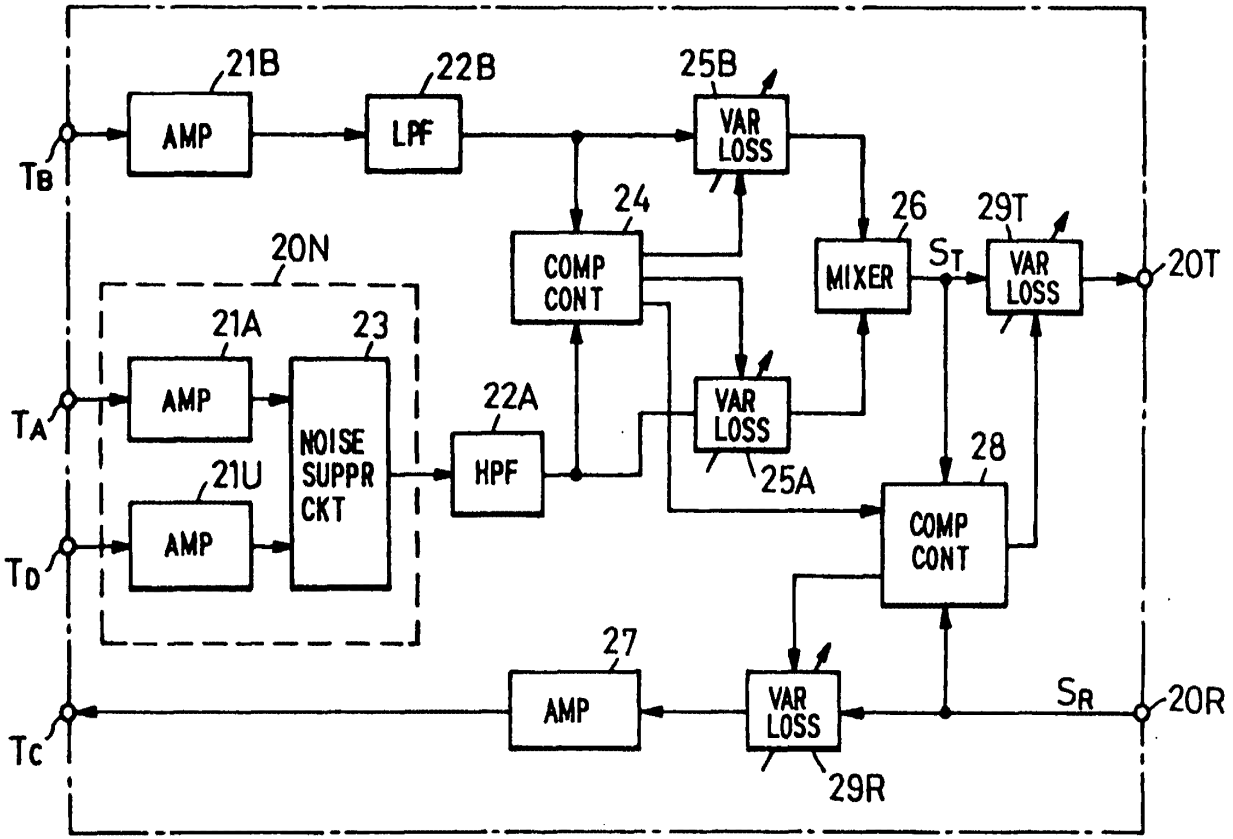
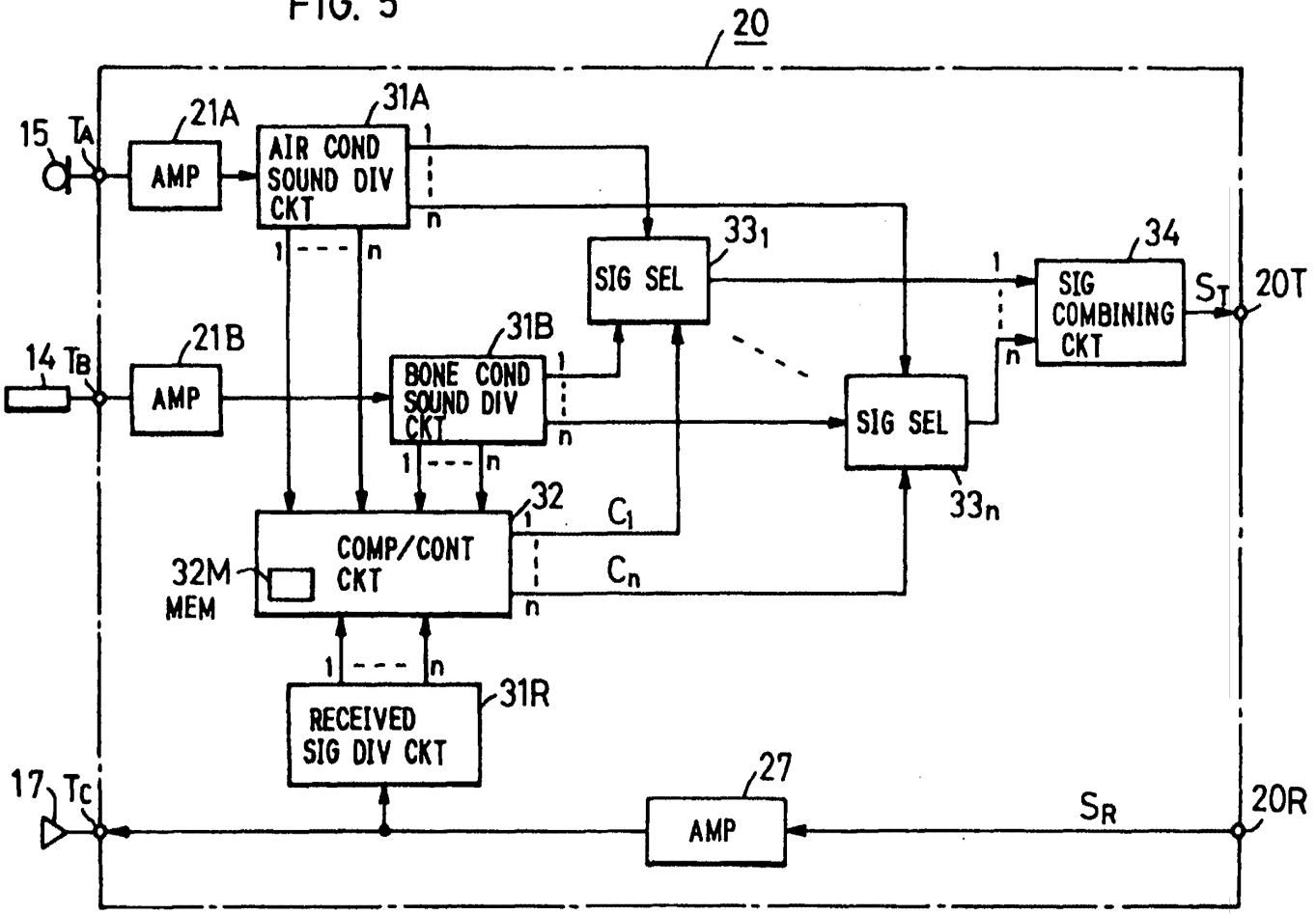

21B
22B
25B
AMP
LPF
VAR LOSS
TB
24
26
29T
20N
COMP CONT
MIXER
ST
VAR LOSS
20T
21A
23
AMP
NOISE SUPPR CKT
22A
VAR LOSS
TA
21U
HPF
25A
28
AMP
COMP CONT
TD
27
AMP
VAR LOSS
SR
20R
TC
29R

FIG. 4

STATE	PRESENCE OR ABSENCE OF SIGNAL		CONTROL OPERATIONS	
	LPF OUTPUT	HPF OUTPUT	LOSS BY 25B	LOSS BY 25A
1	O	O	MAINTAIN SET VALUE OF THE PRECEDING STATE	MAINTAIN SET VALUE OF THE PRECEDING STATE
2	O	×	INCREASE LOSS L_B FROM L_{BO} ACCORDING TO V_B	SET INITIAL VALUE L_{AO}
3	×	O	SET INITIAL VALUE L_{BO}	INCREASE LOSS L_A FROM L_{AO} ACCORDING TO V_A
4	×	×	SET INITIAL VALUE L_{BO}	SET INITIAL VALUE L_{AO}

O: PRESENCE ×: ABSENCE

FIG. 5



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FIG. 6

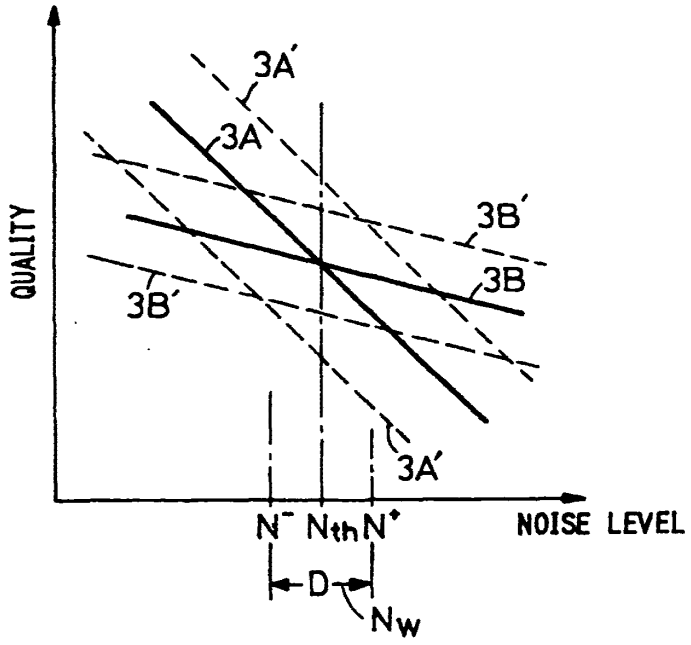

3A'
3A
3B'
3B
3B'
3A'
QUALITY
N⁻ Nth N⁺
NOISE LEVEL
D
Nw

FIG. 7

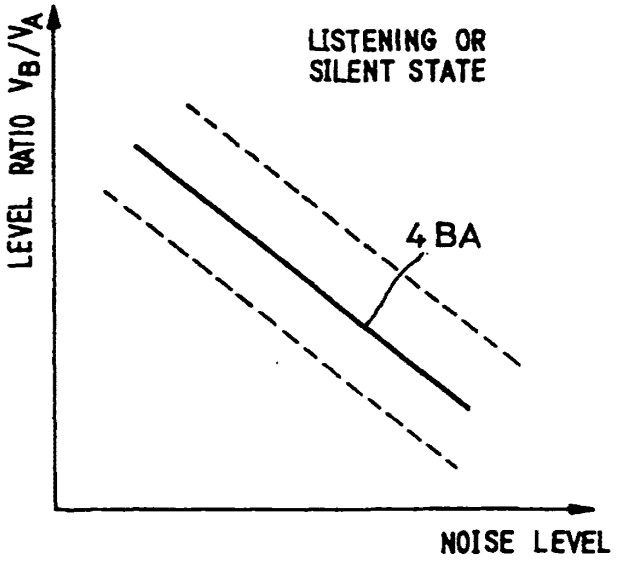

LISTENING OR
SILENT STATE
LEVEL RATIO VB/VA
4BA
NOISE LEVEL

FIG. 8

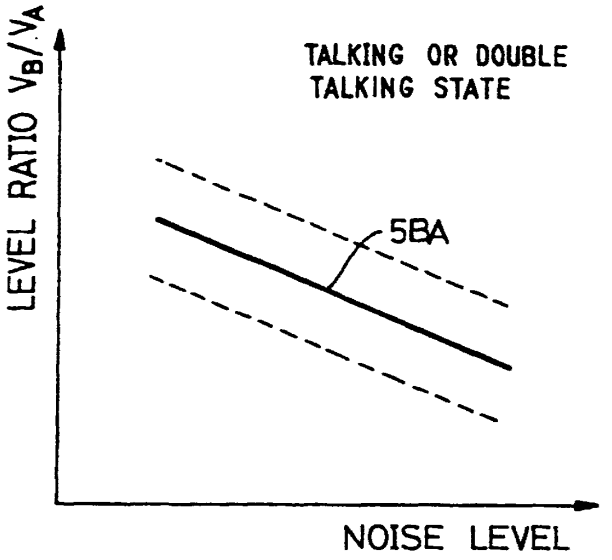


FIG. 9

	RECEIVING SIG	AIR COND SOUND	BONE COND SOUND	
1 LISTENING STATE	○	×	×	USE FIG.7
2 SILENT STATE	×	×	×	
3 TALKING STATE	×	○	○	USE FIG.8
4 DOUBLE-TALK STATE	○	○	○	

× : ABSENCE ○ : PRESENCE

FIG. 10A

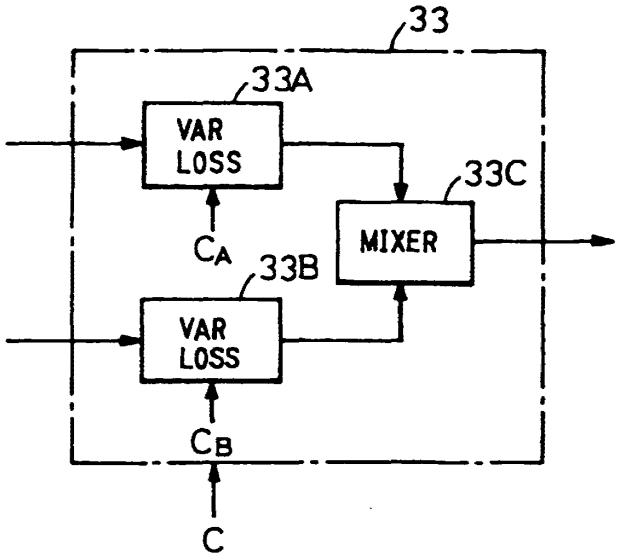


FIG. 10B

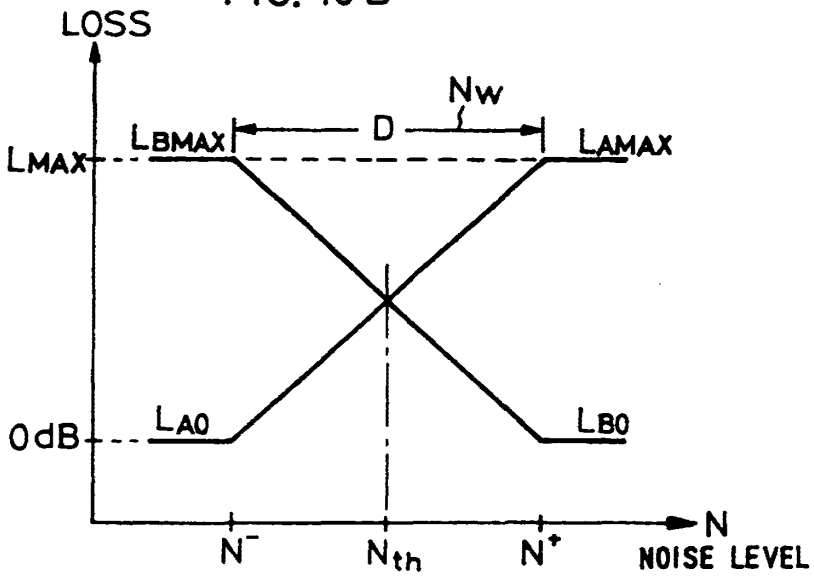
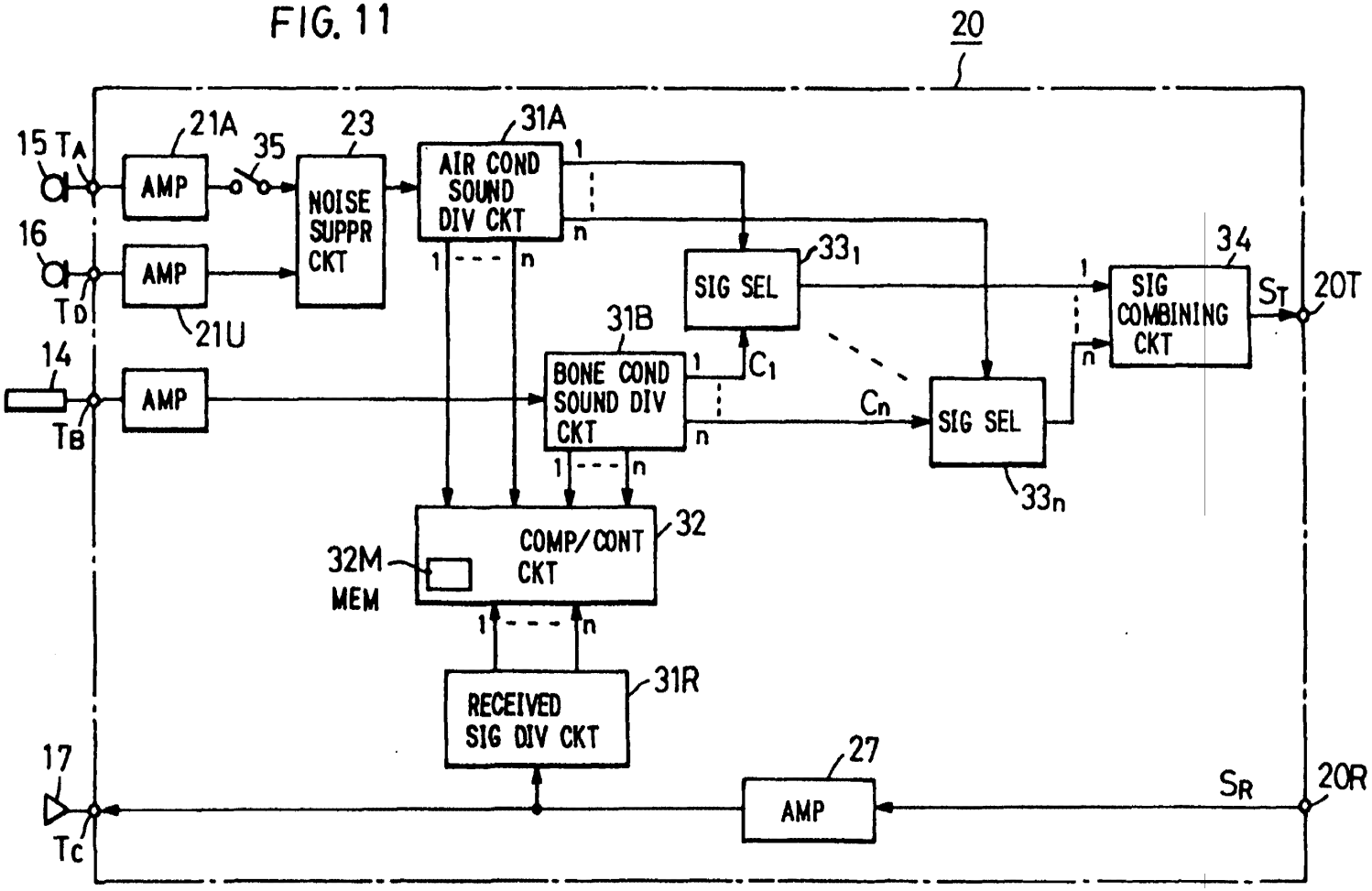
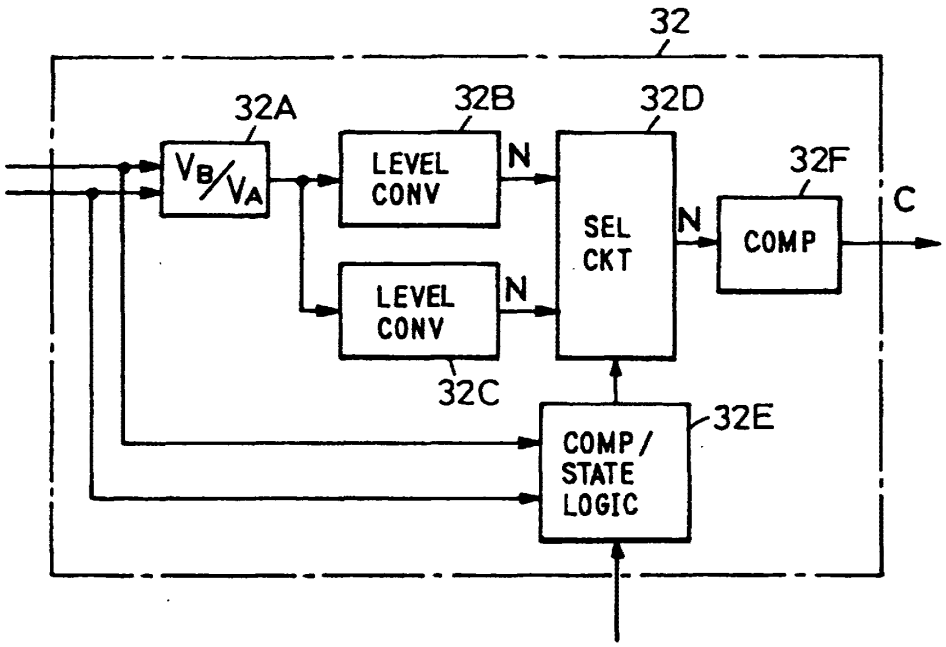


FIG. 11



EP 0 984 660 A2

FIG. 12


32
32A
32B
32D
32F
V_B/V_A
LEVEL CONV
N
SEL CKT
N
COMP
C
LEVEL CONV
N
32C
32E
COMP / STATE LOGIC

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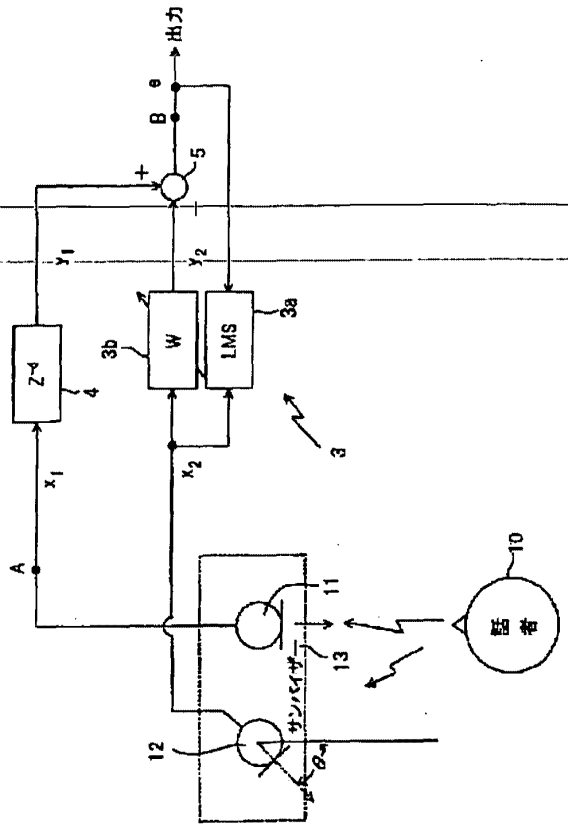
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TITLE : MICROPHONE SYSTEM



ABSTRACT : PROBLEM TO BE SOLVED: To enhance the S/N of a voice signal in a noise reduction system where two microphones are in use.

SOLUTION: In the microphone system that uses output signals of two microphones to output a talker voice signal with the enhanced S/N through adaptive signal processing, the two microphones 11, 12 with directivity are closely arranged and an angle between the directivity of the microphone and a voice utterance direction of a talker differs between the microphones. For example, each of the microphones 11, 12 is mounted on a sunvisor 13 of a vehicle while differentiating the directivity or mounted on a driver's assistant seat or a ceiling at a driver's seat of the vehicle.

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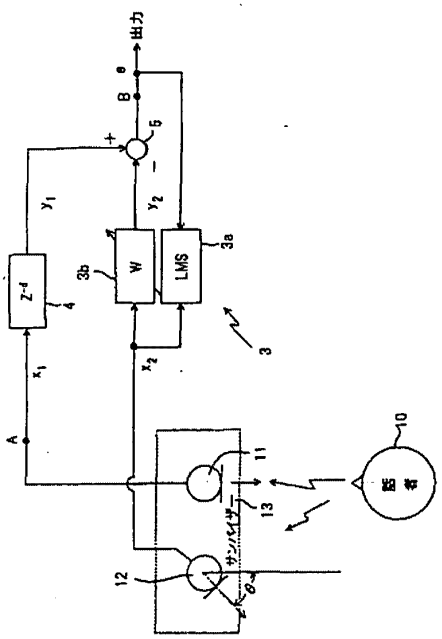
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最終頁に続く

(54)【発明の名称】　マイクロホンシステム

(57)【要約】
【課題】　マイクロホンを２つ使用するノイズリダクションシステムにおいて音声信号のSN比を改善する。
【解決手段】　２つマイクロホンの出力信号を用いて適応信号処理を行いSN比を改善した話者音声信号を出力するマイクロホンシステムにおいて、指向特性を有する２つのマイクロホン１１，１２を接近して配置すると共に、マイクロホンの指向方向と話者の発声方向がなす角度をマイクロホン毎に異ならせる。例えば、各マイクロホン１１，１２の指向方向を異ならせてを車両のサンバイザー１３に取り付け、あるいは、車両の助手席あるいは運転席側天井に取り付ける。

本発明のマイクロホンシステム



【特許請求の範囲】

【請求項１】　２つマイクロホンの出力信号を用いて適応信号処理を行いSN比を改善した話者音声信号を出力するマイクロホンシステムにおいて、
指向特性を有する２つのマイクロホンを接近して配置すると共に、マイクロホンの指向方向と話者の発声方向がなす角度をマイクロホン毎に異ならせることを特徴とするマイクロホンシステム。

【請求項２】　前記各マイクロホンを車両のサンバイザーに取り付けたことを特徴とする請求項１記載のマイクロホンシステム。

【請求項３】　前記各マイクロホンを車両の運転席あるいは助手席側天井に取り付けたことを特徴とする請求項１記載のマイクロホンシステム。

【発明の詳細な説明】

【０００１】

【発明の属する技術分野】本発明はマイクロホンシステムに係わり、特に、受音感度について指向特性を有する２つマイクロホンから出力する信号を用いて適応信号処理を行い、SN比を改善した話者音声信号を出力するマイクロホンシステムに関する。

【０００２】

【従来の技術】現在の音声認識システムは、15dB以上のSN比（S：音声／N：ノイズ）が確保されている場合、約95%の認識率を実現できるくらいの技術レベルにまで達している。しかし、周囲に存在するノイズによりSN比が低下すると、それに伴って認識率が急激に低下する性質も有している。図９はSN比と認識性能との関係をいくつかの種類のマイクロホン（無指向性、単一指向性、狭指向性、AMNOR(AdaptiveMicrophone-array for Noise Reduction)）について評価したもので、SN比と認識率がおおむねＳ字特性100を示す帯の中に包含されている。この図９から明らかなように、認識率はSN比の低下により急激に低下し、SN比が0dBの環境下において約50%にまで低下してしまう。

【０００３】そのため、自動車が発生するノイズ（エンジン音・ロードノイズ・パターンノイズ・風切り音など）が存在する自動車車室内において、上記のような認識性能の劣化は避けられず、音声認識システムを車載化する上で大きな問題の一つとなっている。前記したような事情から、周囲に存在するノイズの影響を少なくし、高いSN比で音声を受音するための方式が種々提案されており、複数のマイクロホンとディジタル信号処理を用いた高SN比受音システムはその一例である。かかる高SN比受音システムの中で最も簡単な構成のものは２つのマイクロホンを使用した図１０のようなシステムであるが、他にも、Griffith-Jim型アレイやAMNORといった、より高度なシステムが提案されている。

【０００４】図１０において、１，２は第１、第２のマイクロホン、３は適応信号処理部であり、誤差信号ｅが入力されると共にマイクロホン２の出力信号x_2が参照信号として入力され、誤差信号ｅのパワーが最小となるようにLMS(Least MeanSquare)アルゴリズムに基づいて適応信号処理を行う。適応信号処理部３において、３ａはLMS演算部、３ｂは例えばFIR型デジタルフィルタ構成の適応フィルタである。LMS演算部３ａは適応信号処理により誤差信号ｅのパワーが最小となるように適応フィルタ３ｂの係数を決定する。

【０００５】４はマイクロホン１から出力する信号を目標信号として入力される目標応答設定部であり、因果性を満たすためのものである。適応フィルタ３ｂのタップ長の半分の信号遅延時間をｄとするとき、目標応答設定部４は該時間ｄの遅延特性を有し、オーディオ周波数帯域でフラットな特性（ゲイン１の特性）を有する。すなわち、目標応答設定部４は、図１１（ａ）に示すようにゲイン１のフラットな周波数特性を備え、図１１（ｂ）に示すように遅延時間ｄを有するインパルス応答特性を有している。５は減算部であり、目標応答設定部４から出力する目標応答より適応フィルタ３ｂの出力信号を減算して誤差信号ｅを出力する。

【０００６】非音声認識時、マイクロホン１、２にはノイズのみが入力し、適応信号処理部３は適応信号処理により誤差信号ｅのパワー、すなわち、ノイズ出力が最小となるようにフィルタ係数Wを決定する。一方、音声認識時には、適応信号処理部３はフィルタ係数の更新をせず、前記非音声認識時に決定したフィルタ係数Wを適応フィルタ３ｂに設定して音声信号を出力する。図１０に示すシステムに本来求められている理想的な性能は、音声認識時に出力信号として音声信号Xs(z)のみ(ノイズ出力は0)を出力することである。すなわち、ノイズ出力En(z)に関して、

$$En(z)=Xn_1(z)z^{-d}-Xn_2(z)W(z) \qquad (1)$$

とするとき、誤差信号ｅのパワーが最小となるよう調整可能なパラメータ(適応フィルタ３ｂの係数)Wを決定することである。ここで

$$Es(z)=Xs_1(z)z^{-d}-Xs_2(z)\cdot W(z)\fallingdotseq Xs(z) \qquad (2)$$

となることが理想である。ただし、$Xn_1(z)$，$Xn_2(z)$はマイクロホン１、２の出力信号に含まれるノイズであり、ノイズ源(ノイズ＝xn)から第１、第２のマイクロホン１，２までの伝搬特性をCN1，CN2とすれば、

$$Xn_1(z)=CN1\cdot xn$$

$$Xn_2(z)=CN2\cdot xn$$

であり、(1)式は

$$En(z)=(CN1\cdot z^{-d}-CN2\cdot W(z))xn \qquad (1)'$$

となる。又、$Xs_1(z)$，$Xs_2(z)$はマイクロホン１、２の出力信号に含まれる音声信号であり、話者口元(話者音声＝xs)から第１、第２マイクロホン１，２までの伝搬特性をCS1，CS2とすれば、

$Xs_1(z) = CS1 \cdot xs$

$Xs_2(z) = CS2 \cdot xs$

であり、（２）式は

$Es(z) = (CS1 \cdot z^{-d} - CS2 \cdot W(z))\, xs$ (2)'

となる。

【０００７】

【発明が解決しようとする課題】自動車には騒音源が多数存在するため、マイクロホン１，２が拾う自動車車室内ノイズのコヒーレンスは、マイクロホン１，２を遠ざけるにしたがって低下する傾向を有している。このため、２つのマイクロホン１，２を遠ざける程、(1)式が大きくなってしまう問題が生じ、マイクロホン１，２はできるだけ近い位置に配置する必要がある。ところが、２つのマイクロホン１，２をできるだけ近い位置に配置すると、２つのマイクロホンにほぼ同様の音声とノイズがそれぞれ入射する可能性が高くなり、ノイズの除去に最適に決定された適応フィルタ係数Wでノイズを消去すると、音声までもが消去されてしまう。一方、逆に(2)式を満たすように適応フィルタ係数Wを決定すると、音声は損傷を受けないかわりにノイズもほとんど消えず、SN比もほとんど改善されなくなってしまうという問題が発生する。

【０００８】以上より、ノイズを最大限抑圧するためには２つのマイクロホンが接近していることが望ましく、一方、音声の損傷を最小限にするためには、２つのマイクロホンが互いに離れている方が望ましく、両者を同時に満足することができない。このため、従来のマイクロホンシステムでは音声信号のSN比を十分に改善できない問題がある。以上から本発明の目的は、マイクロホンを２つ使用するマイクロホンシステム（ノイズリダクションシステム）において、音声信号のSN比を改善できるようにすることである。

【０００９】

【課題を解決するための手段】上記課題は本発明によれば、２つマイクロホンの出力信号を用いて適応信号処理を行いSN比を改善した話者音声信号を出力するマイクロホンシステムにおいて、指向特性を有する２つのマイクロホンを接近して配置すると共に、マイクロホンの指向方向と話者の発声方向がなす角度をマイクロホン毎に異ならせることにより達成される。このようにすれば、２つのマイクロホンを接近して配置するにもかかわらず、1つのマイクロホンは高いSN比で話者音声を拾い、もう一方のマイクロホンは低いSN比で話者音声を拾うようにできる。一方、ノイズはマイクの向きを異ならせることによるコヒーレンスの低下が大きくはないため、各マイクロホンによる受信ノイズの相関を大きく、かつ、各マイクロホンの音声に対する受信感度の差を大きくでき、音声信号のSN比を改善することができる。マイクロホンの配置例としては、車両のサンバイザーあるいは車両の助手席あるいは運転席側天井に、２つのマイクロホンを接近して、かつ、各マイクロホンの指向方向と話者発声方向とのなす角度を異ならせて取り付ける。

【００１０】

【発明の実施の形態】（ａ）本発明の原理

２つのマイクロホンを使用するノイズリダクションシステムにおいて、各マイクロホンによる受信ノイズの相関を大きくし、かつ、各マイクロホンの音声に対する受信感度の差を大きくすることが理想である。しかし、２つのマイクロホンのこれら“受信ノイズの相関”と“音声に対する受信感度の差”にはトレードオフが存在し、距離を調整して一方を満たせば他方が満たされなくなる。例えば、マイクロホン間距離が近い程、２つのマイクロホンの受信ノイズの相関が大きくなるが、音声に対する受信感度の差がなくなり音声を同じように受信してしまう。このため、適応信号処理を行った場合、ノイズを抑圧できるが、音声も同時に抑圧してしまう事になり、結果としてSN比の改善は望めなくなる。

【００１１】そこで、本発明では、指向特性を有する２つのマイクロホンを接近して配置すると共に、マイクロホンの指向方向と話者の発声方向がなす角度をマイクロホン毎に異ならせる。このようにすれば、２つのマイクロホンを接近して配置するにもかかわらず、1つのマイクロホンは高いSN比で話者音声を拾い、もう一方のマイクロホンは低いSN比で話者音声を拾うようにできる。このため、２つのマイクロホンの接近配置により受信ノイズ間の相関を高くでき、かつ、各マイクロホンの音声に対する受信感度の差を大きくでき、音声信号のSN比を改善することができる。

【００１２】（ｂ）マイクロホンシステムの構成

図１は本発明のマイクロホンシステムの構成図であり、図１０のシステムと同一部分には同一符号を付している。図中、１０は話者であり、例えば自動車の運転手、１１，１２は受音感度について指向特性を有する第１、第２のマイクロホンである。マイクロホンの指向特性は、例えば図２に示す単一指向性の感度特性を有している。すなわち、指向方向を$\theta = 0^0$、該指向方向の感度をE_0するとき、任意の角度θ方向の感度は次式

$E(\theta) = E_0 (1 + \cos\theta) / 2$

により表現され、指向方向からずれるにしたがって感度が低下する。

【００１３】第１、第２のマイクロホン１１，１２は一例として運転席側サンバイザー１３にたとえば１０cmの間隔で取り付けられている。第１のマイクロホン１１の指向方向は話者発声方向(話者口が向いている方向)と一致して設けられ、第２のマイクロホン１２の指向方向は助手席方向を向いており、話者発声方向と所定の角度θを形成している。したがって、図２の指向特性より、第１のマイクロホン１１の方が話者音声に対して感度が良く、高いSN比で話者音声を拾い、第２のマイクロホン１２は話者音声に対して感度が悪く、低いSN比で話者音声

を拾う。

【００１４】３は適応信号処理部で、誤差信号ｅが入力されると共にマイクロホン１２の出力信号x_2が参照信号として入力され、誤差信号ｅのパワーが最小となるようにLMSアルゴリズムに基づいて適応信号処理を行う。適応信号処理部３において、３ａはLMS演算部、３ｂはFIR型デジタルフィルタ構成の適応フィルタである。LMS演算部３ａは適応信号処理により誤差信号ｅのパワーが最小となるように適応フィルタ３ｂの係数を決定する。適応信号処理部３は、非音声認識時においてのみ適応信号処理により適応フィルタ３ｂのフィルタ係数Wを決定し、音声認識時にはフィルタ係数の更新をせず、前記非音声認識時に決定したフィルタ係数Wを適応フィルタ３ｂに設定する。

【００１５】４はマイクロホン１１から出力する信号を目標信号として入力される目標応答設定部で、時間ｄの遅延特性を有し、かつ、オーディオ周波数帯域でフラットな特性（ゲイン１の特性）を有している。５は減算部で、目標応答設定部４から出力する目標応答より適応フィルタ３ｂの出力信号を減算して誤差信号ｅを出力する。図１のマイクロホン配置によれば、第１、第２のマイクロホン１１、１２の話者音声に対する感度の相違に加えて、人間の音声放射特性（人間の口を音源としたときの音圧が話者の正面に対して偏向するほど低くなる特性）により、近接配置したにも係らず２つのマイクロホン１１，１２で受音する音声パワーを異ならせることができ、しかも、近接配置により、２つのマイクロホン１１，１２で受音するノイズの相関も高く保つことができる。

【００１６】（ｃ）動作

マイクロホン１１、１２にノイズのみが入力する非音声認識時において、適応信号処理部３は適応信号処理により誤差信号ｅのパワーが最小となるように適応フィルタ３ｂのフィルタ係数Wを決定する。理想的には、フィルタ係数W(Z)は、

$$W(z)=CN1\cdot z^{-d}/CN2 \quad (3)$$

となる。

【００１７】一方、音声認識時、適応信号処理部３はフィルタ係数の更新をせず、前記非音声認識時に決定したフィルタ係数W(Z)を適応フィルタ３ｂに設定して音声信号を出力する。この結果、(2)′及び(3)式より音声信号は次式

$$Es(z)=(CS1\cdot z^{-d}-CS2\cdot W(z))\,xs$$
$$=(CS1-CS2\cdot CN1/CN2)\cdot z^{-d}\cdot xs \quad (4)$$

となる。マイクロホンの接近配置によりCN1≒CN2が成立するとすれば、(4)式の音声信号Es(z)は次式

$$Es(z)=(CS1-CS2)\cdot z^{-d}\cdot xs \quad (4)'$$

で与えられる。マイクロホン１、２の感度差及び音声放射特性によりCS1≒CS2であるため、音声信号Es(z)は0となることはない。すなわち、ノイズ入力時の誤差信号ｅのパワーが最小となるように適応フィルタ係数W(Z)を決定しても、(4)式の音声出力Es(z)は零とならず、音声信号のSN比を改善することができる。又、CN1≒CN2が成り立てば音声信号Es(z)の大きさは、(CS1-CS2)の差、すなわち、マイクロホン１１，１２の感度の差に主に依存する。

【００１８】（ｄ）マイクロホン配置とSN比改善量の検討

以上により、SN比を改善するためには、なるべく相関のあるノイズを２つのマイクロホンで受信しながら、音声はなるべく一方のマイクロホンでのみ受信することが基本である。この基本原理に基づき、最適なマイクロホンレイアウトの検討を行った。マイクロホン配設場所として、(1) 車両のサンバイザーと(2) 車両の助手席側天井を選定した。

【００１９】(d-1) マイクロホンのレイアウト

図３（ａ）はサンバイザーにマイクロホンを設置する場合のレイアウト説明図であり、第１、第２のマイクロホン１１、１２を話者１０前方のサンバイザー（図示せず）に距離ｄの間隔で配置し、かつ、第１のマイクロホン１１の指向方向と発声方向を一致するように固定し、第２のマイクロホン１２の指向方向と話者発声方向の角度をθとした。話者口元から各マイクロホンまでの垂直距離H及びサンバイザーまでの距離Dは一定で共に約30cmである。SN比の改善量の検討に際しては、(1) 第１、第２のマイクロホン１１，１２の位置を固定し、第２のマイクロホン１２の指向方向を可変した（図４参照）、(2) 第１、第２のマイクロホン１１，１２の指向方向を固定し、第２のマイクロホン１２の位置を移動してマイク間距離ｄを可変した（図５参照）。

【００２０】図３（ｂ）は助手席側天井にマイクロホン１１、１２を設置する場合のレイアウト説明図であり、第１、第２のマイクロホン１１、１２を助手席側天井に距離ｄの間隔で前後に配置し、かつ、第１、第２のマイクロホン１１、１２の指向方向を話者発声方向と直角あるいは所定角度θを形成するように配置した。話者口元から各マイクロホンまでの垂直距離H及び水平距離Dは一定で共に約30cmである。SN比の改善量の検討に際しては、(3) 第１、第２のマイクロホン１１，１２をその指向方向が発声方向と直角となるようにし、第２のマイクロホン１２の位置を可変した（図６参照）、(4) 第１のマイクロホン１１の指向方向を発声方向に直角な方向と角度θを形成するように固定し（話者の口元方向に向け設置し）、一方、第２のマイクロホン１２の指向方向を発声方向と直角となるようにし、該第２のマイクロホン位置を可変した（図７参照）。

【００２１】(d-2) 検討結果

図４～図７は上記(1)～(4)において、SN比の改善量が最大になった場合の説明図である。各図において、「Ps」は音声パワー、「Pn」はノイズパワー、「SNR」はSN比、

「改善量」はSN比改善量(dB)、「NR量」はノイズリダクション量(dB)を意味している。又、「NR前」はノイズリダクション制御をしない場合の値(図1のA点におけるPs，Pn)、「NR後」はノイズリダクション制御を行った場合の値(図1のB点におけるPs，Pn)である。又、検討に際しては、「八戸」、「気仙沼」、「行橋」、「札幌」、「北見」の5つの地名の発声を行った場合について、NR前、NR後のPs，Pn、SNRを求め、NR前後のSNRよりSN比改善量を求め、それぞれのSN比改善量の平均値を計算した。

【0022】(1) 図4は、サンバイザーにおいて第1、第2のマイクロホン11，12の位置を固定し、右側の第2のマイクロホン12の指向方向と話者発声方向のなす角度θを可変した場合の検討結果である。θ=15°，30°，45°，60°，90°，120°，180°について検討した結果、θ=45°において最大の平均SN比改善量4.3dBが得られた。

【0023】(2) 図5は、サンバイザーにおいて第1のマイクロホン11の指向方向を話者発声方向に固定し、第2のマイクロホン12の指向方向を話者発声方向と60°の角度をなすように固定し、第2のマイクロホン12の位置を移動してマイク間距離dを可変した場合の検討結果である。d=3cm，6cm，9cm，12cm，15cm，18cmについて検討した結果、d=9cmにおいて最大の平均 SN比改善量4.7dBが得られた。

【0024】(3) 図6は、助手席側天井において第1、第2のマイクロホン11，12の指向方向を話者発声方向と直角となるようにし、第2のマイクロホン12の位置を移動してマイク間距離dを可変した場合の検討結果である。d=2.5cm，5cm，7.5cmについて検討した結果、d=7.5cmにおいて最大の平均 SN比改善量4.5dBが得られた。

【0025】(4) 図7は、助手席側天井において第1のマイクロホン11の指向方向を話者発声方向に直角な方向と角度θを形成するように固定し、又、第2のマイクロホン12の指向方向を話者発声方向と直角となるようにし、該第2のマイクロホン位置を移動してマイク間距離dを可変した場合の検討結果である。d=2cm，4cm,6cmについて検討した結果、d=2cmにおいて最大の平均 SN比改善量4.5dBが得られた。以上のようにマイクロホンを(1)～(4)のようにレイアウトすることにより4～5dBのSN比の改善ができる。このSN比改善により認識率を大幅に改善することが可能になる。以上図6、図7では助手席側天井にマイクロホン11、12を設けた例を示したが、運転席側天井の同様位置にマイクロホンを設けることもできる。

【0026】(e) 別のマイクロホンシステムの構成図
図8は本発明を適用できるマイクロホンシステムの別の構成図であり、図1と同一部分には同一符号を付している。異なる点は、図1の目標応答設定部4を適応信号処理部4′で構成している点である。図1のマイクロホンシステムでは、誤差信号eのパワーが最小となるように適応信号処理部3でのみ適応信号処理を行うが、図8のマイクロホンシステムでは、適応信号処理部3と適応信号処理部4′が誤差信号eのパワーが最小となるように適応信号処理を行う。以上、本発明を実施例により説明したが、本発明は請求の範囲に記載した本発明の主旨に従い種々の変形が可能であり、本発明はこれらを排除するものではない。

【0027】

【発明の効果】以上本発明によれば、指向特性を有する2つのマイクロホンを接近して配置すると共に、マイクロホンの指向方向と話者の発声方向がなす角度をマイクロホン毎に異ならせるようにしたから、一方のマイクロホンから出力する音声信号のSN比を高くし、他方のマイクロホンから出力する音声信号のSN比を低くするようにでき、この結果、ノイズ出力が最小となるように適応フィルタ係数を決定しても、音声出力は零とならず、音声信号のSN比を改善することができる。又、本発明によれば、各マイクロホンを車両のサンバイザーあるいは車両の助手席あるいは運転席側天井に取り付け、それぞれの指向方向を異ならせるだけの簡単な構成で、各マイクロホンを比較的近距離に配置しているにも拘らず、1つのマイクロホンにおいてできるだけ高いSN比で音声を拾い、もう一方のマイクロホンにおいて、できるだけ低いSN比で音声を拾うようにでき、SN比を改善することができる。

【図面の簡単な説明】

【図1】本発明のマイクロホンシステムである。

【図2】指向特性説明図である。

【図3】マイクロホンのレイアウト説明図である。

【図4】サンバイザーに取り付けた右側のマイクロホンの指向方向と話者発声方向のなす角度θを可変した場合のSN比改善量説明図表である。

【図5】サンバイザーにマイクロホンを取り付け、60°の角度を付けた右側のマイクロホンを移動した場合のSN比改善量説明図表である。

【図6】助手席側天井に指向方向が話者発声方向と直角となるように各マイクロホンを取り付け、一方のマイクロホンの位置を移動してマイク間距離を可変した場合のSN比改善量説明図表である。

【図7】助手席側天井前方にマイクロホンを取り付け、マイクロホン間距離を可変した場合のSN比改善量説明図表である。

【図8】別のマイクロホンシステムの構成図である。

【図9】SN比と認識率の関係図である。

【図10】従来のマイクロホンを2つ使用した場合の高SN比受音システムである。

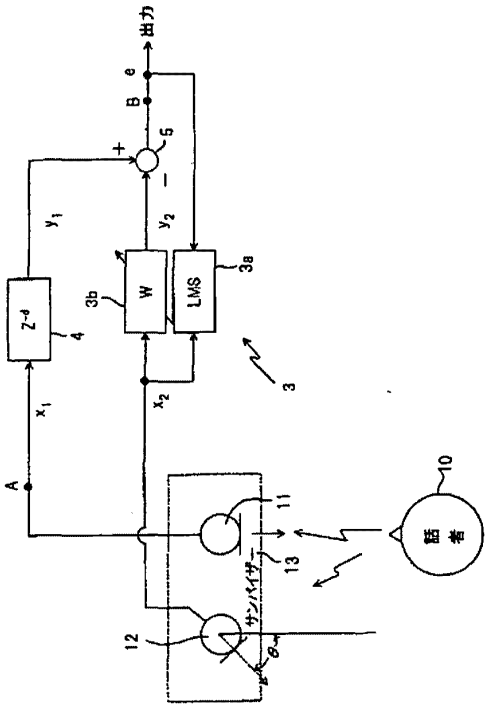
【図11】目標応答設定部の特性図である。

【符号の説明】

11，12・・第1、第2のマイクロホン
13・・サンバイザー
3・・適応信号処理部
3a・・LMS演算部
3b・・適応フィルタ
4・・目標応答設定部
5・・減算部

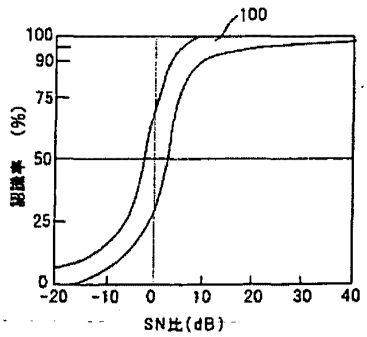
【図1】

本発明のマイクロホンシステム



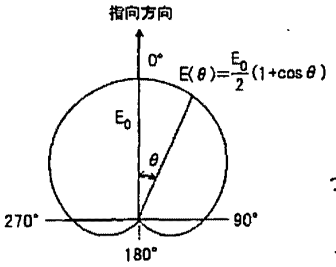
【図9】

SN比と認識率の関係



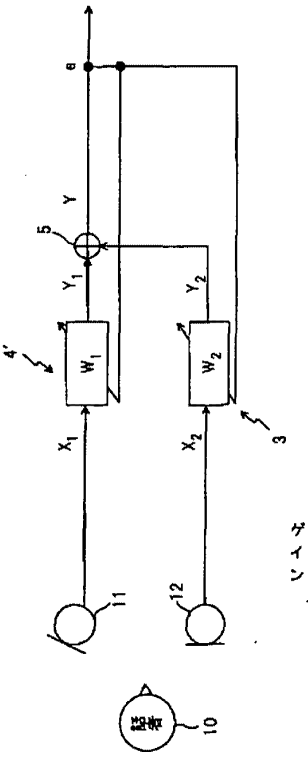
【図2】

指向特性説明図



【図8】

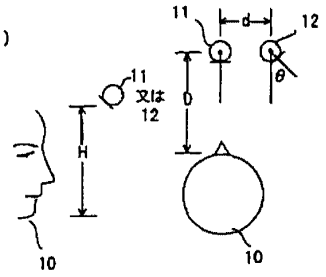
別のマイクロホンシステムの構成



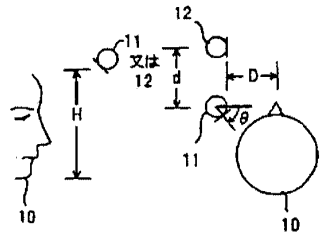
【図3】

マイクロホンレイアウト説明図

(a)
マイクロホンをサンバイザーに設置する場合



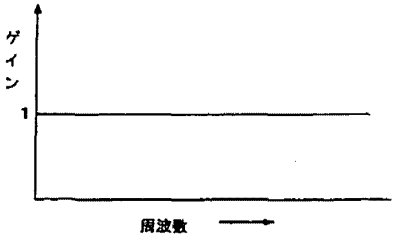
(b)
マイクロホンを助手席側天井に設置する場合



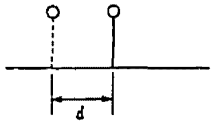
【図11】

目標応答設定部の特性

(a)
周波数特性

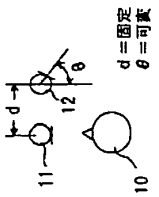


(b)
インパルス応答特性



【図4】

サンバイザー取付け右側のマイクロホンの指向方向を
変えた場合のSN改善量説明図表



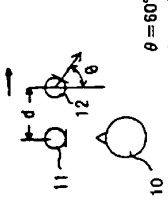
(b)

NR前 NR後

条件		八戸		気仙沼		行橋		札幌		北見		改善量平均
サンバイザー R 45°	Ps	-11.3	-12.0	-14.2	-14.7	-16.2	-16.6	-10.8	-10.3	11.5	-11.0	NR量:6.83
	Pn	-17.0	-21.7	-17.0	-21.7	-17.0	-21.7	-17.0	-21.7	-17.0	-21.7	○
	SNR	5.6	9.7	2.7	7.0	0.7	5.0	6.3	11.4	5.4	9.8	
	改善量	4.1		4.2		4.1		5.1		4.3		4.3

【図5】

サンバイザーにマイクロホンを取付け、60° の角度をつけた
右側のマイクロホンを移動した場合のSN改善量説明図表



(b)

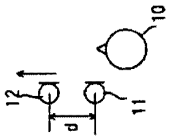
NR前 NR後

条件		八戸		気仙沼		行橋		札幌		北見		改善量平均
サンバイザー R 9cm-60°	Ps	-14.8	-13.0	-13.5	-11.6	-15.8	-13.7	-12.3	-9.5	-10.3	-6.7	NR量:4.51
	Pn	-17.6	-20.0	-17.6	-20.0	-17.8	-20.0	-17.6	-20.0	-17.6	-20.0	○
	SNR	3.0	5.4	4.1	8.4	2.0	6.3	5.3	10.5	7.3	13.3	
	改善量	3.5		4.3		4.4		5.2		6.0		4.7

【図6】

助手席側天井に各マイクロホンを話者発声方向と直角に取付け、一方のマイクロホン位置を移動した場合のSN改善量説明図

(a)



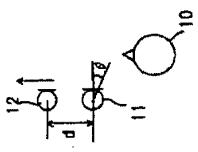
(b)

条件		八戸		気仙沼		行橋		札幌		北見		改善量平均
横天	Ps	-12.2	-16.1	-12.6	-17.1	-13.5	-17.1	-12.3	-15.0	-8.8	-11.1	NR量:3.90
距離	Pn	-12.8	-20.8	-12.8	-20.8	-12.8	-20.8	-12.8	-20.8	-12.8	-20.8	○
7.5cm	SNR	0.7	4.6	0.2	3.7	-0.6	3.8	0.5	5.8	4.0	9.6	
	改善量		4.0		3.6		4.3		5.2		5.6	4.5

【図7】

助手席側天井前方にマイクを取付け、マイク間距離を可変した場合のSN改善量説明図表

(a)

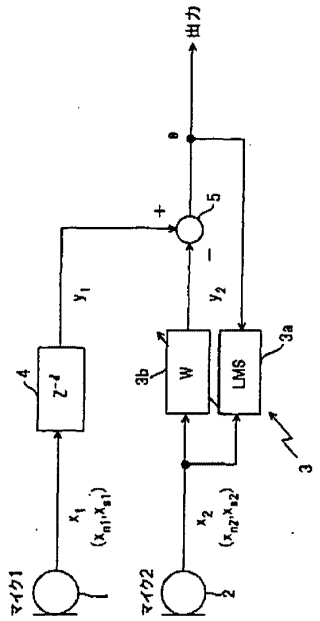


(b)

条件		八戸		気仙沼		行橋		札幌		北見		改善量平均
		NR前	NR後	NR前	NR後	NR前	NR後	NR前	NR後	NR前	NR後	
横天	Ps	-14.3	-14.0	-13.0	-13.7	-15.9	-15.6	-11.2	-11.7	-10.4	-11.0	NR量:5.77
F	Pn	-20.2	-24.7	-20.2	-24.7	-20.2	-24.7	-20.2	-24.7	-20.2	-24.7	○
平行−2cm	SNR	6.0	10.7	8.5	11.0	4.7	9.2	8.5	13.0	9.3	13.8	
	改善量		4.7		4.5		4.5		4.5		4.5	4.5

【図10】

従来のマイクロホンを2つ使用した場合の高SN比受音システム



フロントページの続き

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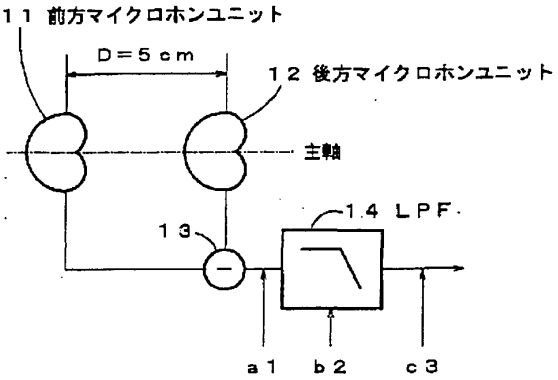
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(54)【発明の名称】　狭指向性マイクロホンユニット

(57)【要約】
【課題】　全音声帯域で狭指向性を得、人の声の明瞭度を高めるとともに、走行ノイズや振動ノイズを拾いにくくする。
【解決手段】　２個の単一指向性マイクロホン１１、１２を同一軸上に所定間隔で配列し、この２個のマイクロホン出力を演算器１３で減算出力し、ローパスフィルタ１４で高周波成分を除去する。例えば、マイクロホンの設置間隔を５０ｍｍ（電話音声帯域の上限の半波長相当）に設定することにより、正面特性が電話の音声帯域の上限である３ｋＨｚからマイクロホンに生じる位相差で６ｄＢ／ｏｃｔの特性が実現され、カットオフ周波数５００Ｈｚ、６ｄＢ／ｏｃｔのローパスフィルタ１４を組み込むことにより、マイクロホン出力が５００Ｈｚから３．４ｋＨｚまでのバンドパスフィルタを通した特性になる。



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【特許請求の範囲】
【請求項１】　所定の間隔を隔て同一軸上に配置された２個の単一指向性マイクロホンユニットと、
前記単一指向性マイクロホンユニットの一方の出力信号と他方の出力信号の差を演算する演算器と、
前記演算器出力の高周波成分を除去するローパスフィルタと、を備えたことを特徴とする狭指向性マイクロホン。
【発明の詳細な説明】
【０００１】
【発明の属する技術分野】本発明は、特に、車載用ハンズフリー通話システム及び音声認識システムに用いて好適な、狭指向性マイクロホンに関する。
【０００２】
【従来の技術】図３、図４に従来の狭指向性マイクロホンの基本構成が示されている。図３に示されるように従来の狭指向性マイクロホンは、主軸方向に正面方向を向けた２つの単一指向性マイクロホンユニット３１、３２と、遅延回路網３３、そして減算器３４で構成されるか、あるいは図４に示されるように、主軸方向に正面方向を向けた２つの単一指向性マイクロホンユニット４１、４２と、ハイパスフィルタ４３、そして減算器４４で構成される。
【０００３】狭指向性マイクロホンは、二次音圧傾度型（１つの無指向性マイクロホンユニットの指向性をｆ（θ）＝１としたとき、単一指向性マイクロホンユニット２個を使用した場合の指向性ｆ（θ）＝（１＋ｃｏｓθ）^{2}で示される）が一般的であるが、正面特性の均一性をはかるため、図３、図４に示すように２つのマイクロホンユニット３１（４１）、３２（４２）のうち、、一方のマイクロホンユニットの出力を遅延回路網３３で遅延させるか、あるいはハイパスフィルタ４３を介して高周波成分を除去した後、減算器３４（４３）で減算していた。
【０００４】
【発明が解決しようとする課題】上述した従来技術によれば、全音声帯域、特に低域での狭指向性が得られないため、走行ノイズや振動ノイズを拾いやすく、かつ、音声の明瞭度が低い。従って、ハンズフリー通話システム等車内での実使用に耐えない。
【０００５】本発明は上記事情に鑑みてなされたものであり、２個の単一指向性マイクロホンユニットを同一軸上に電話音声帯域の半波長相当間隔を置いて配置し、前記２個のマイクロホンユニット出力を減算して高周成分を除去することで、低域から高域に至る全音声帯域で狭指向性を得ることのできる、特に車内での使用に適した狭指向性マイクロホンを提供することを目的とする。
【０００６】
【課題を解決するための手段】上記した課題を解決するために請求項１に記載の狭指向性マイクロホンは、所定の間隔を隔て同一軸上に配置された２個の単一指向性マイクロホンユニットと、前記単一指向性マイクロホンユニットの一方の出力信号と他方の出力信号の差を演算する演算器と、前記演算器出力の高周波成分を除去するローパスフィルタとを備えたことを特徴とする。
【０００７】このことにより、マイクロホンユニットの設置間隔を例えば５０ｍｍ（電話音声帯域の上限の半波長相当）に設定することにより、正面特性が電話の音声帯域の上限である３．４ｋＨｚからマイクロホンユニットに生じる位相差で６ｄＢ／ｏｃｔの特性が実現され、例えば、カットオフ周波数５００Ｈｚ、６ｄＢ／ｏｃｔのローパスフィルタを組み込むことにより、マイクロホン出力が５００Ｈｚから３．４ｋＨｚまでのバンドパスフィルタを通した特性となり、全音声帯域に渡り狭指向性を実現でき、人の声の明瞭度が高まると共に、低周波成分の感度が低くなるためにロードノイズを拾いにくくなり、従って、車内でのハンズフリー通話での実用に耐えることができる。
【０００８】
【発明の実施の形態】図１は本発明の狭指向性マイクロホンの実施形態を示す図である。図２は本発明実施形態の動作を説明するために引用した図であり、図１における各ポイントａ、ｂ、ｃにおける信号の周波数特性を示した図である。
【０００９】図１において、１１、１２は、単一指向性マイクロホンユニットであり、５ｃｍの間隔を隔て同一軸上に設置されるものである。ここでは、設置位置の関係上、正面に近いところから、前方マイクロホンユニット１１、後方マイクロホンユニット１２と称する。なお、５ｃｍの間隔は、電話音声帯域の上限３．４ｋＨｚの半波長に相当する。１３は演算器である。演算器１３は、前方マイクロホンユニット１１の出力から後方マイクロホンユニット１２の出力を減算することにより、図２（ａ）に示す周波数特性を持つ出力信号を生成するものである。すなわち、２つのマイクロホンユニット１１、１２出力を減算することにより、正面特性は、位相差によって位相が反転する周波数（マイクロホンユニット１１、１２の設置間隔Ｄが５ｃｍのとき３．４ＭＨｚ）まで、６ｄＢ／ｏｃｔで上昇する。なお、側面から拾われる音は、全ての周波数において同相入力となるため、減算されて出力は０になる。
【００１０】１４はローパスフィルタである。ローパスフィルタ１４は、図２（ｂ）に示す周波数特性を持つものを使用する。すなわち、カットオフ周波数５００Ｈｚ、６ｄＢ／ｏｃｔのローパスフィルタを用い、演算器１３出力の高周波成分を除去する。このことにより、図２（ｃ）に示す特性を持つ出力が生成される。従って、マイクロホンの出力が５００Ｈｚから３．４ｋＨｚまでバンドパスフィルタを通した特性となり、人の声の明瞭度を高めると共に、走行ノイズ等低周波成分を押さえる

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ことができる。

【0011】このように、図2（a）に示す特性を持ったマイクロホン出力（減算器13出力）に、図2（b）に示す特性を持つローパスフィルタ14を通すことにより、正面特性が所望のバンドパスフィルタを通した、すなわち、図2（c）に示す特性を得ることができる。

【0012】図2（a）の特性を得るマイクロホンは、2つの単一指向性マイクロホンユニットを同一軸上に配列し、前方マイクロホンユニット11から後方マイクロホンユニット12の出力を減算器13で減算することによって得ることができる。これらマイクロホンユニット11、12において、設置間隔を50mmに設定することにより、マイクロホンの正面特性が電話の音声帯域の上限である3．4MHzからマイクロホンユニット11（12）に生じる位相差で6dB／octの特性を実現し、このことにより、低周波の感度が低くなり、従ってロードノイズを拾いにくくなる。また、マイクロホンユニット11、12の側面から拾われる音は、2つのマイクロホンユニット11、12に同相で入力されるため、減算することによって相殺され、結果として正面のみの音が出力され、全音声帯域での狭指向性が実現される。

【0013】

【発明の効果】以上説明のように本発明は、2個の単一指向性マイクロホンユニットを同一軸上に所定間隔で配列し、この2個のマイクロホンユニット出力を減算器で減算出力して高周波成分を除去したものである。このことにより、マイクロホンの設置間隔を例えば50mm（電話音声帯域の上限の半波長相当）に設定することにより、正面特性が電話の音声帯域の上限である3．4kHzからマイクロホンユニットに生じる位相差で6dB／oct.の特性が実現され、例えば、カットオフ周波数500Hz、6dB／octのローパスフィルタを組み込むことにより、マイクロホン出力が500Hzから3．4kHzまでのバンドパスフィルタを通した特性となる。従って、全音声帯域に渡り狭指向性を実現でき、人の声の明瞭度が高まると共に、低周波成分の感度が低くなるためにロードノイズを拾いにくくなり、従って、車内でのハンズフリー通話での実用に耐えることができる。

【図面の簡単な説明】

【図1】本発明の狭指向性マイクロホンの一実施形態を示す図である。

【図2】本発明実施形態の動作を説明するために引用した図であり、図1の各ポイントにおける信号の周波数特性を示す図である。

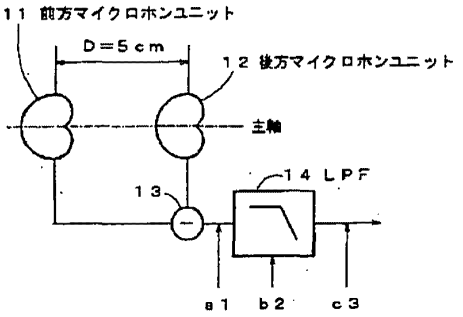
【図3】従来における狭指向性マイロホンの一構成例を示す図である。

【図4】従来における狭指向性マイロホンの他の構成例を示す図である。

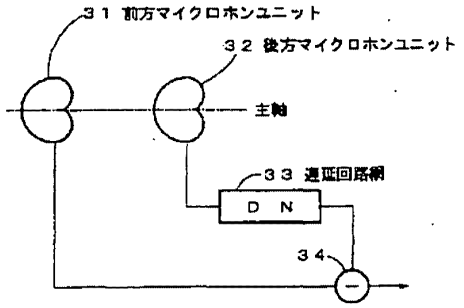
【符号の説明】

11　前方マイクロホンユニット
12　後方マイクロホンユニット
13　減算器
14　ローパスフィルタ

【図1】

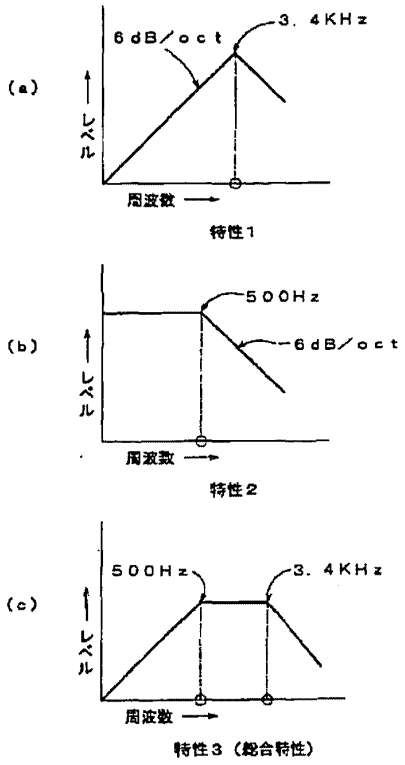


【図3】

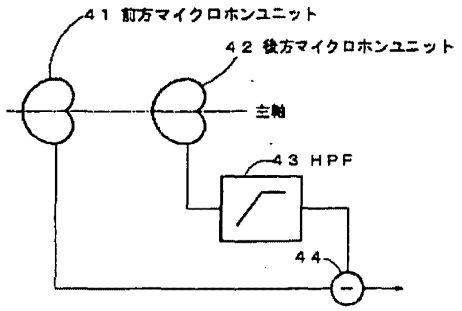


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【図2】



【図4】



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(54) Title: METHOD AND APPARATUS FOR REMOVING NOISE FROM ELECTRONIC SIGNALS

(57) Abstract: A method and system are provided for acoustic noise removal from human speech, wherein noise is removed without respect to noise type, amplitude, or orientation. The system includes microphones and a voice activity detection (VAD) data stream coupled among a processor. The microphones receive acoustic signals and the VAD produces a signal including a binary one when speech (voiced and unvoiced) is occurring and a binary zero in the absence of speech. The processor includes denoising algorithms that generate transfer functions. The transfer functions include a transfer function generated in response to a determination that voicing information is absent from the received acoustic signal during a specified time period. The transfer functions also include transfer functions generated in response to a determination that voicing information is present in the acoustic signal during a specified time period. At least one denoised acoustic data stream is generated using the transfer functions.

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METHOD AND APPARATUS FOR REMOVING NOISE FROM ELECTRONIC SIGNALS

FIELD OF THE INVENTION

The invention is in the field of mathematical methods and electronic systems for removing or suppressing undesired acoustical noise from acoustic transmissions or recordings.

BACKGROUND

In a typical acoustic application, speech from a human user is recorded or stored and transmitted to a receiver in a different location. In the environment of the user, there may exist one or more noise sources that pollute the signal of interest (the user's speech) with unwanted acoustic noise. This makes it difficult or impossible for the receiver, whether human or machine, to understand the user's speech. This is especially problematic now with the proliferation of portable communication devices like cellular telephones and personal digital assistants. There are existing methods for suppressing these noise additions, but they either require far too much computing time or cumbersome hardware, distort the signal of interest too much, or lack in performance to be useful. Many of these methods are described in textbooks such as "Advanced Digital Signal Processing and Noise Reduction" by Vaseghi, ISBN 0-471-62692-9. Consequently, there is a need for noise removal and reduction methods that address the shortcomings of typical systems and offer new techniques for cleaning acoustic signals of interest without distortion.

SUMMARY

A method and system are provided for acoustic noise removal from human speech, wherein the noise can be removed and the signal restored without respect to noise type, amplitude, or orientation. The system includes microphones and sensors coupled with a processor. The microphones receive acoustic signals including both noise and speech signals from human signal sources. The sensors yield a binary Voice Activity Detection (VAD) signal that provides a signal that is a binary "1" when speech (both voiced and unvoiced) is occurring and a binary "0" when no speech is occurring. The VAD signal can be obtained in numerous ways, for example, using acoustic gain, accelerometers, and radio frequency (RF) sensors.

The processor system and method includes denoising algorithms that calculate the transfer function among the noise sources and the microphones as well as the transfer function among the human user and the microphones. The transfer functions are used to remove noise from the received acoustic signal to produce at least one denoised acoustic data stream.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of a denoising system of an embodiment.

Figure 2 is a block diagram of a noise removal algorithm of an embodiment, assuming a single noise source and a direct path to the microphones.

Figure 3 is a block diagram of a front end of a noise removal algorithm of an embodiment, generalized to *n* distinct noise sources (these noise sources may be reflections or echoes of one another).

Figure 4 is a block diagram of a front end of a noise removal algorithm of an embodiment in the most general case where there are *n* distinct noise sources and signal reflections.

Figure 5 is a flow diagram of a denoising method of an embodiment.

Figure 6 shows results of a noise suppression algorithm of an embodiment for an American English female speaker in the presence of airport terminal noise that includes many other human speakers and public announcements.

DETAILED DESCRIPTION

Figure 1 is a block diagram of a denoising system of an embodiment that uses knowledge of when speech is occurring derived from physiological information on voicing activity. The system includes microphones 10 and sensors 20 that provide signals to at least one processor 30. The processor includes a denoising subsystem or algorithm.

Figure 2 is a block diagram of a noise removal system/algorithm of an embodiment, assuming a single noise source and a direct path to the microphones. The noise removal system diagram includes a graphic description of the process of an embodiment, with a single signal source (100) and a single noise source (101). This algorithm uses two microphones, a "signal" microphone (MIC 1, 102) and a "noise" microphone (MIC 2, 103), but is not so limited. MIC 1 is assumed to capture mostly signal with some noise, while MIC 2 captures mostly noise with some signal. This is the common configuration with conventional advanced acoustic systems. The data from the signal to MIC 1 is denoted by s(n), from the signal to MIC 2 by $s_2(n)$, from the noise to MIC 2 by n(n), and from the noise to MIC 1 by $n_2(n)$. Similarly, the data from MIC 1 is denoted by $m_1(n)$, and the data from MIC 2 $m_2(n)$, where s(n) denotes a discrete sample of the analog signal from the source.

The transfer functions from the signal to MIC 1 and from the noise to MIC 2 are assumed to be unity, but the transfer function from the signal to MIC 2 is denoted by $H_2(z)$ and from the noise to MIC 1 by $H_1(z)$. The assumption of unity transfer functions does not inhibit the generality of this algorithm, as the actual relations between the signal, noise, and microphones are simply ratios and the ratios are redefined in this manner for simplicity.

In conventional noise removal systems, the information from MIC 2 is used to attempt to remove noise from MIC 1. However, an unspoken assumption is that the Voice Activity Detection (VAD) is never perfect, and thus the denoising must be performed cautiously, so as not to remove too much of the signal along with the noise. However, if the VAD is assumed to be perfect and is equal to zero when there is no speech being produced by the user,

and one when speech is produced, a substantial improvement in the noise removal can be made.

In analyzing the single noise source and direct path to the microphones, with reference to **Figure 2**, the acoustic information coming into MIC 1 is denoted by $m_1(n)$. The information coming into MIC 2 is similarly labeled $m_2(n)$. In the z (digital frequency) domain, these are represented as $M_1(z)$ and $M_2(z)$. Then

$$M_1(z) = S(z) + N_2(z)$$
$$M_2(z) = N(z) + S_2(z)$$

with

$$N_2(z) = N(z)H_1(z)$$
$$S_2(z) = S(z)H_2(z)$$

so that

$$M_1(z) = S(z) + N(z)H_1(z)$$
$$M_2(z) = N(z) + S(z)H_2(z) \qquad \text{Eq. 1}$$

This is the general case for all two microphone systems. In a practical system there is always going to be some leakage of noise into MIC 1, and some leakage of signal into MIC 2. Equation 1 has four unknowns and only two known relationships and therefore cannot be solved explicitly.

However, there is another way to solve for some of the unknowns in Equation 1. The analysis starts with an examination of the case where the signal is not being generated, that is, where the VAD signal equals zero and speech is not being produced. In this case, $s(n) = S(z) = 0$, and Equation 1 reduces to

$$M_{1n}(z) = N(z)H_1(z)$$
$$M_{2n}(z) = N(z)$$

where the n subscript on the M variables indicate that only noise is being received. This leads to

$$M_{1n}(z) = M_{2n}(z)H_1(z)$$
$$H_1(z) = \frac{M_{1n}(z)}{M_{2n}(z)} \qquad \text{Eq. 2}$$

$H_1(z)$ can be calculated using any of the available system identification algorithms and the microphone outputs when the system is certain that only noise is being received. The calculation can be done adaptively, so that the system can react to changes in the noise.

A solution is now available for one of the unknowns in Equation 1. Another unknown, $H_2(z)$, can be determined by using the instances where the VAD equals one and speech is being produced. When this is occurring, but the recent (perhaps less than 1 second) history of the microphones indicate low levels of noise, it can be assumed that $n(s) = N(z) \sim 0$. Then Equation 1 reduces to

$$M_{1s}(z) = S(z)$$
$$M_{2s}(z) = S(z)H_2(z)$$

which in turn leads to

$$M_{2s}(z) = M_{1s}(z)H_2(z)$$
$$H_2(z) = \frac{M_{2s}(z)}{M_{1s}(z)}$$

which is the inverse of the $H_1(z)$ calculation. However, it is noted that different inputs are being used – now only the signal is occurring whereas before only the noise was occurring. While calculating $H_2(z)$, the values calculated for $H_1(z)$ are held constant and vice versa. Thus, it is assumed that $H_1(z)$ and $H_2(z)$ do not change substantially while the other is being calculated.

After calculating $H_1(z)$ and $H_2(z)$, they are used to remove the noise from the signal. If Equation 1 is rewritten as

$$S(z) = M_1(z) - N(z)H_1(z)$$
$$N(z) = M_2(z) - S(z)H_2(z)$$
$$S(z) = M_1(z) - [M_2(z) - S(z)H_2(z)]H_1(z)$$
$$S(z)[1 - H_2(z)H_1(z)] = M_1(z) - M_2(z)H_1(z)$$

then N(z) may be substituted as shown to solve for S(z) as:

$$S(z) = \frac{M_1(z) - M_2(z)H_1(z)}{1 - H_2(z)H_1(z)} \qquad \text{Eq. 3}$$

If the transfer functions $H_1(z)$ and $H_2(z)$ can be described with sufficient accuracy, then the noise can be completely removed and the original signal recovered. This remains true without respect to the amplitude or spectral

characteristics of the noise. The only assumptions made are a perfect VAD, sufficiently accurate $H_1(z)$ and $H_2(z)$, and that $H_1(z)$ and $H_2(z)$ do not change substantially when the other is being calculated. In practice these assumptions have proven reasonable.

The noise removal algorithm described herein is easily generalized to include any number of noise sources. **Figure 3** is a block diagram of a front end of a noise removal algorithm of an embodiment, generalized to *n* distinct noise sources. These distinct noise sources may be reflections or echoes of one another, but are not so limited. There are several noise sources shown, each with a transfer function, or path, to each microphone. The previously named path H_2 has been relabeled as H_0, so that labeling noise source 2's path to MIC 1 is more convenient. The outputs of each microphone, when transformed to the z domain, are:

$$\begin{aligned} M_1(z) &= S(z) + N_1(z)H_1(z) + N_2(z)H_2(z) + \ldots N_n(z)H_n(z) \\ M_2(z) &= S(z)H_0(z) + N_1(z)G_1(z) + N_2(z)G_2(z) + \ldots N_n(z)G_n(z) \end{aligned} \quad \text{Eq. 4}$$

When there is no signal (VAD = 0), then (suppressing the z's for clarity)

$$\begin{aligned} M_{1n} &= N_1H_1 + N_2H_2 + \ldots N_nH_n \\ M_{2n} &= N_1G_1 + N_2G_2 + \ldots N_nG_n \end{aligned} \quad \text{Eq. 5}$$

A new transfer function can now be defined, analogous to $H_1(z)$ above:

$$\tilde{H}_1 = \frac{M_{1n}}{M_{2n}} = \frac{N_1H_1 + N_2H_2 + \ldots N_nH_n}{N_1G_1 + N_2G_2 + \ldots N_nG_n} \quad \text{Eq. 6}$$

Thus $\tilde{H}_1$ depends only on the noise sources and their respective transfer functions and can be calculated any time there is no signal being transmitted. Once again, the *n* subscripts on the microphone inputs denote only that noise is being detected, while an *s* subscript denotes that only signal is being received by the microphones.

Examining Equation 4 while assuming that there is no noise produces

$$\begin{aligned} M_{1s} &= S \\ M_{2s} &= SH_0 \end{aligned}$$

Thus H_0 can be solved for as before, using any available transfer function calculating algorithm. Mathematically

$$H_0 = \frac{M_{2s}}{M_{1s}}$$

Rewriting Equation 4, using $\tilde{H}_1$ defined in Equation 6, provides,

$$\tilde{H}_1 = \frac{M_1 - S}{M_2 - SH_0} \quad \text{Eq. 7}$$

Solving for S yields,

$$S = \frac{M_1 - M_2\tilde{H}_1}{1 - H_0\tilde{H}_1} \quad \text{Eq. 8}$$

which is the same as Equation 3, with H_0 taking the place of H_2, and $\tilde{H}_1$ taking the place of H_1. Thus the noise removal algorithm still is mathematically valid for any number of noise sources, including multiple echoes of noise sources. Again, if H_0 and $\tilde{H}_1$ can be estimated to a high enough accuracy, and the above assumption of only one path from the signal to the microphones holds, the noise may be removed completely.

The most general case involves multiple noise sources and multiple signal sources. **Figure 4** is a block diagram of a front end of a noise removal algorithm of an embodiment in the most general case where there are *n* distinct noise sources and signal reflections. Here, reflections of the signal enter both microphones. This is the most general case, as reflections of the noise source into the microphones can be modeled accurately as simple additional noise sources. For clarity, the direct path from the signal to MIC 2 has changed from $H_0(z)$ to $H_{00}(z)$, and the reflected paths to Microphones 1 and 2 are denoted by $H_{01}(z)$ and $H_{02}(z)$, respectively.

The input into the microphones now becomes

$$M_1(z) = S(z) + S(z)H_{01}(z) + N_1(z)H_1(z) + N_2(z)H_2(z) + \ldots N_n(z)H_n(z)$$
$$M_2(z) = S(z)H_{00}(z) + S(z)H_{02}(z) + N_1(z)G_1(z) + N_2(z)G_2(z) + \ldots N_n(z)G_n(z) \quad \text{Eq. 9}$$

When the VAD = 0, the inputs become (suppressing the z's again)

$$M_{1n} = N_1H_1 + N_2H_2 + \ldots N_nH_n$$
$$M_{2n} = N_1G_1 + N_2G_2 + \ldots N_nG_n$$

which is the same as Equation 5. Thus, the calculation of $\tilde{H}_1$ in Equation 6 is unchanged, as expected. In examining the situation where there is no noise, Equation 9 reduces to

$$M_{1s} = S + SH_{01}$$
$$M_{2s} = SH_{00} + SH_{02}$$

This leads to the definition of $\tilde{H}_2$:

$$\tilde{H}_2 = \frac{M_{2s}}{M_{1s}} = \frac{H_{00} + H_{02}}{1 + H_{01}} \qquad \text{Eq. 10}$$

Rewriting Equation 9 again using the definition for $\tilde{H}_1$ (as in Equation 7) provides

$$\tilde{H}_1 = \frac{M_1 - S(1 + H_{01})}{M_2 - S(H_{00} + H_{02})} \qquad \text{Eq. 11}$$

Some algebraic manipulation yields

$$S(1 + H_{01} - \tilde{H}_1(H_{00} + H_{02})) = M_1 - M_2\tilde{H}_1$$
$$S(1 + H_{01})\left[1 - \tilde{H}_1\frac{(H_{00} + H_{02})}{(1 + H_{01})}\right] = M_1 - M_2\tilde{H}_1$$
$$S(1 + H_{01})[1 - \tilde{H}_1\tilde{H}_2] = M_1 - M_2\tilde{H}_1$$

and finally

$$S(1 + H_{01}) = \frac{M_1 - M_2\tilde{H}_1}{1 - \tilde{H}_1\tilde{H}_2} \qquad \text{Eq. 12}$$

Equation 12 is the same as equation 8, with the replacement of H_0 by $\tilde{H}_2$, and the addition of the $(1+H_{01})$ factor on the left side. This extra factor means that S cannot be solved for directly in this situation, but a solution can be generated for the signal plus the addition of all of its echoes. This is not such a bad situation, as there are many conventional methods for dealing with echo suppression, and even if the echoes are not suppressed, it is unlikely that they will affect the comprehensibility of the speech to any meaningful extent. The more complex calculation of $\tilde{H}_2$ is needed to account for the signal echoes in Microphone 2, which act as noise sources.

Figure 5 is a flow diagram of a denoising method of an embodiment. In operation, the acoustic signals are received 502. Further, physiological

information associated with human voicing activity is received 504. A first transfer function representative of the acoustic signal is calculated upon determining that voicing information is absent from the acoustic signal for at least one specified period of time 506. A second transfer function representative of the acoustic signal is calculated upon determining that voicing information is present in the acoustic signal for at least one specified period of time 508. Noise is removed from the acoustic signal using at least one combination of the first transfer function and the second transfer function, producing denoised acoustic data streams 510.

An algorithm for noise removal, or denoising algorithm, is described herein, from the simplest case of a single noise source with a direct path to multiple noise sources with reflections and echoes. The algorithm has been shown herein to be viable under any environmental conditions. The type and amount of noise are inconsequential if a good estimate has been made of $\tilde{H}_1$ and $\tilde{H}_2$, and if they do not change substantially while the other is calculated. If the user environment is such that echoes are present, they can be compensated for if coming from a noise source. If signal echoes are also present, they will affect the cleaned signal, but the effect should be negligible in most environments.

In operation, the algorithm of an embodiment has shown excellent results in dealing with a variety of noise types, amplitudes, and orientations. However, there are always approximations and adjustments that have to be made when moving from mathematical concepts to engineering applications. One assumption is made in Equation 3, where $H_2(z)$ is assumed small and therefore $H_2(z)H_1(z) \approx 0$, so that Equation 3 reduces to

$$S(z) \approx M_1(z) - M_2(z)H_1(z).$$

This means that only $H_1(z)$ has to be calculated, speeding up the process and reducing the number of computations required considerably. With the proper selection of microphones, this approximation is easily realized.

Another approximation involves the filter used in an embodiment. The actual $H_1(z)$ will undoubtedly have both poles and zeros, but for stability and

simplicity an all-zero Finite Impulse Response (FIR) filter is used. With enough taps (around 60) the approximation to the actual $H_1(z)$ is very good.

Regarding subband selection, the wider the range of frequencies over which a transfer function must be calculated, the more difficult it is to calculate it accurately. Therefore the acoustic data was divided into 16 subbands, with the lowest frequency at 50 Hz and the highest at 3700. The denoising algorithm was then applied to each subband in turn, and the 16 denoised data streams were recombined to yield the denoised acoustic data. This works very well, but any combinations of subbands (i.e. 4, 6, 8, 32, equally spaced, perceptually spaced, etc.) can be used and has been found to work as well.

The amplitude of the noise was constrained in an embodiment so that the microphones used did not saturate (i.e. operate outside a linear response region). It is important that the microphones operate linearly to ensure the best performance. Even with this restriction, very high signal-to-noise ratios (SNR) can be tested (down to about –10 dB).

The calculation of $H_1(z)$ was accomplished every 10 milliseconds using the Least-Mean Squares (LMS) method, a common adaptive transfer function. An explanation may be found in "Adaptive Signal Processing" (1985), by Widrow and Stearns, published by Prentice-Hall, ISBN 0-13-004029-0.

The VAD for an embodiment was derived from a radio frequency sensor and the two microphones, yielding very high accuracy (>99%) for both voiced and unvoiced speech. The VAD of an embodiment uses a radio frequency (RF) interferometer to detect tissue motion associated with human speech production, but is not so limited. It is therefore completely acoustic-noise free, and is able to function in any acoustic noise environment. A simple energy measurement can be used to determine if voiced speech is occurring. Unvoiced speech can be determined using conventional frequency-based methods, by proximity to voiced sections, or through a combination of the above. Since there is much less energy in unvoiced speech, its activation accuracy is not as critical as voiced speech.

With voiced and unvoiced speech detected reliably, the algorithm of an embodiment can be implemented. Once again, it is useful to repeat that the

noise removal algorithm does not depend on how the VAD is obtained, only that it is accurate, especially for voiced speech. If speech is not detected and training occurs on the speech, the subsequent denoised acoustic data can be distorted.

Data was collected in four channels, one for MIC 1, one for MIC 2, and two for the radio frequency sensor that detected the tissue motions associated with voiced speech. The data were sampled simultaneously at 40 kHz, then digitally filtered and decimated down to 8 kHz. The high sampling rate was used to reduce any aliasing that might result from the analog to digital process. A four-channel National Instruments A/D board was used along with Labview to capture and store the data. The data was then read into a C program and denoised 10 milliseconds at a time.

Figure 6 shows results of a noise suppression algorithm of an embodiment for an American English speaking female in the presence of airport terminal noise that includes many other human speakers and public announcements. The speaker is uttering the numbers 406-5562 in the midst of moderate airport terminal noise. The dirty acoustic data was denoised 10 milliseconds at a time, and before denoising the 10 milliseconds of data were prefiltered from 50 to 3700 Hz. A reduction in the noise of approximately 17 dB is evident. No post filtering was done on this sample; thus, all of the noise reduction realized is due to the algorithm of an embodiment. It is clear that the algorithm adjusts to the noise instantly, and is capable of removing the very difficult noise of other human speakers. Many different types of noise have all been tested with similar results, including street noise, helicopters, music, and sine waves, to name a few. Also, the orientation of the noise can be varied substantially without significantly changing the noise suppression performance. Finally, the distortion of the cleaned speech is very low, ensuring good performance for speech recognition engines and human receivers alike.

The noise removal algorithm of an embodiment has been shown to be viable under any environmental conditions. The type and amount of noise are inconsequential if a good estimate has been made of $\tilde{H}_1$ and $\tilde{H}_2$. If the user environment is such that echoes are present, they can be compensated for if

coming from a noise source. If signal echoes are also present, they will affect the cleaned signal, but the effect should be negligible in most environments.

Various embodiments are described herein with reference to the figures, but the detailed description and the figures are not intended to be limiting. Various combinations of the elements described have not been shown, but are within the scope of the invention which is defined by the following claims.

CLAIMS

What is claimed is:

1. A method for removing noise from acoustic signals, comprising:

receiving a plurality of acoustic signals;

receiving physiological information associated with human voicing activity;

generating at least one first transfer function representative of the plurality of acoustic signals upon determining that voicing information is absent from the plurality of acoustic signals for at least one specified period of time;

generating at least one second transfer function representative of the plurality of acoustic signals upon determining that voicing information is present in the plurality of acoustic signals for the at least one specified period of time;

removing noise from the plurality of acoustic signals using at least one combination of the at least one first transfer function and the at least one second transfer function to produce at least one denoised acoustic data stream.

2. The method of claim 1, wherein the plurality of acoustic signals include at least one reflection of at least one associated noise source signal and at least one reflection of at least one acoustic source signal.

3. The method of claim 1, wherein receiving physiological information comprises receiving physiological data associated with human voicing using at least one detector selected from a group consisting of radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

4. The method of claim 1, wherein receiving the plurality of acoustic signals includes receiving using a plurality of independently located microphones.

5. The method of claim 1, wherein removing noise further includes generating at least one third transfer function using the at least one first transfer function and the at least one second transfer function.

6. The method of claim 1, wherein generating the at least one first transfer function comprises recalculating the at least one first transfer function during at least one prespecified interval.

7. The method of claim 1, wherein generating the at least one second transfer function comprises recalculating the at least one second transfer function during at least one prespecified interval.

8. The method of claim 1, wherein generating the at least one first transfer function and the at least one second transfer function comprises use of at least one technique selected from a group consisting of adaptive techniques and recursive techniques.

9. A method for removing noise from electronic signals, comprising:

detecting an absence of voiced information during at least one period;

receiving at least one noise source signal during the at least one period;

generating at least one transfer function representative of the at least one noise source signal;

receiving at least one composite signal comprising acoustic and noise signals; and

removing the noise signal from the at least one composite signal using the at least one transfer function to produce at least one denoised acoustic data stream.

10. The method of claim 9, wherein the at least one noise source signal includes at least one reflection of at least one associated noise source signal.

11. The method of claim 9, wherein the at least one composite signal includes at least one reflection of at least one associated composite signal.

12. The method of claim 9, wherein detecting comprises collecting physiological data associated with human voicing using at least one detector selected from a group consisting of radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

13. The method of claim 9, wherein receiving includes receiving the at least one noise source signal using at least one microphone.

14. The method of claim 13, wherein the at least one microphone includes a plurality of independently located microphones.

15. The method of claim 9, wherein removing the noise signal from the at least one composite signal using the at least one transfer function includes generating at least one other transfer function using the at least one transfer function.

16. The method of claim 9, wherein generating at least one transfer function comprises recalculating the at least one transfer function during at least one prespecified interval.

17. The method of claim 9, wherein generating the at least one transfer function comprises calculating the at least one transfer function using at least one technique selected from a group consisting of adaptive techniques and recursive techniques.

18. A method for removing noise from electronic signals, comprising:

determining at least one unvoicing period during which voiced information is absent;

receiving at least one noise signal input during the at least one unvoicing period and generating at least one unvoicing transfer function representative of the at least one noise signal;

determining at least one voicing period during which voiced information is present;

receiving at least one acoustic signal input from at least one signal sensing device during the at least one voicing period and generating at least one voicing transfer function representative of the at least one acoustic signal;

receiving at least one composite signal comprising acoustic and noise signals; and

removing the noise signal from the at least one composite signal using at least one combination of the at least one unvoicing transfer function and the at least one voicing transfer function to produce at least one denoised acoustic data stream.

19. A system for removing noise from acoustic signals, comprising:

at least one receiver that receives at least one acoustic signal;

at least one sensor that receives physiological information associated with human voicing activity;

at least one processor coupled among the at least one receiver and the at least one sensor that generates a plurality of transfer functions, wherein at least one first transfer function representative of the at least one acoustic signal is generated in response to a determination that voicing information is absent from the at least one acoustic signal for at least one specified period of time, wherein at least one second transfer function representative of the at least one acoustic signal is generated in response to a determination that voicing information is present in the at least one acoustic signal for at least one specified period of time, wherein noise is removed from the at least one acoustic signal using at least one combination of the at least one first transfer function and the at least one second transfer function to produce at least one denoised acoustic data stream.

20. The system of claim 19, wherein the at least one sensor includes at least one radio frequency (RF) interferometer that detects tissue motion associated with human speech production.

21. The system of claim 19, wherein the at least one sensor includes at least one sensor selected from a group consisting of radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

22. The system of claim 19, further comprising:

dividing acoustic data of the at least one acoustic signal into a plurality of subbands;

removing noise from each of the plurality of subbands using the at least one combination of the at least one first transfer function and the at least one second transfer function, wherein a plurality of denoised acoustic data streams are generated; and

combining the plurality of denoised acoustic data streams to generate the at least one denoised acoustic data stream.

23. The system of claim 19, wherein the at least one receiver includes a plurality of independently located microphones.

24. A system for removing noise from acoustic signals, comprising at least one processor coupled among at least one microphone and at least one voicing sensor, wherein the at least one voicing sensor collects physiological data associated with voicing, wherein an absence of voiced information is detected during at least one period using the at least one voicing sensor, wherein at least one noise source signal is received during the at least one period using the at least one microphone, wherein the at least one processor generates at least one transfer function representative of the at least one noise source signal, wherein the at least one microphone receives at least one composite signal comprising acoustic and noise signals, and the at least one processor removes the noise

signal from the at least one composite signal using the at least one transfer function to produce at least one denoised acoustic data stream.

25. A signal processing system coupled among at least one user and at least one electronic device, wherein the signal processing system includes at least one denoising subsystem for removing noise from acoustic signals, the denoising subsystem comprising at least one processor coupled among at least one receiver and at least one sensor, wherein the at least one receiver is coupled to receive at least one acoustic signal, wherein the at least one sensor is coupled to receive physiological information associated with human voicing activity, wherein the at least one processor generates a plurality of transfer functions, wherein at least one first transfer function representative of the at least one acoustic signal is generated in response to a determination that voicing information is absent from the at least one acoustic signal for at least one specified period of time, wherein at least one second transfer function representative of the at least one acoustic signal is generated in response to a determination that voicing information is present in the at least one acoustic signal for at least one specified period of time, wherein noise is removed from the at least one acoustic signal using at least one combination of the at least one first transfer function and the at least one second transfer function to produce at least one denoised acoustic data stream.

26. The signal processing system of claim 25, wherein the at least one electronic device includes at least one device selected from a group consisting of cellular telephones, personal digital assistants, portable communication devices, computers, video cameras, digital cameras, and telematics systems.

27. A computer readable medium comprising executable instructions which, when executed in a processing system, remove noise from received acoustic signals by:

receiving at least one acoustic signal;

receiving physiological information associated with human voicing activity;

generating at least one first transfer function representative of the at least one acoustic signal upon determining that voicing information is absent from the at least one acoustic signal for at least one specified period of time;

generating at least one second transfer function representative of the at least one acoustic signal upon determining that voicing information is present in the at least one acoustic signal for at least one specified period of time;

removing noise from the at least one acoustic signal using at least one combination of the at least one first transfer function and the at least one second transfer function to produce at least one denoised acoustic data stream.

28. An electromagnetic medium comprising executable instructions which, when executed in a processing system, remove noise from received acoustic signals by:

receiving at least one acoustic signal;

receiving physiological information associated with human voicing activity;

generating at least one first transfer function representative of the at least one acoustic signal upon determining that voicing information is absent from the at least one acoustic signal for at least one specified period of time;

generating at least one second transfer function representative of the at least one acoustic signal upon determining that voicing information is present in the at least one acoustic signal for at least one specified period of time;

removing noise from the at least one acoustic signal using at least one combination of the at least one first transfer function and the at least one second transfer function to produce at least one denoised acoustic data stream.

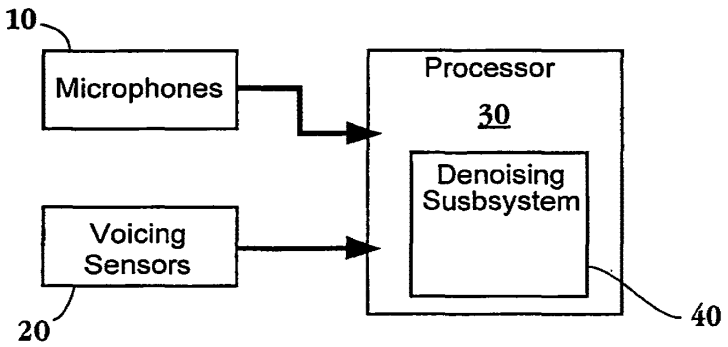

10
Microphones
Processor
30
Denoising
Susbsystem
Voicing
Sensors
20
40

Fig. 1

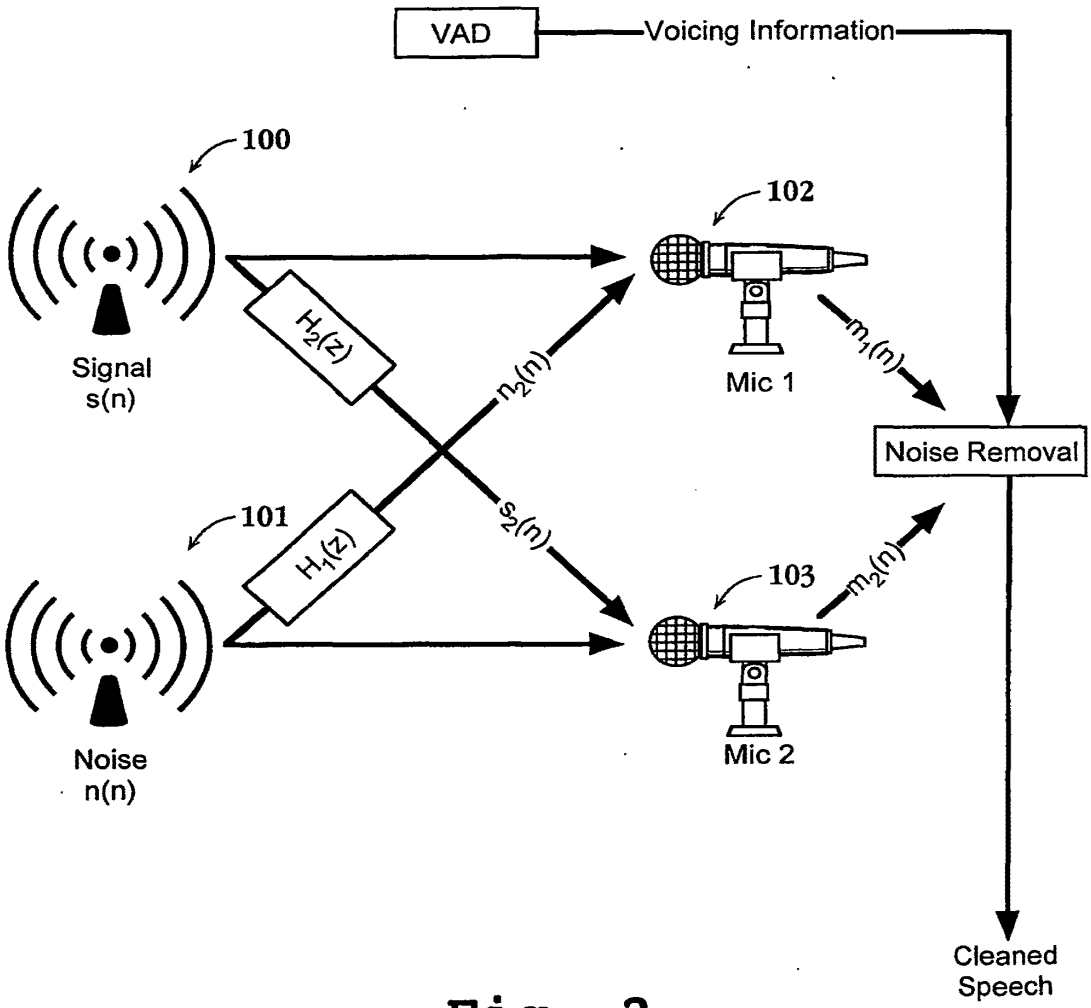

VAD
Voicing Information
100
Signal
s(n)
102
Mic 1
$H_2(z)$
$n_2(n)$
$m_1(n)$
Noise Removal
101
$H_1(z)$
$s_2(n)$
103
$m_2(n)$
Noise
n(n)
Mic 2
Cleaned
Speech

Fig. 2

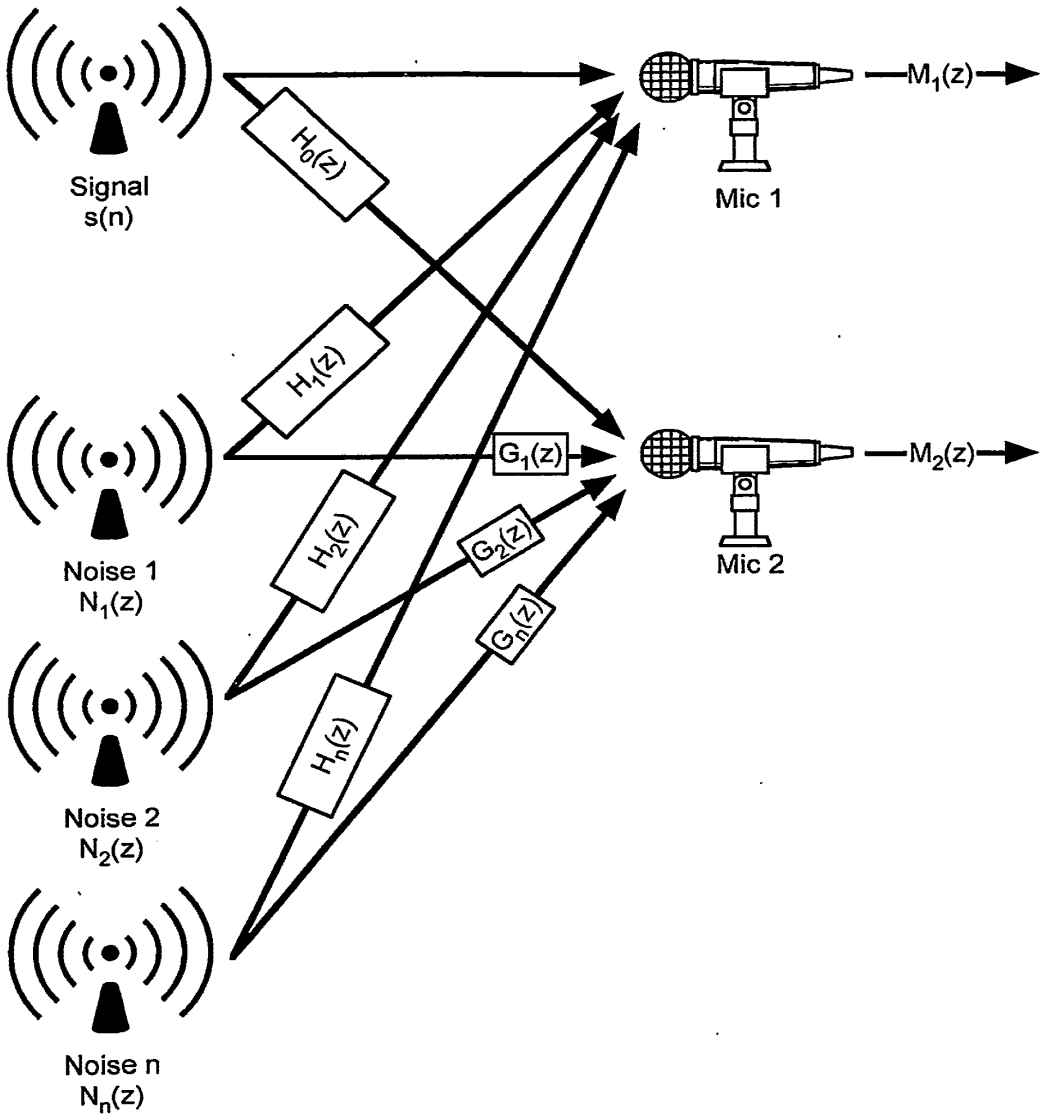

Signal
s(n)
Noise 1
N1(z)
Noise 2
N2(z)
Noise n
Nn(z)
H0(z)
H1(z)
H2(z)
Hn(z)
G1(z)
G2(z)
Gn(z)
Mic 1
Mic 2
M1(z)
M2(z)

Fig. 3

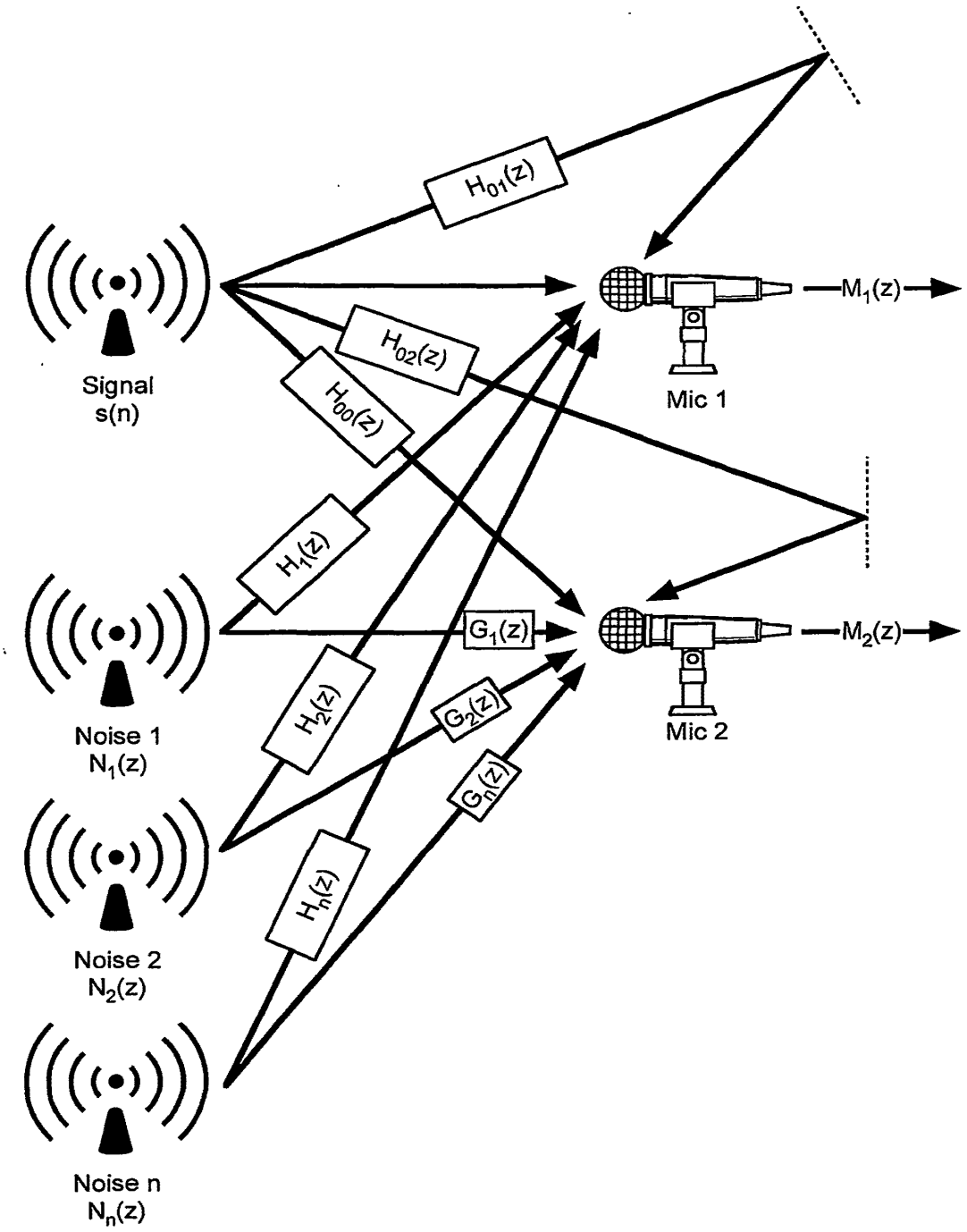

$H_{01}(z)$
$H_{02}(z)$
$H_{00}(z)$
$H_1(z)$
$H_2(z)$
$H_n(z)$
$G_1(z)$
$G_2(z)$
$G_n(z)$
$M_1(z)$
$M_2(z)$
Signal
s(n)
Noise 1
$N_1(z)$
Noise 2
$N_2(z)$
Noise n
$N_n(z)$
Mic 1
Mic 2

Fig. 4

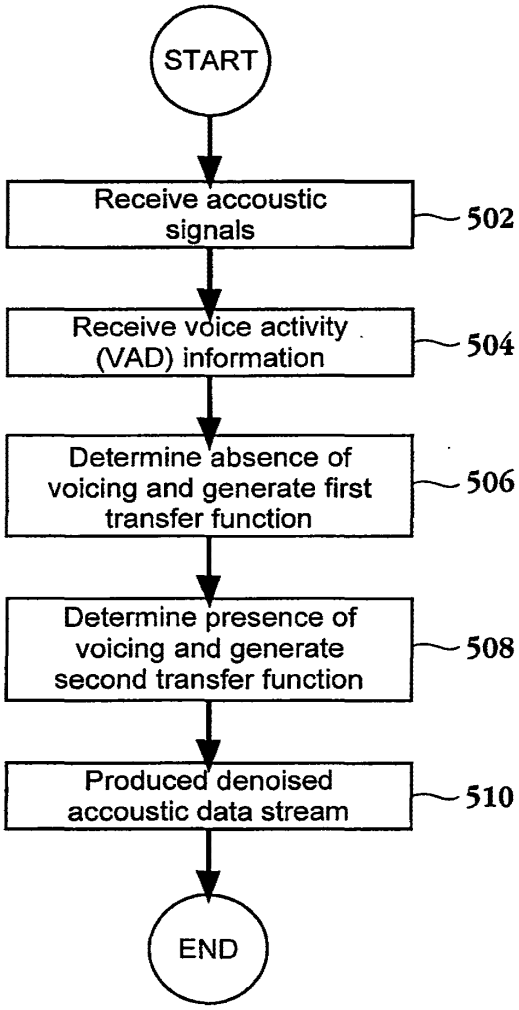

START
Receive accoustic signals
502
Receive voice activity (VAD) information
504
Determine absence of voicing and generate first transfer function
506
Determine presence of voicing and generate second transfer function
508
Produced denoised accoustic data stream
510
END

Fig. 5

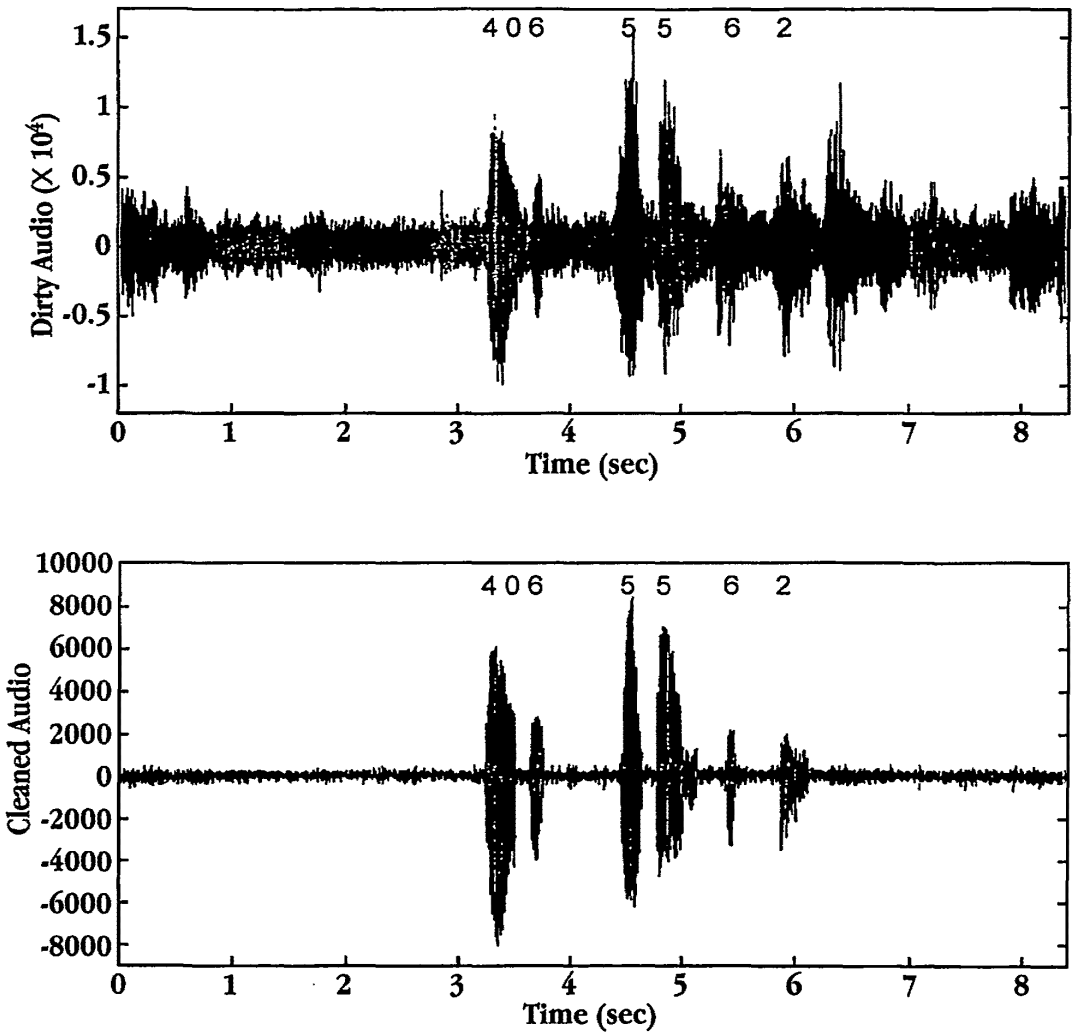

4 0 6 5 5 6 2
Dirty Audio (X 10^4)
1.5
1
0.5
0
-0.5
-1
0 1 2 3 4 5 6 7 8
Time (sec)
4 0 6 5 5 6 2
Cleaned Audio
10000
8000
6000
4000
2000
0
-2000
-4000
-6000
-8000
0 1 2 3 4 5 6 7 8
Time (sec)

Fig. 6

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2644
#5
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Atty. Docket No. ALPH P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:

Gregory C. Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Examiner: To be assigned

Art Unit: 2644

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Assistant Commissioner for Patents
Washington, D.C. 20231

INFORMATION DISCLOSURE STATEMENT UNDER 37 CFR 1.97

Sir:

Enclosed is an Information Disclosure Citation Form PTO/SB/08 together with copies of the references cited therein. It is respectfully requested that the cited references be considered and that the enclosed copy of the Form PTO/SB/08 be initialed by the Examiner to indicate such consideration and a copy thereof returned to Applicant.

Pursuant to 37 CFR 1.97(h), the submission of this Information Disclosure Statement is not to be construed as an admission that the information cited in this statement is material to patentability.

This Information Disclosure Statement is being submitted pursuant to 37 CFR 1.97(b)(3).

The Commissioner is hereby authorized to charge any fees which may be required in connection with this submission to Deposit Account No. 501914.

Respectfully submitted,

SHEMWELL GREGORY & COURTNEY LLP

Dated: December 21, 2002

Barbara B. Courtney
Registration No. 42,442

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(use as many sheets as necessary)

Sheet 1 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	2644
Examiner Name	Unknown
Attorney Docket Number	ALPH.P009

U.S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	U.S. Patent Document Number	Kind Code[2] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
		5,406,622		Silverberg et al.	04-11-1995	
		5,517,435		Sugiyama	05-14-1996	

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FOREIGN PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	Foreign Patent Document Office[3]	Number[4]	Kind Code[5] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear	T[6]

Examiner Signature		Date Considered	

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[1] Unique citation designation number. [2] See attached Kinds of U.S. Patent Documents. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST. 16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

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Sheet 2 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	2644
Examiner Name	Unknown
Attorney Docket Number	ALPH.P009

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OTHER PRIOR ART -- NON PATENT LITERATURE DOCUMENTS			
Examiner Initials*	Cite No.[1]	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.), date, page(s), volume-issue number(s), publisher, city and/or country where published.	T[2]
	✓	GREGORY C. BURNETT: "The Physiological Basis of Glottal Electromagnetic Micropower Sensors (GEMS) and Their Use in Defining an Excitation Function for the Human Vocal Tract", Dissertation, University of California at Davis, January, 1999, USA.	
	✓	TODD J. GABLE et al.: "Speaker Verification Using Combined Acoustic and EM Sensor Signal Processing", IEEE Intl. Conf. on Acoustics, Speech & Signal Processing (ICASSP-2001), Salt Lake City, USA, 2001.	
	✓	A. HUSSAIN: "Intelligibility Assessment of a Multi-Band Speech Enhancement Scheme", Proceedings IEEE Intl. Conf. on Acoustics, Speech & Signal Processing (ICASSP-2000). Istanbul, Turkey, June 2000.	

Examiner Signature		Date Considered	

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2644
11-29-02

Atty. Docket No. ALPH.P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:

Gregory C. Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Examiner: To be assigned

Art Unit: 2644

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INFORMATION DISCLOSURE STATEMENT UNDER 37 CFR 1.97

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Enclosed is an Information Disclosure Citation Form PTO/SB/08 together with copies of the references cited therein. It is respectfully requested that the cited references be considered and that the enclosed copy of the Form PTO/SB/08 be initialed by the Examiner to indicate such consideration and a copy thereof returned to Applicant.

Pursuant to 37 CFR 1.97(h), the submission of this Information Disclosure Statement is not to be construed as an admission that the information cited in this statement is material to patentability.

This Information Disclosure Statement is being submitted pursuant to 37 CFR 1.97(b)(3).

The Commissioner is hereby authorized to charge any fees which may be required in connection with this submission to Deposit Account No. 501914.

Respectfully submitted,

SHEMWELL GREGORY & COURTNEY LLP

Dated: November 18, 2002

Barbara B. Courtney
Registration No. 42,442

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Sheet 1 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	2644
Examiner Name	Unknown
Attorney Docket Number	ALPH.P009

U.S. PATENT DOCUMENTS

Examiner Initials*	Cite No.[1]	U.S. Patent Document Number	Kind Code[2] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
		5,212,764		Ariyoshi	05-18-1993	
		5,400,409		Linhard	03-21,1995	
		5,414,776		Sims, Jr.	05-09-1995	
		5,539,859		Robbe et al.	07-23-1996	
		5,633,935		Kanamori et al.	05-27-1997	
		5,835,608		Warnaka et al.	11-10-1998	
		5,917,921		Sasaki et al.	06-29-1999	
		6,000,396		Nagata	12-28-1999	
		6,191,724	B1	McEwan	02-20-2001	
		5,986,600		McEwan	11-16-1999	
		5,966,090		McEwan	10-12-1999	
		6,006,175		Holzrichter	12-21-1999	

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Examiner Initials*	Cite No.[1]	Foreign Patent Document Office[3]	Number[4]	Kind Code[5] (if known)	Name of Patentee or Applicant of Cited Document	Date of Publication of Cited Document MM-DD-YYYY	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear	T[6]

Examiner Signature		Date Considered	

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[1] Unique citation designation number. [2] See attached Kinds of U.S. Patent Documents. [3] Enter Office that issued the document, by the two-letter code (WIPO Standard ST.3). [4] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [5] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST. 16 if possible. [6] Applicant is to place a check mark here if English language Translation is attached.

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

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Sheet 2 of 2

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Group Art Unit	2644
Examiner Name	Unknown
Attorney Docket Number	ALPH.P009

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OTHER PRIOR ART – NON PATENT LITERATURE DOCUMENTS			
Examiner Initials*	Cite No.[1]	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.), date, page(s), volume-issue number(s), publisher, city and/or country where published.	T[2]
	✓	Zhao Li et al: "Robust Speech Coding Using Microphone Arrays", Signals Systems and Computers, 1997. Conf. record of 31st Asilomar Conf., Nov. 2-5, 1997, IEEE Comput. Soc. Nov. 2, 1997, USA.	
	✓	L.C. Ng et al.: "Denoising of Human Speech Using Combined Acoustic and EM Sensor Signal Processing", 2000 IEEE Intl Conf on Acoustics Speech and Signal Processing. Proceedings (Cat. No. 00CH37100), Istanbul, Turkey, June 5-9, 2000 XP002186255, ISBN 0-7803-6293-4	
	✓	S. Affes et al.: "A Signal Subspace Tracking Algorithm for Microphone Array Processing of Speech". IEEE Transactions on Speech and Audio Processing, N.Y, USA Vol. 5, No. 5, Sept. 1, 1997, XP000774303, ISBN 1063-6676	

Examiner Signature		Date Considered	

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[1] Unique citation designation number. [2] Applicant is to place a check mark here if English language Translation is attached.

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Atty. Docket No. ALPH.P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

COPY OF PAPERS ORIGINALLY FILED

In Re Patent Application of:

Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Examiner: Not Yet Assigned

Art Unit: 2644

I hereby certify that this correspondence is being deposited with the United States Postal I Service as first class mail with sufficient postage in an envelope addressed to the Commissioner of Patents and Trademarks, Washington, D.C. 20231 on August 01, 2002

Richard L. Gregory, Jr.
(Name of Person Mailing Correspondence)

R.L. Gregory 08-01-2002
Signature Date

Box Missing Parts
Assistant Commissioner for Patents
Washington, D.C. 20231

RESPONSE TO NOTICE TO FILE MISSING PARTS

Sir:

In response to a Notice to File Missing Parts mailed July 12, 2002 in connection with the above-referenced application, applicant has enclosed the following:

(1) a Declaration pursuant to 37 CFR 1.63;

(2) a check in the amount of $130.00 for the fee under 37 CFR 1.16(e);

(3) a copy of the Notice to File Missing Parts; and

(4) replacement drawings for Figures 1-10 (10 sheets).

Please charge Deposit Account No. 501914 for any fee deficiency that may be due. A duplicate of this Response is enclosed for Deposit Account charging purposes.

Respectfully Submitted,

Date 08-01-2002

R.L. Gregory
Richard L. Gregory, Jr.
Reg. No. 42,607

UNITED STATES PATENT AND TRADEMARK OFFICE

COMMISSIONER FOR PATENTS
UNITED STATES PATENT AND TRADEMARK OFFICE
WASHINGTON, D.C. 20231
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AUG 0 8 2002
JC144
PATENT & TRADEMARK OFFICE

APPLICATION NUMBER	FILING/RECEIPT DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NUMBER
10/159,770	05/30/2002	Gregory C. Burnett	37212.8009.US00

CONFIRMATION NO. 9714

22918
PERKINS COIE LLP
P.O. BOX 2168
MENLO PARK, CA 94026

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FORMALITIES LETTER

OC000000008443147

Date Mailed: 07/12/2002

NOTICE TO FILE MISSING PARTS OF NONPROVISIONAL APPLICATION

FILED UNDER 37 CFR 1.53(b)

Filing Date Granted

Items Required To Avoid Abandonment:

An application number and filing date have been accorded to this 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 oath or declaration is unsigned.
- To avoid abandonment, a late filing fee or oath or declaration surcharge as set forth in 37 CFR 1.16(l) of $130 for a non-small entity, must be submitted with the missing items identified in this letter.

The application is informal since it does not comply with the regulations for the reason(s) indicated below.
The required item(s) identified below must be timely submitted to avoid abandonment:

- Replacement drawings in compliance with 37 CFR 1.84 and 37 CFR 1.121 are required. The drawings submitted are not acceptable because:
  - The drawings must be made on paper that has a white background, (see 37 CFR 1.84(e)). For example. Graph paper, lined paper, or paper that has a non-white background is not acceptable.

Items Required To Avoid Processing Delays:
The item(s) indicated below are also required and should be submitted with any reply to this notice to avoid further processing delays.

SUMMARY OF FEES DUE:
Total additional fee(s) required for this application is **$130** for a Large Entity

- **$130** Late oath or declaration Surcharge.

08/09/2002 BABRAHA1 00000063 10159770
01 FC:105 130.00 OP

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WASHINGTON, D.C. 20231
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*A copy of this notice **MUST** be returned with the reply.*

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Page 279 of 2754

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PATENT & TRADEMARK OFFICE

Atty. Docket No. ALPH.P009 *PATENT*

IN THE UNITED STATES PATENT OFFICE

In Re Patent Application of:

Burnett

Application No. 10/159,770

Filed: May 30, 2002

For: DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

Examiner: Not Yet Assigned

Art Unit: 2644

I hereby certify that this correspondence is being deposited with the United States Postal l Service as first class mail with sufficient postage in an envelope addressed to the Commissioner of Patents and Trademarks, Washington, D.C. 20231 on August 01, 2002
Richard L. Gregory, Jr.
(Name of Person Mailing Correspondence)
R. L. Gregory 08-01-2002
Signature Date

Assistant Commissioner for Patents
Washington, D.C. 20231

TRANSMITTAL OF FORMAL DRAWINGS

Sir:

Enclosed herewith for filing in the above-identified patent application are ten (10) sheets of formal drawings.

The Commissioner is hereby authorized to charge any deficiencies in fees or credit any overpayment to our Deposit Account No. 501914. A duplicate of this transmittal letter is enclosed.

Respectfully Submitted,

Date 08-01-2002

R. L. Gregory

Richard L. Gregory, Jr.
Reg. No. 42,607

09/159770 05/30/02

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#3

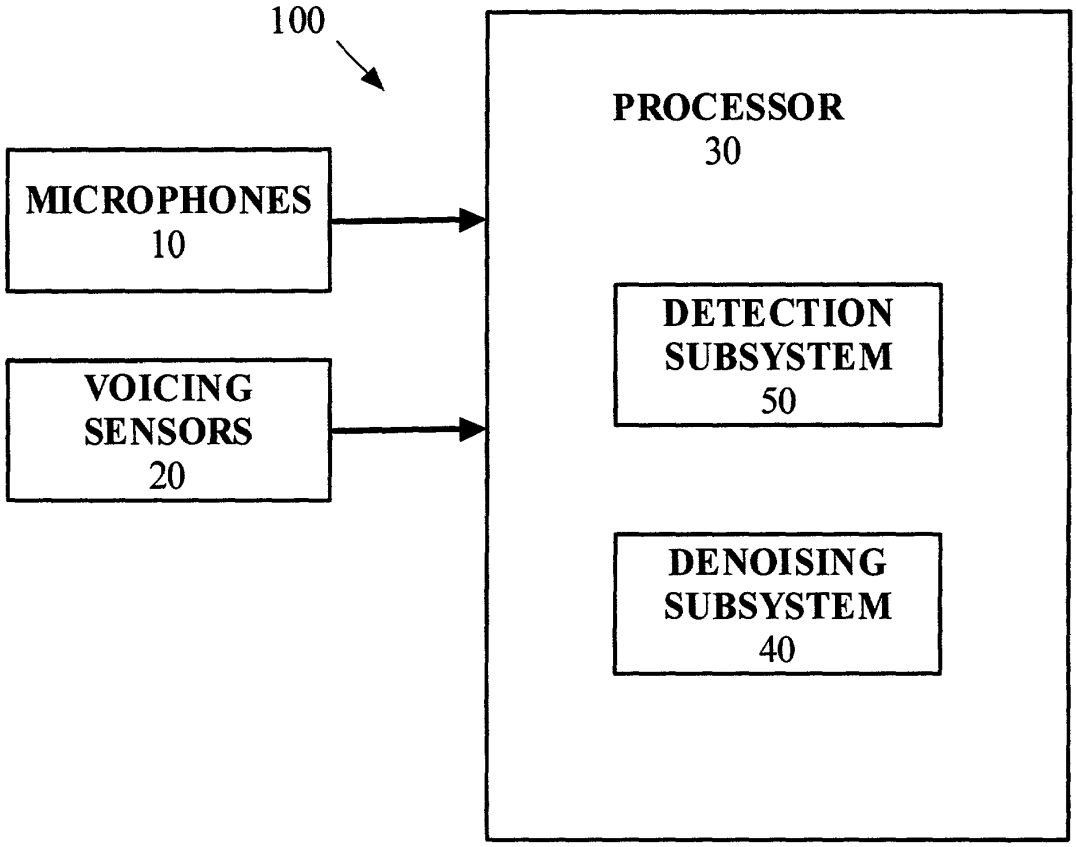


Figure 1

20020807.0745.0001

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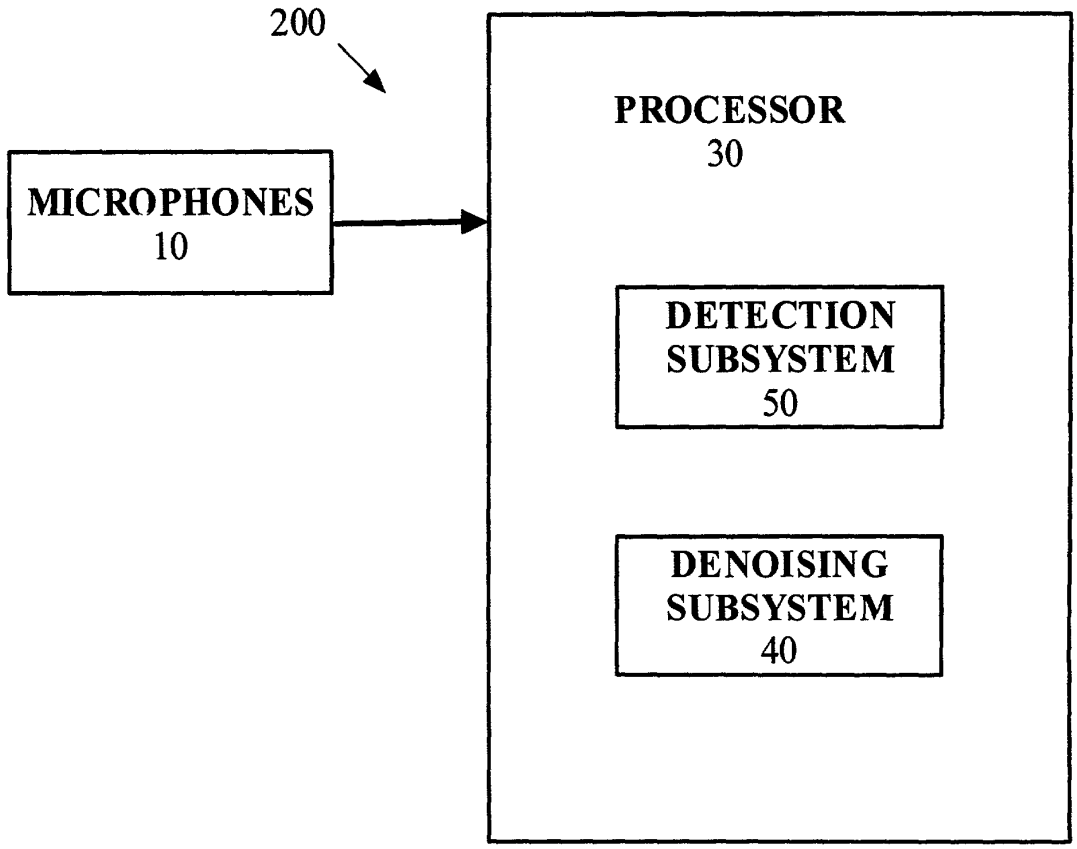


Figure 2

20808 "07659701

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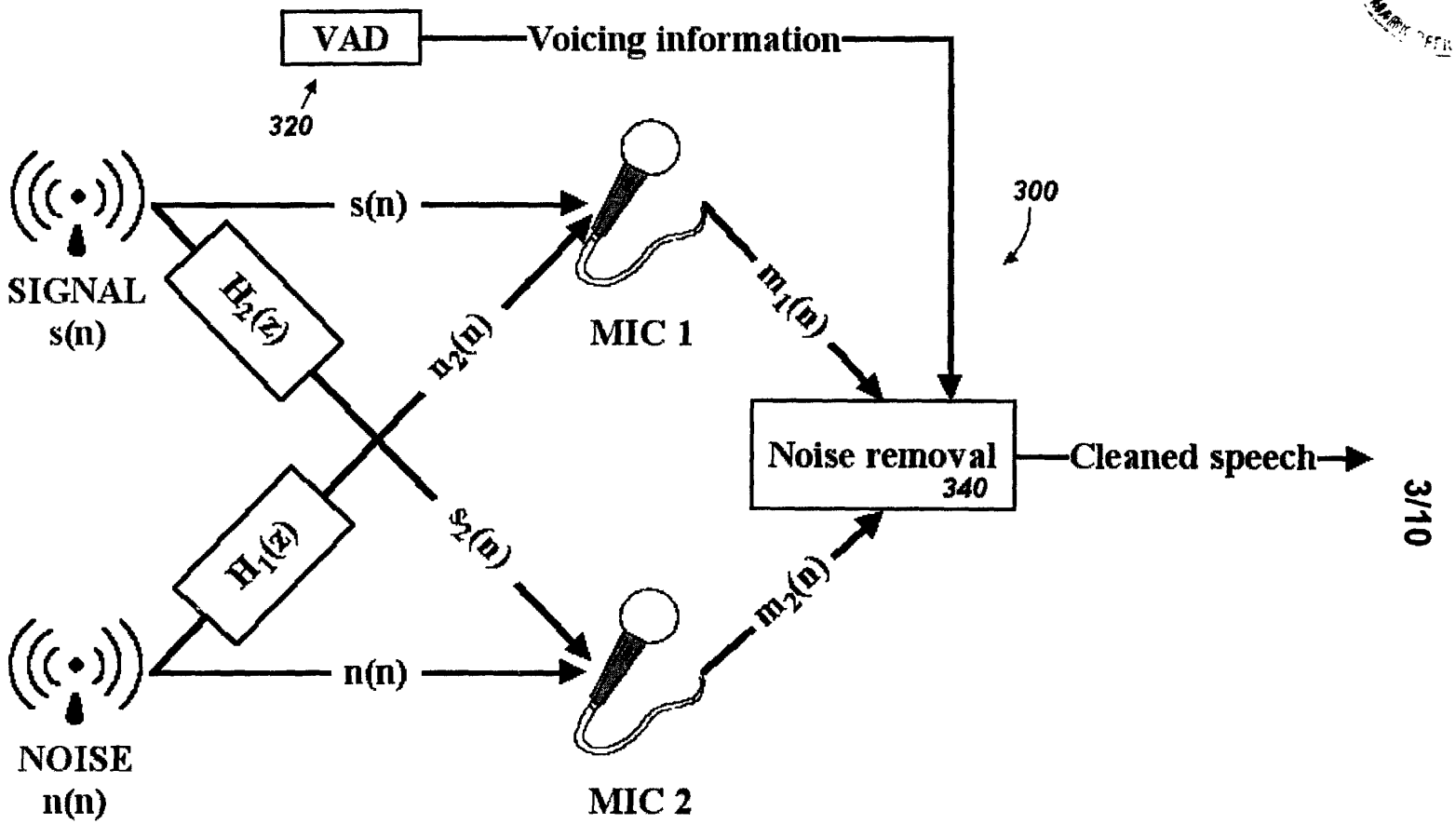
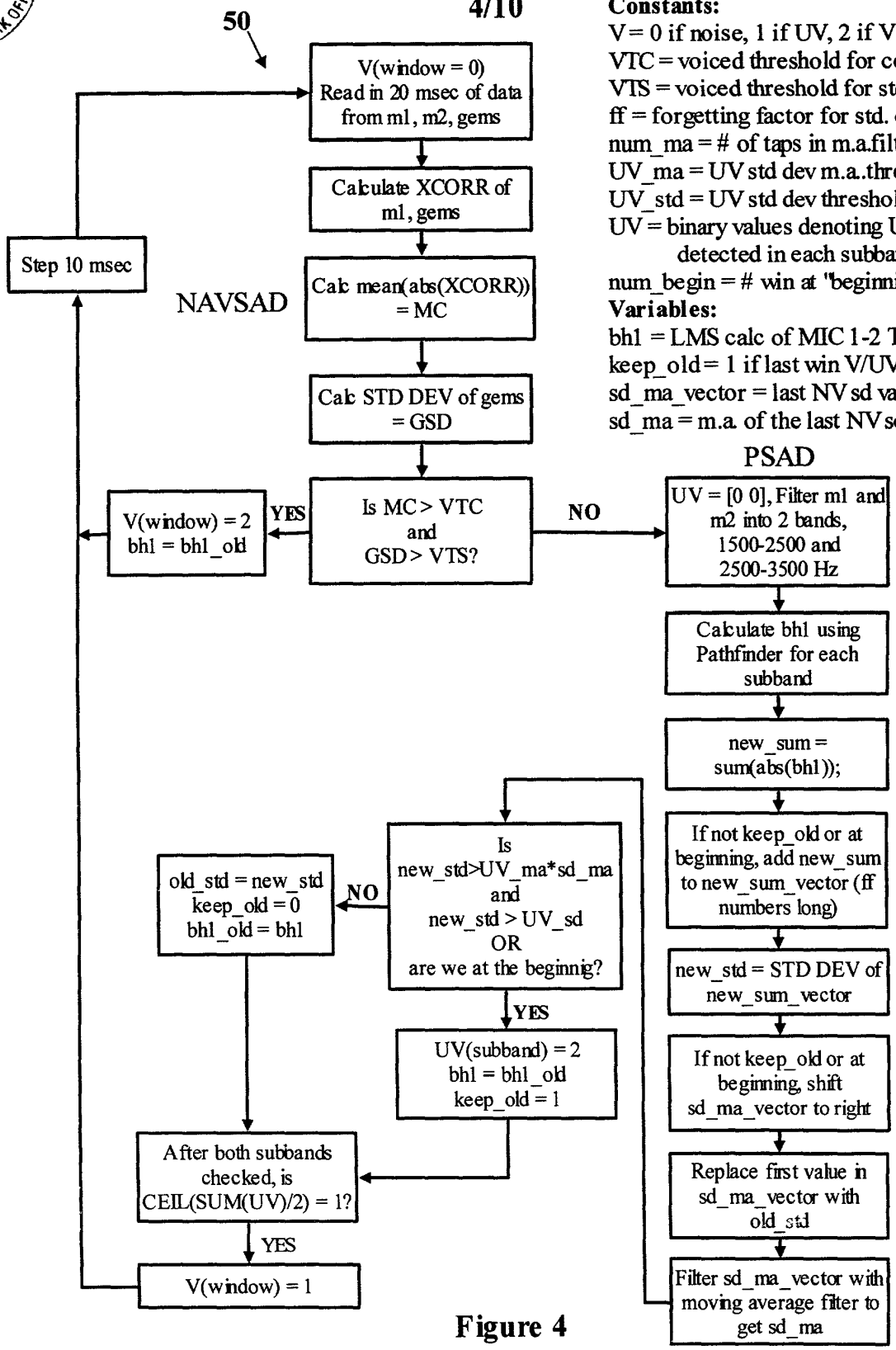


Figure 3

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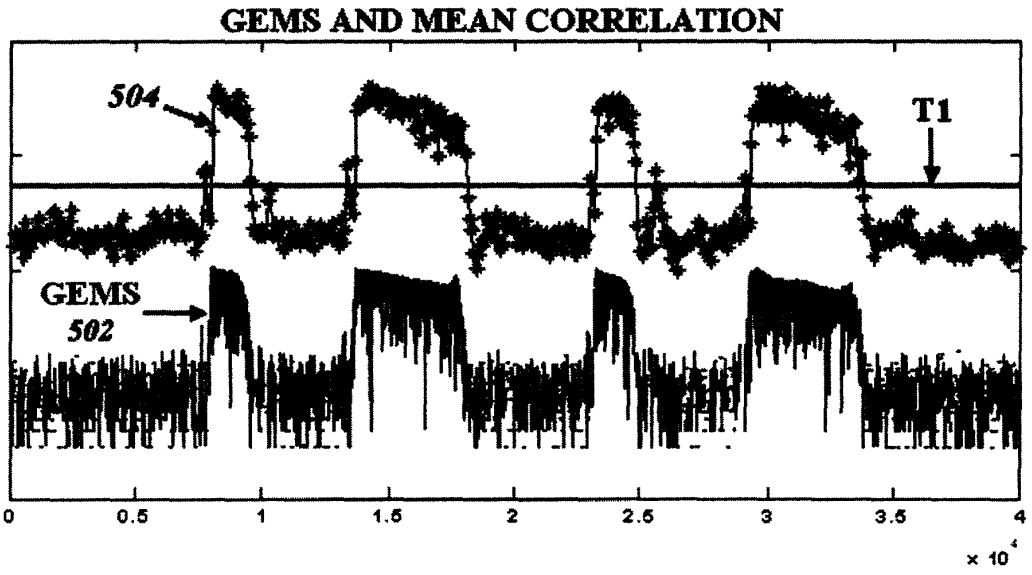
Constants:
V = 0 if noise, 1 if UV, 2 if V
VTC = voiced threshold for corr
VTS = voiced threshold for std. dev.
ff = forgetting factor for std. dev.
num_ma = # of taps in m.a.filter
UV_ma = UV std dev m.a..thresh
UV_std = UV std dev threshold
UV = binary values denoting UV detected in each subband
num_begin = # win at "beginning"

Variables:
bh1 = LMS calc of MIC 1-2 TF
keep_old = 1 if last win V/UV, 0 ow
sd_ma_vector = last NV sd values
sd_ma = m.a. of the last NV sd

Figure 4

OIPE JC144 AUG 0 8 2002 PATENT & TRADEMARK OFFICE

Figure 5A



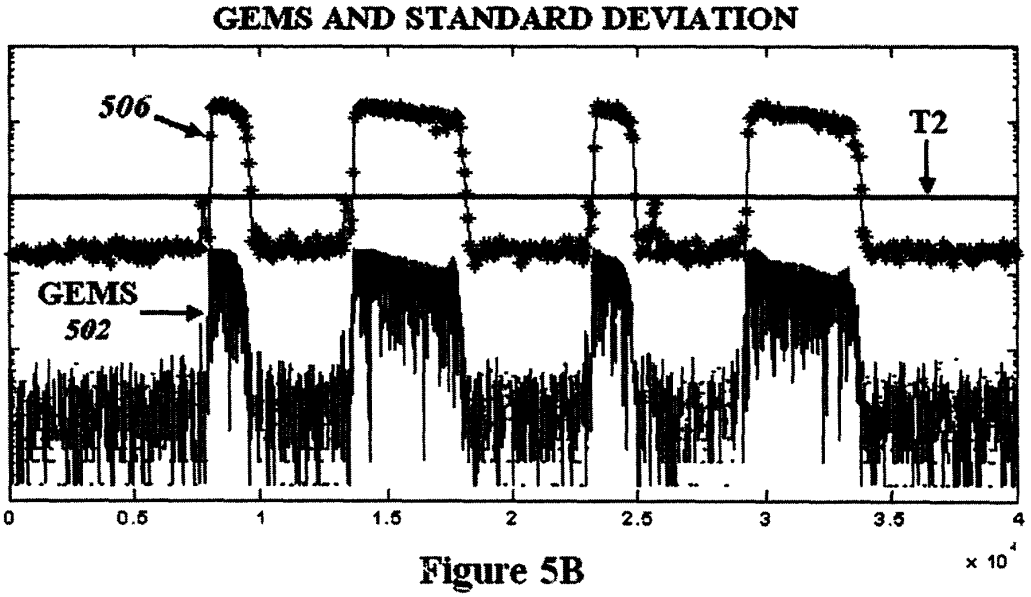


Figure 5B

20020807.0157.0

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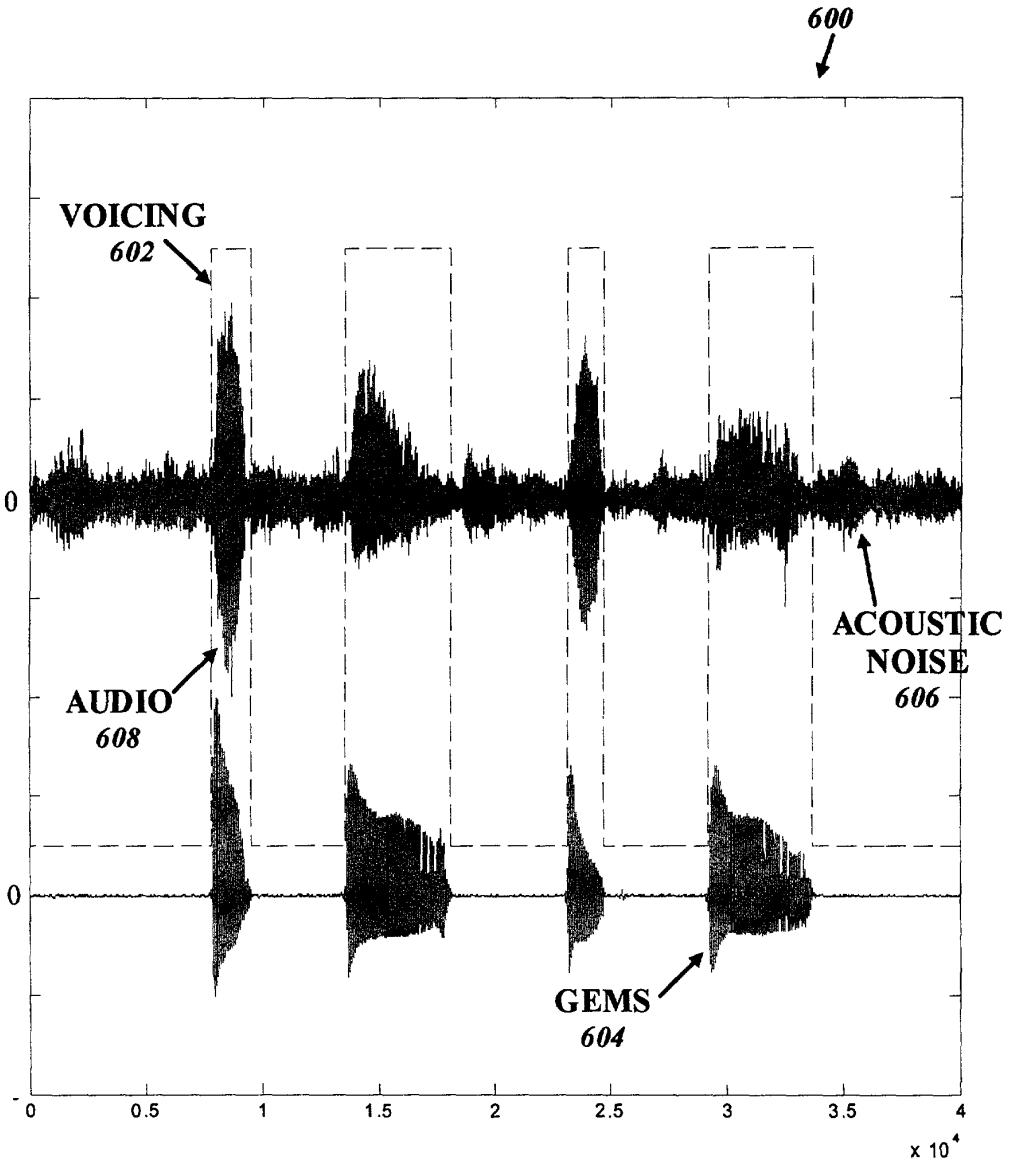


Figure 6

OIPE JC14
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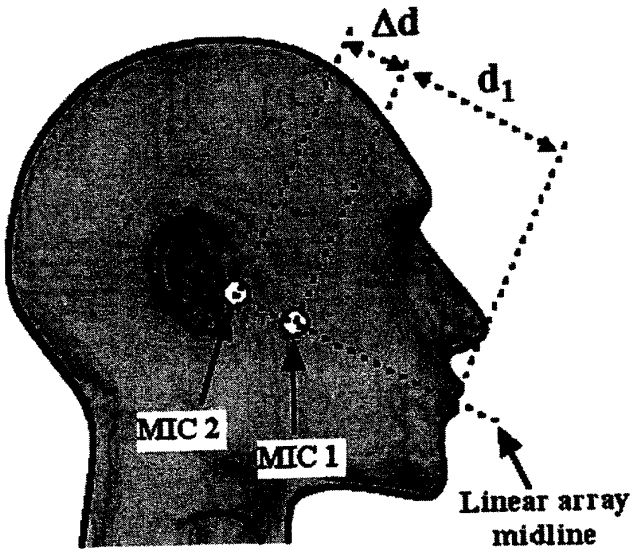


Figure 7

20020802.0247651

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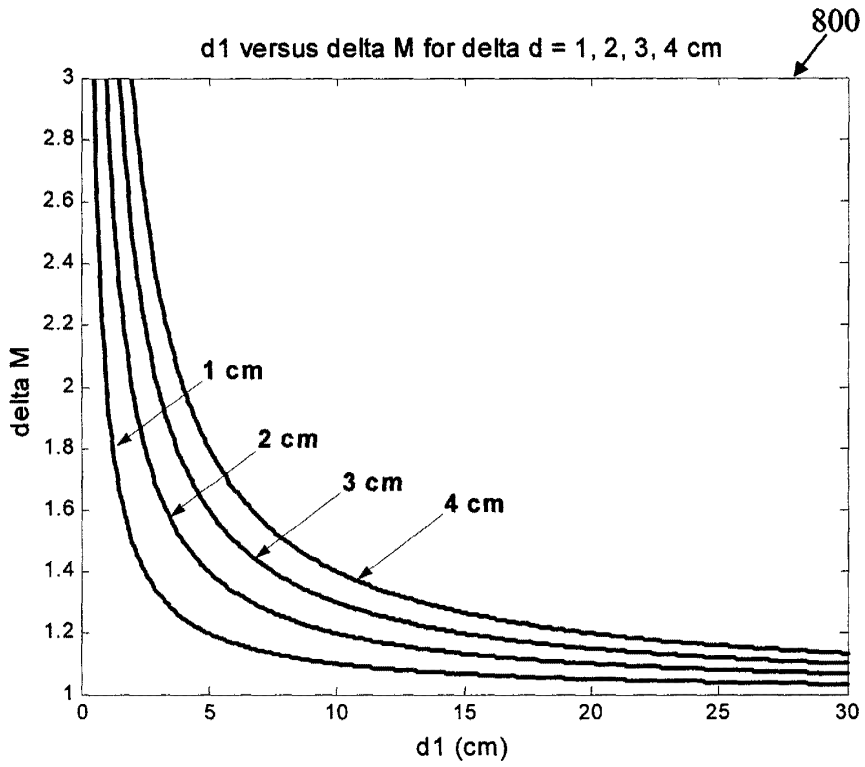


Figure 8

20020820-02751.txt

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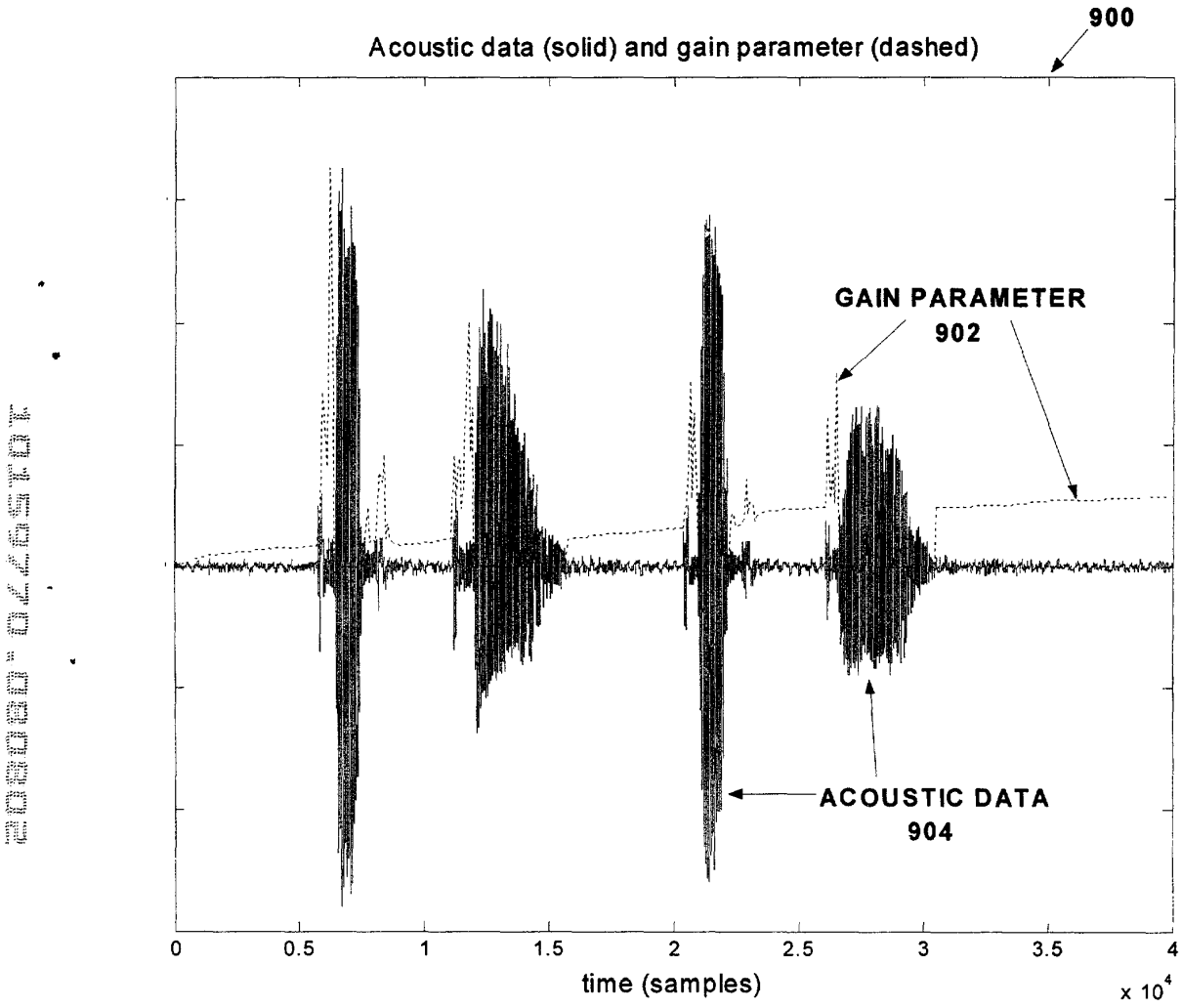


Figure 9

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AUG 0 8 2002
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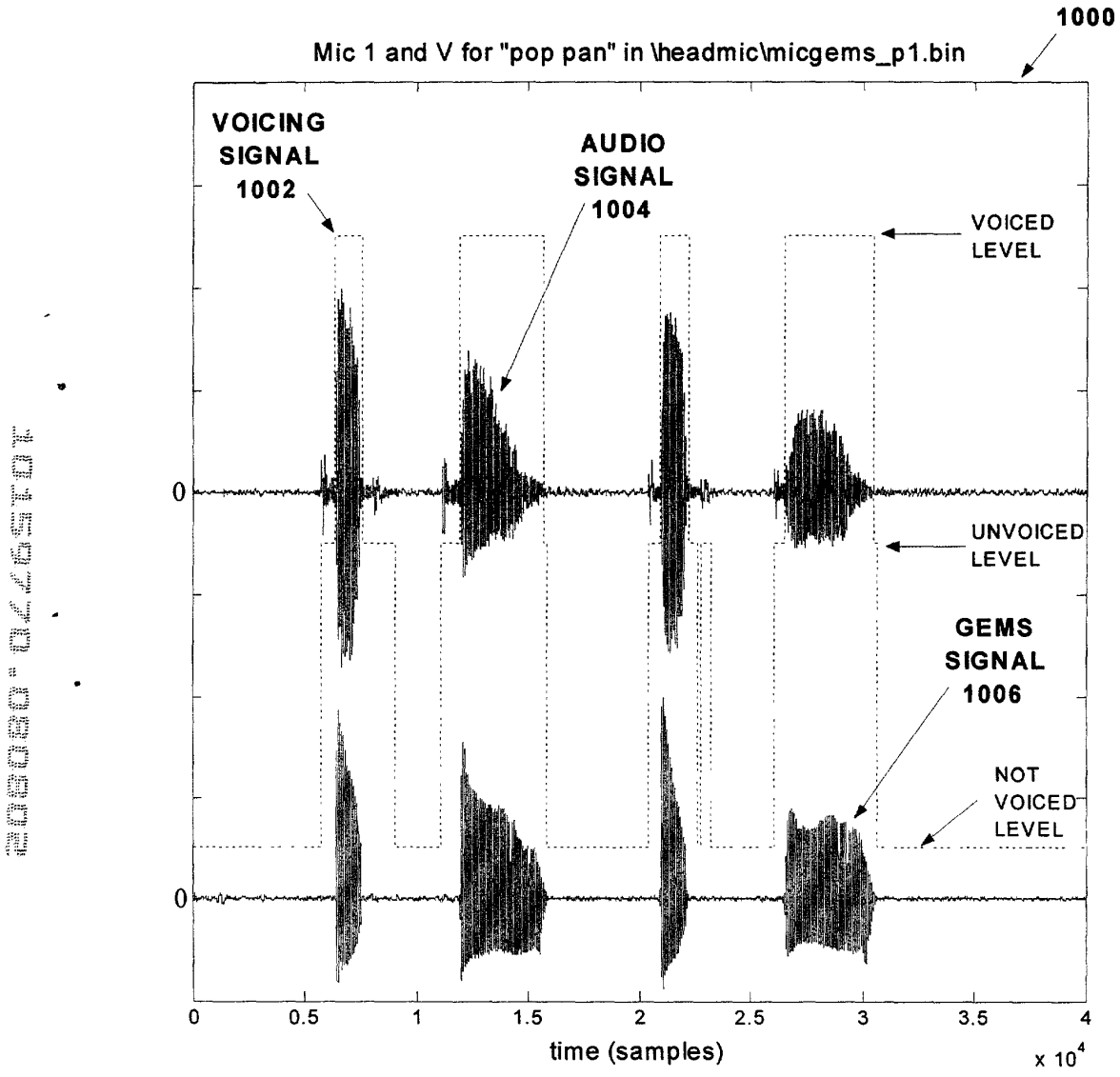


Figure 10

AUG 0 8 2002
PATENT & TRADEMARK OFFICE

INVENTORSHIP DECLARATION BY SOLE INVENTOR

I HEREBY DECLARE THAT:

My residence, mailing address, and citizenship are stated next to my name in PART A on hereof.

I believe I am the original, first, and sole inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

the specification of which:

☒ is attached hereto.

☐ was filed on _____ as Application Serial No. _____ and was amended on _____ (if applicable).

I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56.

I hereby claim priority benefits under Title 35, United States Code §119 of any foreign application(s) for patent or inventor's certificate listed in PART B hereof and have also identified in PART B hereof any foreign application for patent or inventor's certificate having a filing date before that of the application of which priority is claimed.

I hereby claim the benefit under Title 35, United States Code, §119(e) and §120 of any United States application(s) listed in PART C hereof and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph by the first paragraph of Title 35, United States Code §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations §1.56 which occurred between the filing date of the prior application and any national or PCT international filing date of this application.

I hereby declare that all statements made herein of my knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Please direct all telephone calls to Richard L. Gregory, Jr. at (650) 838-4408. Address all correspondence to:

Customer No. 22918
Perkins Coie LLP
101 Jefferson Drive
Menlo Park, California 94025-1114
(650) 838-4300

See page 2 attached, signed, and made a part hereof.

INVENTORSHIP DECLARATION BY SOLE INVENTOR

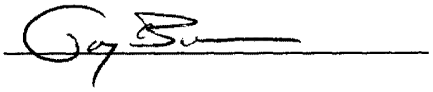
PART A: INVENTOR INFORMATION AND SIGNATURE

Full name of Sole inventor: Gregory C. Burnett

Citizenship: U.S.A.

Mailing Address: 675 South H Street
Livermore, CA 94550

Residence (if different): _____

Inventor's Signature:  Date: 6/20/02

PART B: CLAIM TO PRIORITY OF FOREIGN APPLICATION(S) UNDER 35 U.S.C. 119(a-d)

Country	App. No.	Filing Date

PART C: CLAIM TO PRIORITY OF U.S. PROVISIONAL AND NONPROVISIONAL APPLICATION(S) UNDER 35 U.S.C. 119(e) AND 120

Serial No.	Filing Date	Status
60/294,383	5/30/01	☒ Pending
09/905,361	7/12/01	☒ Pending
60/335,100	10/30/01	☒ Pending
60/332,202	11/21/01	☒ Pending
09/990,847	11/21/01	☒ Pending
60/362,162	3/5/02	☒ Pending
60/362,103	3/5/02	☒ Pending
60/362,170	3/5/02	☒ Pending
60/361,981	3/5/02	☒ Pending
60/362,161	3/5/02	☒ Pending
60/368,209	3/27/02	☒ Pending
60/368,208	3/27/02	☒ Pending
60/368,343	3/27/02	☒ Pending

Express Mail Label No. EV 076 080 930 US

OIPE JC14 AUG 08 2002 PATENT & TRADEMARK OFFICE

COPY OF PAPERS ORIGINALLY FILED

PTO/SB/17 (11-01)
Approved for use through 10/31/2002. 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.

FEE TRANSMITTAL for FY 2002

Patent fees are subject to annual revision.

☐ Applicant claims small entity status. See 37 CFR 1.27

TOTAL AMOUNT OF PAYMENT ($) 130.00

Complete if Known	
Application Number	10/159,770
Filing Date	May 30, 2002
First Named Inventor	Gregory C. Burnett
Examiner Name	
Group Art Unit	2644
Attorney Docket No.	ALPH.P009

METHOD OF PAYMENT *(check all that apply)*

☒ Check ☐ Credit card ☐ Money Order ☐ Other ☐ None

☐ Deposit Account:

Deposit Account Number: 501914

Deposit Account Name:

The Commissioner is authorized to: *(check all that apply)*

☐ Charge fee(s) indicated below ☒ Credit any overpayments

☒ Charge any additional fee(s) during the pendency of this application

☐ Charge fee(s) indicated below, **except for the filing fee** to the above-identified deposit account.

FEE CALCULATION

1. BASIC FILING FEE

Large Entity Fee Code	Large Entity Fee ($)	Small Entity Fee Code	Small Entity Fee ($)	Fee Description	Fee Paid
101	740	201	370	Utility filing fee	
106	330	206	165	Design filing fee	
107	510	207	255	Plant filing fee	
108	740	208	370	Reissue filing fee	
114	160	214	80	Provisional filing fee	

SUBTOTAL (1) ($) 0.00

2. EXTRA CLAIM FEES FOR UTILITY AND REISSUE

	Extra Claims		Fee from below	Fee Paid
Total Claims	-20** =	-20	X $18	=
Independent Claims	- 3** =	-3	X $84	=
Multiple Dependent				=

Large Entity Fee Code	Large Entity Fee ($)	Small Entity Fee Code	Small Entity Fee ($)	Fee Description
103	18	203	9	Claims in excess of 20
102	84	202	42	Independent claims in excess of 3
104	280	204	140	Multiple dependent claim, if not paid
109	84	209	42	** Reissue independent claims over original patent
110	18	210	9	** Reissue claims in excess of 20 and over original patent

SUBTOTAL (2) ($) 0.00

***or number previously paid, if greater; For Reissues, see above*

FEE CALCULATION (continued)

3. ADDITIONAL FEES

Large Entity Fee Code	Large Entity Fee ($)	Small Entity Fee Code	Small Entity Fee ($)	Fee Description	Fee Paid
105	130	205	65	Surcharge - late filing fee or oath	$130
127	50	227	25	Surcharge - late provisional filing fee or cover sheet	
139	130	139	130	Non-English specification	
147	2,520	147	2,520	For filing a request for *ex parte* reexamination	
112	920*	112	920*	Requesting publication of SIR prior to Examiner action	
113	1,840*	113	1,840*	Requesting publication of SIR after Examiner action	
115	110	215	55	Extension for reply within first month	
116	400	216	200	Extension for reply within second month	
117	920	217	460	Extension for reply within third month	
118	1,440	218	720	Extension for reply within fourth month	
128	1,960	228	980	Extension for reply within fifth month	
119	320	219	160	Notice of Appeal	
120	320	220	160	Filing a brief in support of an appeal	
121	280	221	140	Request for oral hearing	
138	1,510	138	1,510	Petition to institute a public use proceeding	
140	110	240	55	Petition to revive - unavoidable	
141	1,280	241	640	Petition to revive - unintentional	
142	1,280	242	640	Utility issue fee (or reissue)	
143	460	243	230	Design issue fee	
144	620	244	310	Plant issue fee	
122	130	122	130	Petitions to the Commissioner	
123	50	123	50	Processing fee under 37 CFR 1.17(q)	
126	180	126	180	Submission of Information Disclosure Stmt	
581	40	581	40	Recording each patent assignment per property (times number of properties)	
146	740	246	370	Filing a submission after final rejection (37 CFR § 1.129(a))	
149	740	249	370	For each additional invention to be examined (37 CFR § 1.129(b))	
179	740	279	370	Request for Continued Examination (RCE)	
169	900	169	900	Request for expedited examination of a design application	

Other fee (specify) _____

*Reduced by Basic Filing Fee Paid **SUBTOTAL (3)** ($) 130.00

SUBMITTED BY				Complete *(if applicable)*	
Name (Print/Type)	Richard L. Gregory, Jr.	Registration No. (Attorney/Agent)	42,607	Telephone	408-236-6641
Signature				Date	08-01-2002

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.

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending upon the needs of the individual case. Any comments on the amount of time you are required to complete this form should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, Washington, DC 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Assistant Commissioner for Patents, Washington, DC 20231.

UNITED STATES PATENT AND TRADEMARK OFFICE

COMMISSIONER FOR PATENTS
UNITED STATES PATENT AND TRADEMARK OFFICE
WASHINGTON, D.C. 20231
www.uspto.gov

APPLICATION NUMBER	FILING/RECEIPT DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NUMBER
10/159,770	05/30/2002	Gregory C. Burnett	37212.8009.US00

CONFIRMATION NO. 9714

22918
PERKINS COIE LLP
P.O. BOX 2168
MENLO PARK, CA 94026

FORMALITIES LETTER

OC000000008443147

Date Mailed: 07/12/2002

NOTICE TO FILE MISSING PARTS OF NONPROVISIONAL APPLICATION

FILED UNDER 37 CFR 1.53(b)

Filing Date Granted

Items Required To Avoid Abandonment:

An application number and filing date have been accorded to this 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 oath or declaration is unsigned.
- To avoid abandonment, a late filing fee or oath or declaration surcharge as set forth in 37 CFR 1.16(l) of $130 for a non-small entity, must be submitted with the missing items identified in this letter.

The application is informal since it does not comply with the regulations for the reason(s) indicated below.
The required item(s) identified below must be timely submitted to avoid abandonment:

- Replacement drawings in compliance with 37 CFR 1.84 and 37 CFR 1.121 are required. The drawings submitted are not acceptable because:
  - The drawings must be made on paper that has a white background, (see 37 CFR 1.84(e)). For example. Graph paper, lined paper, or paper that has a non-white background is not acceptable.

Items Required To Avoid Processing Delays:
The item(s) indicated below are also required and should be submitted with any reply to this notice to avoid further processing delays.

SUMMARY OF FEES DUE:
Total additional fee(s) required for this application is **$130** for a Large Entity

- **$130** Late oath or declaration Surcharge.

A copy of this notice MUST be returned with the reply.

OIPE

Customer Service Center
Initial Patent Examination Division (703) 308-1202
PART 1 - ATTORNEY/APPLICANT COPY

06-03-02

A

Attorney Docket No. 37212.8009.US00

jc682 U.S. PTO 05/30/02

J1040 U.S. PTO 10/159770 05/30/02

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Certification under 37 CFR §1.10 (*if applicable*)

EV 057 398 806 US	May 30, 2002
Express Mail Label Number	Date of Deposit

I hereby certify that this Transmittal Letter, enclosed application and any other documents referred to as enclosed herein, are being deposited in an envelope with the United States Postal Service "Express Mail Post Office to Addressee" service under 37 CFR §1.10 on the date above and addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231.

Jennifer L. Mahoney
(Print Name of Person Mailing Application)

Jennifer Mahoney
(Signature of Person Mailing Application)

Transmittal of Utility Patent Application for Filing Under 37 CFR §1.53(b)

Box Patent Application
Assistant Commissioner for Patents
Washington, D.C. 20231

Sir: Transmitted herewith for filing is a patent application by inventor Gregory C. Burnett and entitled:

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

1. Enclosed are:
   - ☒ This Transmittal letter.
   - ☒ One stamped, self-addressed postcard for PTO datestamp.
   - ☒ Certificate of Express Mail.
   - ☒ One utility patent application containing text pages 1-19 and
     - ☒ 9 Sheets of drawings.
   - ☒ Declaration of Inventorship (unsigned).

2. U.S. Priority
   - ☒ This application claims the benefit of U.S. Application Nos. 60/294,383 filed May 30, 2001; 09/905,361 filed July 12, 2001; 60/335,100 filed October 30, 2001; 60/332,202 and 09/990,847, both filed November 21, 2001; 60/362,162, 60/362,103, 60/362,170, 60/361,981, and 60/362,161, all filed March 5, 2002; 60/368,209, 60/368,208, and 60/368,343, all filed March 27, 2002; all of which are incorporated herein by reference in their entirety.
   - ☒ Conditional Petition for Extension of Time: An Extension of Time is requested to provide for timely filing *if* required to establish copendency with the parent after all papers filed herewith have been considered.

3. Foreign Priority
   - ☐ Priority of Application No. filed in on is claimed under 35 USC §119.
   - ☐ A certified copy of this priority document is enclosed.

4. Fees
The filing fee has been calculated as shown below:

For:	(Col. 1) No. Filed	(Col. 2) No. Extra	Small Entity			Other Than a Small Entity	
			Rate	Fee		Rate	Fee
Basic Fee				$370.00	or		$740.00
Total Claims	7 - 20	0	x $ 9 =	$	or	x $ 18 =	$
Independent Claims	4 - 3	1	x $42 =	$	or	x $ 84 =	$84.00
☐ Multiple Dependent Claim Presented			+ $140 =	$	or	+ $280 =	$
*If the difference in Col. 1 is less than zero, enter "0" in Col. 2.			TOTAL	$	or	TOTAL	$824.00

☐ Applicant claims small entity status. See 37 CFR §1.27.

☒ Please charge Deposit Account No. 50-2207 in the amount of $824.00.

☒ The Commissioner is hereby authorized to charge fees under 37 CFR §1.16 and §1.17 which may be required, or credit any overpayment to Deposit Account No. 50-2207.

Respectfully submitted,
Perkins Coie LLP

Date: 5-30-2002

R. L. Gregory

Richard L. Gregory, Jr.
Registration No. 42,607

Correspondence Address:
Customer No. 22918
Perkins Coie LLP
101 Jefferson Drive
Menlo Park, California 94025-1114
(650) 838-4300

20050602.0245 PCT

06-03-02

A

Attorney Docket No. 37212.8009.US00

jc682 U.S. PTO 05/30/02

J1040 U.S. PTO 10/159770 05/30/02

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Certification under 37 CFR §1.10 (*if applicable*)

EV 057 398 806 US	May 30, 2002
Express Mail Label Number	Date of Deposit

I hereby certify that this Transmittal Letter, enclosed application and any other documents referred to as enclosed herein, are being deposited in an envelope with the United States Postal Service "Express Mail Post Office to Addressee" service under 37 CFR §1.10 on the date above and addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231.

Jennifer L. Mahoney
(Print Name of Person Mailing Application)

Jennifer Mahoney
(Signature of Person Mailing Application)

Transmittal of Utility Patent Application for Filing Under 37 CFR §1.53(b)

Box Patent Application
Assistant Commissioner for Patents
Washington, D.C. 20231

Sir: Transmitted herewith for filing is a patent application by inventor Gregory C. Burnett and entitled:

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

1. Enclosed are:
   - ☒ This Transmittal letter.
   - ☒ One stamped, self-addressed postcard for PTO datestamp.
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   - ☒ Conditional Petition for Extension of Time: An Extension of Time is requested to provide for timely filing *if* required to establish copendency with the parent after all papers filed herewith have been considered.

3. Foreign Priority
   - ☐ Priority of Application No. filed in on is claimed under 35 USC §119.
   - ☐ A certified copy of this priority document is enclosed.

4. Fees
The filing fee has been calculated as shown below:

For:	(Col. 1) No. Filed	(Col. 2) No. Extra	Small Entity			Other Than a Small Entity	
			Rate	Fee		Rate	Fee
Basic Fee				$370.00	or		$740.00
Total Claims	7 - 20	0	x $ 9 =	$	or	x $ 18 =	$
Independent Claims	4 - 3	1	x $42 =	$	or	x $ 84 =	$84.00
☐ Multiple Dependent Claim Presented			+ $140 =	$	or	+ $280 =	$
*If the difference in Col. 1 is less than zero, enter "0" in Col. 2.			TOTAL	$	or	TOTAL	$824.00

☐ Applicant claims small entity status. See 37 CFR §1.27.

☒ Please charge Deposit Account No. 50-2207 in the amount of $824.00.

☒ The Commissioner is hereby authorized to charge fees under 37 CFR §1.16 and §1.17 which may be required, or credit any overpayment to Deposit Account No. 50-2207.

Respectfully submitted,
Perkins Coie LLP

Date: 5-30-2002

R. L. Gregory

Richard L. Gregory, Jr.
Registration No. 42,607

Correspondence Address:
Customer No. 22918
Perkins Coie LLP
101 Jefferson Drive
Menlo Park, California 94025-1114
(650) 838-4300

20050602.0457.pdf

Attorney Docket No. 37212.8009.US00 Patent

Transmittal of Utility Patent Application for Filing

Certification Under 37 C.F.R. §1.10 (if applicable)

EV 057 398 806 US
"Express Mail" Label Number

May 30, 2002
Date of Deposit

I hereby certify that this application, and any other documents referred to as enclosed herein are being deposited in an envelope with the United States Postal Service "Express Mail Post Office to Addressee" service under 37 CFR §1.10 on the date indicated above and addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231

Jennifer L. Mahoney
(Print Name of Person Mailing Application)

Jennifer Mahoney
(Signature of Person Mailing Application)

11040 U.S. PTO
10/159770
05/30/02

20050504.0756PCT

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

RELATED APPLICATIONS

This application claims the benefit of U.S. Application Numbers 60/294,383 filed May 30, 2001; 09/905,361 filed July 12, 2001; 60/335,100 filed October 30, 2001; 60/332,202 and 09/990,847, both filed November 21, 2001; 60/362,103, 60/362,161, 60/362,162, 60/362,170, and 60/361,981, all filed March 5, 2002; 60/368,208, 60/368,209, and 60/368,343, all filed March 27, 2002; all of which are incorporated herein by reference in their entirety.

TECHNICAL FIELD

The disclosed embodiments relate to the processing of speech signals.

BACKGROUND

The ability to correctly identify voiced and unvoiced speech is critical to many speech applications including speech recognition, speaker verification, noise suppression, and many others. In a typical acoustic application, speech from a human speaker is captured and transmitted to a receiver in a different location. In the speaker's environment there may exist one or more noise sources that pollute the speech signal, or the signal of interest, with unwanted acoustic noise. This makes it difficult or impossible for the receiver, whether human or machine, to understand the user's speech.

Typical methods for classifying voiced and unvoiced speech have relied mainly on the acoustic content of microphone data, which is plagued by problems with noise and the corresponding uncertainties in signal content. This is especially problematic now with the proliferation of portable communication devices like cellular telephones and personal digital assistants because, in many cases, the quality of service provided by the device depends on the quality of the voice services offered by the device. There are methods known in the art for suppressing the noise present in the speech signals, but these methods demonstrate performance shortcomings that include unusually long computing time, requirements for cumbersome hardware to perform the signal processing, and distorting the signals of interest.

BRIEF DESCRIPTION OF THE FIGURES

Figure 1 is a block diagram of a NAVSAD system, under an embodiment.

Figure 2 is a block diagram of a PSAD system, under an embodiment.

Figure 3 is a block diagram of a denoising system, referred to herein as the Pathfinder system, under an embodiment.

Figure 4 is a flow diagram of a detection algorithm for use in detecting voiced and unvoiced speech, under an embodiment.

Figure 5A plots the received GEMS signal for an utterance along with the mean correlation between the GEMS signal and the Mic 1 signal and the threshold for voiced speech detection.

Figure 5B plots the received GEMS signal for an utterance along with the standard deviation of the GEMS signal and the threshold for voiced speech detection.

Figure 6 plots voiced speech detected from an utterance along with the GEMS signal and the acoustic noise.

Figure 7 is a microphone array for use under an embodiment of the PSAD system.

Figure 8 is a plot of ΔM versus d_1 for several Δd values, under an embodiment.

Figure 9 shows a plot of the gain parameter as the sum of the absolute values of $H_1(z)$ and the acoustic data or audio from microphone 1.

Figure 10 is an alternative plot of acoustic data presented in Figure 9.

In the figures, the same reference numbers identify identical or substantially similar elements or acts.

Any headings provided herein are for convenience only and do not necessarily affect the scope or meaning of the claimed invention.

DETAILED DESCRIPTION

Systems and methods for discriminating voiced and unvoiced speech from background noise are provided below including a Non-Acoustic Sensor Voiced Speech Activity Detection (NAVSAD) system and a Pathfinder Speech Activity Detection (PSAD) system. The noise removal and reduction methods provided herein, while allowing for the separation and classification of unvoiced and voiced human speech from background noise, address the shortcomings of typical systems known in the art by cleaning acoustic signals of interest without distortion.

Figure 1 is a block diagram of a NAVSAD system 100, under an embodiment. The NAVSAD system couples microphones 10 and sensors 20 to at least one processor 30. The sensors 20 of an embodiment include voicing activity detectors or non-acoustic sensors. The processor 30 controls subsystems including a detection subsystem 50, referred to herein as a detection algorithm, and a denoising subsystem 40. Operation of the denoising subsystem 40 is described in detail in the Related Applications. The NAVSAD system works extremely well in any background acoustic noise environment.

Figure 2 is a block diagram of a PSAD system 200, under an embodiment. The PSAD system couples microphones 10 to at least one processor 30. The processor 30 includes a detection subsystem 50, referred to herein as a detection algorithm, and a denoising subsystem 40. The PSAD system is highly sensitive in low acoustic noise environments and relatively insensitive in high acoustic noise environments. The PSAD can operate independently or as a backup to the NAVSAD, detecting voiced speech if the NAVSAD fails.

Note that the detection subsystems 50 and denoising subsystems 40 of both the NAVSAD and PSAD systems of an embodiment are algorithms controlled by the processor 30, but are not so limited. Alternative embodiments of the NAVSAD and PSAD systems can include detection subsystems 50 and/or denoising subsystems 40 that comprise additional hardware, firmware, software, and/or combinations of hardware, firmware, and software. Furthermore, functions

of the detection subsystems 50 and denoising subsystems 40 may be distributed across numerous components of the NAVSAD and PSAD systems.

Figure 3 is a block diagram of a denoising subsystem 300, referred to herein as the Pathfinder system, under an embodiment. The Pathfinder system is briefly described below, and is described in detail in the Related Applications. Two microphones Mic 1 and Mic 2 are used in the Pathfinder system, and Mic 1 is considered the "signal" microphone. With reference to **Figure 1**, the Pathfinder system 300 is equivalent to the NAVSAD system 100 when the voicing activity detector (VAD) 320 is a non-acoustic voicing sensor 20 and the noise removal subsystem 340 includes the detection subsystem 50 and the denoising subsystem 40. With reference to **Figure 2**, the Pathfinder system 300 is equivalent to the PSAD system 200 in the absence of the VAD 320, and when the noise removal subsystem 340 includes the detection subsystem 50 and the denoising subsystem 40.

The NAVSAD and PSAD systems support a two-level commercial approach in which (i) a relatively less expensive PSAD system supports an acoustic approach that functions in most low- to medium-noise environments, and (ii) a NAVSAD system adds a non-acoustic sensor to enable detection of voiced speech in any environment. Unvoiced speech is normally not detected using the sensor, as it normally does not sufficiently vibrate human tissue. However, in high noise situations detecting the unvoiced speech is not as important, as it is normally very low in energy and easily washed out by the noise. Therefore in high noise environments the unvoiced speech is unlikely to affect the voiced speech denoising. Unvoiced speech information is most important in the presence of little to no noise and, therefore, the unvoiced detection should be highly sensitive in low noise situations, and insensitive in high noise situations. This is not easily accomplished, and comparable acoustic unvoiced detectors known in the art are incapable of operating under these environmental constraints.

The NAVSAD and PSAD systems include an array algorithm for speech detection that uses the difference in frequency content between two microphones to calculate a relationship between the signals of the two microphones. This is in contrast to conventional arrays that attempt to use the time/phase difference of each microphone to remove the noise outside of an "area of sensitivity". The

methods described herein provide a significant advantage, as they do not require a specific orientation of the array with respect to the signal.

Further, the systems described herein are sensitive to noise of every type and every orientation, unlike conventional arrays that depend on specific noise orientations. Consequently, the frequency-based arrays presented herein are unique as they depend only on the relative orientation of the two microphones themselves with no dependence on the orientation of the noise and signal with respect to the microphones. This results in a robust signal processing system with respect to the type of noise, microphones, and orientation between the noise/signal source and the microphones.

The systems described herein use the information derived from the Pathfinder noise suppression system and/or a non-acoustic sensor described in the Related Applications to determine the voicing state of an input signal, as described in detail below. The voicing state includes silent, voiced, and unvoiced states. The NAVSAD system, for example, includes a non-acoustic sensor to detect the vibration of human tissue associated with speech. The non-acoustic sensor of an embodiment is a General Electromagnetic Movement Sensor (GEMS) as described briefly below and in detail in the Related Applications, but is not so limited. Alternative embodiments, however, may use any sensor that is able to detect human tissue motion associated with speech and is unaffected by environmental acoustic noise.

The GEMS is a radio frequency device (2.4 GHz) that allows the detection of moving human tissue dielectric interfaces. The GEMS includes an RF interferometer that uses homodyne mixing to detect small phase shifts associated with target motion. In essence, the sensor sends out weak electromagnetic waves (less than 1 milliwatt) that reflect off of whatever is around the sensor. The reflected waves are mixed with the original transmitted waves and the results analyzed for any change in position of the targets. Anything that moves near the sensor will cause a change in phase of the reflected wave that will be amplified and displayed as a change in voltage output from the sensor. A similar sensor is described by Gregory C. Burnett (1999) in "The physiological basis of glottal electromagnetic micropower sensors (GEMS) and their use in defining an excitation function for the human vocal tract"; Ph.D. Thesis, University of California at Davis.

Figure 4 is a flow diagram of a detection algorithm 50 for use in detecting voiced and unvoiced speech, under an embodiment. With reference to **Figures 1 and 2**, both the NAVSAD and PSAD systems of an embodiment include the detection algorithm 50 as the detection subsystem 50. This detection algorithm 50 operates in real-time and, in an embodiment, operates on 20 millisecond windows and steps 10 milliseconds at a time, but is not so limited. The voice activity determination is recorded for the first 10 milliseconds, and the second 10 milliseconds functions as a "look-ahead" buffer. While an embodiment uses the 20/10 windows, alternative embodiments may use numerous other combinations of window values.

Consideration was given to a number of multi-dimensional factors in developing the detection algorithm 50. The biggest consideration was to maintaining the effectiveness of the Pathfinder denoising technique, described in detail in the Related Applications and reviewed herein. Pathfinder performance can be compromised if the adaptive filter training is conducted on speech rather than on noise. It is therefore important not to exclude any significant amount of speech from the VAD to keep such disturbances to a minimum.

Consideration was also given to the accuracy of the characterization between voiced and unvoiced speech signals, and distinguishing each of these speech signals from noise signals. This type of characterization can be useful in such applications as speech recognition and speaker verification.

Furthermore, the systems using the detection algorithm of an embodiment function in environments containing varying amounts of background acoustic noise. If the non-acoustic sensor is available, this external noise is not a problem for voiced speech. However, for unvoiced speech (and voiced if the non-acoustic sensor is not available or has malfunctioned) reliance is placed on acoustic data alone to separate noise from unvoiced speech. An advantage inheres in the use of two microphones in an embodiment of the Pathfinder noise suppression system, and the spatial relationship between the microphones is exploited to assist in the detection of unvoiced speech. However, there may occasionally be noise levels high enough that the speech will be nearly undetectable and the acoustic-only method will fail. In these situations, the non-acoustic sensor (or hereafter just the sensor) will be required to ensure good performance.

In the two-microphone system, the speech source should be relatively louder in one designated microphone when compared to the other microphone. Tests have shown that this requirement is easily met with conventional microphones when the microphones are placed on the head, as any noise should result in an H_1 with a gain near unity.

Regarding the NAVSAD system, and with reference to **Figure 1 and Figure 3**, the NAVSAD relies on two parameters to detect voiced speech. These two parameters include the energy of the sensor in the window of interest, determined in an embodiment by the standard deviation (SD), and optionally the cross-correlation (XCORR) between the acoustic signal from microphone 1 and the sensor data. The energy of the sensor can be determined in any one of a number of ways, and the SD is just one convenient way to determine the energy.

For the sensor, the SD is akin to the energy of the signal, which normally corresponds quite accurately to the voicing state, but may be susceptible to movement noise (relative motion of the sensor with respect to the human user) and/or electromagnetic noise. To further differentiate sensor noise from tissue motion, the XCORR can be used. The XCORR is only calculated to 15 delays, which corresponds to just under 2 milliseconds at 8000 Hz.

The XCORR can also be useful when the sensor signal is distorted or modulated in some fashion. For example, there are sensor locations (such as the jaw or back of the neck) where speech production can be detected but where the signal may have incorrect or distorted time-based information. That is, they may not have well defined features in time that will match with the acoustic waveform. However, XCORR is more susceptible to errors from acoustic noise, and in high (<0 dB SNR) environments is almost useless. Therefore it should not be the sole source of voicing information.

The sensor detects human tissue motion associated with the closure of the vocal folds, so the acoustic signal produced by the closure of the folds is highly correlated with the closures. Therefore, sensor data that correlates highly with the acoustic signal is declared as speech, and sensor data that does not correlate well is termed noise. The acoustic data is expected to lag behind the sensor data by about 0.1 to 0.8 milliseconds (or about 1-7 samples) as a result of the delay time due to the relatively slower speed of sound (around 330 m/s). However, an embodiment uses a 15-sample correlation, as the acoustic wave shape varies

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significantly depending on the sound produced, and a larger correlation width is needed to ensure detection.

The SD and XCORR signals are related, but are sufficiently different so that the voiced speech detection is more reliable. For simplicity, though, either parameter may be used. The values for the SD and XCORR are compared to empirical thresholds, and if both are above their threshold, voiced speech is declared. Example data is presented and described below.

Figures 5A, 5B, and 6 show data plots for an example in which a subject twice speaks the phrase "pop pan", under an embodiment. **Figure 5A** plots the received GEMS signal 502 for this utterance along with the mean correlation 504 between the GEMS signal and the Mic 1 signal and the threshold T1 used for voiced speech detection. **Figure 5B** plots the received GEMS signal 502 for this utterance along with the standard deviation 506 of the GEMS signal and the threshold T2 used for voiced speech detection. **Figure 6** plots voiced speech 602 detected from the acoustic or audio signal 608, along with the GEMS signal 604 and the acoustic noise 606; no unvoiced speech is detected in this example because of the heavy background babble noise 606. The thresholds have been set so that there are virtually no false negatives, and only occasional false positives. A voiced speech activity detection accuracy of greater than 99% has been attained under any acoustic background noise conditions.

The NAVSAD can determine when voiced speech is occurring with high degrees of accuracy due to the non-acoustic sensor data. However, the sensor offers little assistance in separating unvoiced speech from noise, as unvoiced speech normally causes no detectable signal in most non-acoustic sensors. If there is a detectable signal, the NAVSAD can be used, although use of the SD method is dictated as unvoiced speech is normally poorly correlated. In the absence of a detectable signal use is made of the system and methods of the Pathfinder noise removal algorithm in determining when unvoiced speech is occurring. A brief review of the Pathfinder algorithm is described below, while a detailed description is provided in the Related Applications.

With reference to **Figure 3**, the acoustic information coming into Microphone 1 is denoted by $m_1(n)$, the information coming into Microphone 2 is similarly labeled $m_2(n)$, and the GEMS sensor is assumed available to determine

voiced speech areas. In the *z* (digital frequency) domain, these signals are represented as $M_1(z)$ and $M_2(z)$. Then

$$M_1(z) = S(z) + N_2(z)$$
$$M_2(z) = N(z) + S_2(z)$$

with

$$N_2(z) = N(z)H_1(z)$$
$$S_2(z) = S(z)H_2(z)$$

so that

$$M_1(z) = S(z) + N(z)H_1(z)$$
$$M_2(z) = N(z) + S(z)H_2(z) \quad (1)$$

This is the general case for all two microphone systems. There is always going to be some leakage of noise into Mic 1, and some leakage of signal into Mic 2. Equation 1 has four unknowns and only two relationships and cannot be solved explicitly.

However, there is another way to solve for some of the unknowns in Equation 1. Examine the case where the signal is not being generated – that is, where the GEMS signal indicates voicing is not occurring. In this case, s(n) = S(z) = 0, and Equation 1 reduces to

$$M_{1n}(z) = N(z)H_1(z)$$
$$M_{2n}(z) = N(z)$$

where the *n* subscript on the M variables indicate that only noise is being received. This leads to

$$M_{1n}(z) = M_{2n}(z)H_1(z)$$
$$H_1(z) = \frac{M_{1n}(z)}{M_{2n}(z)} \quad (2)$$

$H_1(z)$ can be calculated using any of the available system identification algorithms and the microphone outputs when only noise is being received. The calculation can be done adaptively, so that if the noise changes significantly $H_1(z)$ can be recalculated quickly.

With a solution for one of the unknowns in Equation 1, solutions can be found for another, $H_2(z)$, by using the amplitude of the GEMS or similar device along with the amplitude of the two microphones. When the GEMS indicates

voicing, but the recent (less than 1 second) history of the microphones indicate low levels of noise, assume that n(s) = N(z) ~ 0. Then Equation 1 reduces to

$$M_{1s}(z) = S(z)$$
$$M_{2s}(z) = S(z)H_2(z)$$

which in turn leads to

$$M_{2s}(z) = M_{1s}(z)H_2(z)$$
$$H_2(z) = \frac{M_{2s}(z)}{M_{1s}(z)}$$

which is the inverse of the $H_1(z)$ calculation, but note that different inputs are being used.

After calculating $H_1(z)$ and $H_2(z)$ above, they are used to remove the noise from the signal. Rewrite Equation 1 as

$$S(z) = M_1(z) - N(z)H_1(z)$$
$$N(z) = M_2(z) - S(z)H_2(z)$$
$$S(z) = M_1(z) - [M_2(z) - S(z)H_2(z)]H_1(z)$$
$$S(z)[1 - H_2(z)H_1(z)] = M_1(z) - M_2(z)H_1(z)$$

and solve for S(z) as:

$$S(z) = \frac{M_1(z) - M_2(z)H_1(z)}{1 - H_2(z)H_1(z)}. \qquad (3)$$

In practice $H_2(z)$ is usually quite small, so that $H_2(z)H_1(z) << 1$, and

$$S(z) \approx M_1(z) - M_2(z)H_1(z),$$

obviating the need for the $H_2(z)$ calculation.

With reference to **Figure 2** and **Figure 3**, the PSAD system is described. As sound waves propagate, they normally lose energy as they travel due to diffraction and dispersion. Assuming the sound waves originate from a point source and radiate isotropically, their amplitude will decrease as a function of 1/r, where r is the distance from the originating point. This function of 1/r proportional to amplitude is the worst case, if confined to a smaller area the reduction will be less. However it is an adequate model for the configurations of interest, specifically the propagation of noise and speech to microphones located somewhere on the user's head.

Figure 7 is a microphone array for use under an embodiment of the PSAD system. Placing the microphones Mic 1 and Mic 2 in a linear array with the mouth on the array midline, the difference in signal strength in Mic 1 and Mic 2 (assuming the microphones have identical frequency responses) will be proportional to both d_1 and Δd. Assuming a 1/r (or in this case 1/d) relationship, it is seen that

$$\Delta M = \frac{|Mic1|}{|Mic2|} = \Delta H_1(z) \propto \frac{d_1 + \Delta d}{d_1},$$

where ΔM is the difference in gain between Mic 1 and Mic 2 and therefore $H_1(z)$, as above in Equation 2. The variable d_1 is the distance from Mic 1 to the speech or noise source.

Figure 8 is a plot 800 of ΔM versus d_1 for several Δd values, under an embodiment. It is clear that as Δd becomes larger and the noise source is closer, ΔM becomes larger. The variable Δd will change depending on the orientation to the speech/noise source, from the maximum value on the array midline to zero perpendicular to the array midline. From the plot 800 it is clear that for small Δd and for distances over approximately 30 centimeters (cm), ΔM is close to unity. Since most noise sources are farther away than 30 cm and are unlikely to be on the midline on the array, it is probable that when calculating $H_1(z)$ as above in Equation 2, ΔM (or equivalently the gain of $H_1(z)$) will be close to unity. Conversely, for noise sources that are close (within a few centimeters), there could be a substantial difference in gain depending on which microphone is closer to the noise.

If the "noise" is the user speaking, and Mic 1 is closer to the mouth than Mic 2, the gain increases. Since environmental noise normally originates much farther away from the user's head than speech, noise will be found during the time when the gain of $H_1(z)$ is near unity or some fixed value, and speech can be found after a sharp rise in gain. The speech can be unvoiced or voiced, as long as it is of sufficient volume compared to the surrounding noise. The gain will stay somewhat high during the speech portions, then descend quickly after speech ceases. The rapid increase and decrease in the gain of $H_1(z)$ should be sufficient to allow the detection of speech under almost any circumstances. The gain in this example is calculated by the sum of the absolute value of the filter coefficients. This sum is

not equivalent to the gain, but the two are related in that a rise in the sum of the absolute value reflects a rise in the gain.

As an example of this behavior, **Figure 9** shows a plot 900 of the gain parameter 902 as the sum of the absolute values of $H_1(z)$ and the acoustic data 904 or audio from microphone 1. The speech signal was an utterance of the phrase "pop pan", repeated twice. The evaluated bandwidth included the frequency range from 2500 Hz to 3500 Hz, although 1500Hz to 2500 Hz was additionally used in practice. Note the rapid increase in the gain when the unvoiced speech is first encountered, then the rapid return to normal when the speech ends. The large changes in gain that result from transitions between noise and speech can be detected by any standard signal processing techniques. The standard deviation of the last few gain calculations is used, with thresholds being defined by a running average of the standard deviations and the standard deviation noise floor. The later changes in gain for the voiced speech are suppressed in this plot 900 for clarity.

Figure 10 is an alternative plot 1000 of acoustic data presented in Figure 9. The data used to form plot 900 is presented again in this plot 1000, along with audio data 1004 and GEMS data 1006 without noise to make the unvoiced speech apparent. The voiced signal 1002 has three possible values: 0 for noise, 1 for unvoiced, and 2 for voiced. Denoising is only accomplished when V = 0. It is clear that the unvoiced speech is captured very well, aside from two single dropouts in the unvoiced detection near the end of each "pop". However, these single-window dropouts are not common and do not significantly affect the denoising algorithm. They can easily be removed using standard smoothing techniques.

What is not clear from this plot 1000 is that the PSAD system functions as an automatic backup to the NAVSAD. This is because the voiced speech (since it has the same spatial relationship to the mics as the unvoiced) will be detected as unvoiced if the sensor or NAVSAD system fail for any reason. The voiced speech will be misclassified as unvoiced, but the denoising will still not take place, preserving the quality of the speech signal.

However, this automatic backup of the NAVSAD system functions best in an environment with low noise (approximately 10+ dB SNR), as high amounts (10 dB of SNR or less) of acoustic noise can quickly overwhelm any acoustic-only unvoiced detector, including the PSAD. This is evident in the difference in the

voiced signal data 602 and 1002 shown in plots 600 and 100 of **Figures 6 and 10**, respectively, where the same utterance is spoken, but the data of plot 600 shows no unvoiced speech because the unvoiced speech is undetectable. *This is the* desired behavior when performing denoising, since if the unvoiced speech is not detectable then it will not significantly affect the denoising process. Using the Pathfinder system to detect unvoiced speech ensures detection of any unvoiced speech loud enough to distort the denoising.

Regarding hardware considerations, and with reference to **Figure 7**, the configuration of the microphones can have an effect on the change in gain associated with speech and the thresholds needed to detect speech. In general, each configuration will require testing to determine the proper thresholds, but tests with two very different microphone configurations showed the same thresholds and other parameters to work well. The first microphone set had the signal microphone near the mouth and the noise microphone several centimeters away at the ear, while the second configuration placed the noise and signal microphones back-to-back within a few centimeters of the mouth. The results presented herein were derived using the first microphone configuration, but the results using the other set are virtually identical, so the detection algorithm is relatively robust with respect to microphone placement.

A number of configurations are possible using the NAVSAD and PSAD systems to detect voiced and unvoiced speech. One configuration uses the NAVSAD system (non-acoustic only) to detect voiced speech along with the PSAD system to detect unvoiced speech; the PSAD also functions as a backup to the NAVSAD system for detecting voiced speech. An alternative configuration uses the NAVSAD system (non-acoustic correlated with acoustic) to detect voiced speech along with the PSAD system to detect unvoiced speech; the PSAD also functions as a backup to the NAVSAD system for detecting voiced speech. Another alternative configuration uses the PSAD system to detect both voiced and unvoiced speech.

While the systems described above have been described with reference to separating voiced and unvoiced speech from background acoustic noise, there are no reasons more complex classifications can not be made. For more in-depth characterization of speech, the system can bandpass the information from Mic 1 and Mic 2 so that it is possible to see which bands in the Mic 1 data are more

heavily composed of noise and which are more weighted with speech. Using this knowledge, it is possible to group the utterances by their spectral characteristics similar to conventional acoustic methods; this method would work better in noisy environments.

As an example, the "k" in "kick" has significant frequency content form 500 Hz to 4000 Hz, but a "sh" in "she" only contains significant energy from 1700-4000 Hz. Voiced speech could be classified in a similar manner. For instance, an /i/ ("ee") has significant energy around 300 Hz and 2500 Hz, and an /a/ ("ah") has energy at around 900 Hz and 1200 Hz. This ability to discriminate unvoiced and voiced speech in the presence of noise is, thus, very useful.

Each of the steps depicted in the flow diagrams presented herein can itself include a sequence of operations that need not be described herein. Those skilled in the relevant art can create routines, algorithms, source code, microcode, program logic arrays or otherwise implement the invention based on the flow diagrams and the detailed description provided herein. The routines described herein can be provided with one or more of the following, or one or more combinations of the following: stored in non-volatile memory (not shown) that forms part of an associated processor or processors, or implemented using conventional programmed logic arrays or circuit elements, or stored in removable media such as disks, or downloaded from a server and stored locally at a client, or hardwired or preprogrammed in chips such as EEPROM semiconductor chips, application specific integrated circuits (ASICs), or by digital signal processing (DSP) integrated circuits.

Unless described otherwise herein, the information described herein is well known or described in detail in the Related Applications. Indeed, much of the detailed description provided herein is explicitly disclosed in the Related Applications; most or all of the additional material of aspects of the invention will be recognized by those skilled in the relevant art as being inherent in the detailed description provided in such Related Applications, or well known to those skilled in the relevant art. Those skilled in the relevant art can implement aspects of the invention based on the material presented herein and the detailed description provided in the Related Applications.

Unless the context clearly requires otherwise, throughout the description and the claims, the words "comprise," "comprising," and the like are to be

construed in an inclusive sense as opposed to an exclusive or exhaustive sense; that is to say, in a sense of "including, but not limited to." Words using the singular or plural number also include the plural or singular number respectively. Additionally, the words "herein," "hereunder," and words of similar import, when used in this application, shall refer to this application as a whole and not to any particular portions of this application.

The above description of illustrated embodiments of the invention is not intended to be exhaustive or to limit the invention to the precise form disclosed. While specific embodiments of, and examples for, the invention are described herein for illustrative purposes, various equivalent modifications are possible within the scope of the invention, as those skilled in the relevant art will recognize. The teachings of the invention provided herein can be applied to signal processing systems, not only for the speech signal processing described above. Further, the elements and acts of the various embodiments described above can be combined to provide further embodiments.

All of the above references and Related Applications are incorporated herein by reference. Aspects of the invention can be modified, if necessary, to employ the systems, functions and concepts of the various references described above to provide yet further embodiments of the invention.

These and other changes can be made to the invention in light of the above detailed description. In general, in the following claims, the terms used should not be construed to limit the invention to the specific embodiments disclosed in the specification and the claims, but should be construed to include all speech signal systems that operate under the claims to provide a method for procurement. Accordingly, the invention is not limited by the disclosure, but instead the scope of the invention is to be determined entirely by the claims.

While certain aspects of the invention are presented below in certain claim forms, the inventor contemplates the various aspects of the invention in any number of claim forms. Thus, the inventor reserves the right to add additional claims after filing the application to pursue such additional claim forms for other aspects of the invention.

CLAIMS

What I claim is:

1. A system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

at least two microphones for receiving the acoustic signals;

at least one processor coupled among the microphones, wherein the at least one processor;

generates difference parameters between the acoustic signals received at each of the two microphones, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold; and

identifies information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold.

2. A method for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

receiving the acoustic signals at two receivers;

generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold; and

identifying information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold.

3. The method of claim 2, further comprising generating the first and second thresholds using standard deviations corresponding to the generation of the difference parameters.

4. The method of claim 2, further comprising:

identifying information of the acoustic signals as noise when the difference parameters are less than the first threshold; and

performing denoising on the identified noise.

5. The method of claim 2, further comprising receiving physiological information associated with human voicing activity, wherein the physiological information comprises receiving physiological data associated with human voicing using at least one detector selected from a group including radio frequency devices, electroglottographs, ultrasound devices, acoustic throat microphones, and airflow detectors.

6. A system for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise, comprising:

at least two microphones that receive the acoustic signals;

at least one voicing sensor that receives physiological information associated with human voicing activity; and

at least one processor coupled among the microphones and the voicing sensor, wherein the at least one processor;

generates cross correlation data between the physiological information and an acoustic signal received at one of the two microphones;

identifies information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generates difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifies information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifies information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

7. A method for removing noise from acoustic signals, comprising:

receiving the acoustic signals at two receivers and receiving physiological information associated with human voicing activity at a voicing sensor;

generating cross correlation data between the physiological information and an acoustic signal received at one of the two receivers;

identifying information of the acoustic signals as voiced speech when the cross correlation data corresponding to a portion of the acoustic signal received at the one receiver exceeds a correlation threshold;

generating difference parameters between the acoustic signals received at each of the two receivers, wherein the difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals;

identifying information of the acoustic signals as unvoiced speech when the difference parameters exceed a gain threshold; and

identifying information of the acoustic signals as noise when the difference parameters are less than the gain threshold.

Express Mail No. EV 339 920 715 US

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

ABSTRACT OF THE DISCLOSURE

Systems and methods are provided for detecting voiced and unvoiced speech in acoustic signals having varying levels of background noise. The systems receive acoustic signals at two microphones, and generate difference parameters between the acoustic signals received at each of the two microphones. The difference parameters are representative of the relative difference in signal gain between portions of the received acoustic signals. The systems identify information of the acoustic signals as unvoiced speech when the difference parameters exceed a first threshold, and identify information of the acoustic signals as voiced speech when the difference parameters exceed a second threshold. Further, embodiments of the systems include non-acoustic sensors that receive physiological information to aid in identifying voiced speech.

Express Mail Label No. EV 339 255 525 US

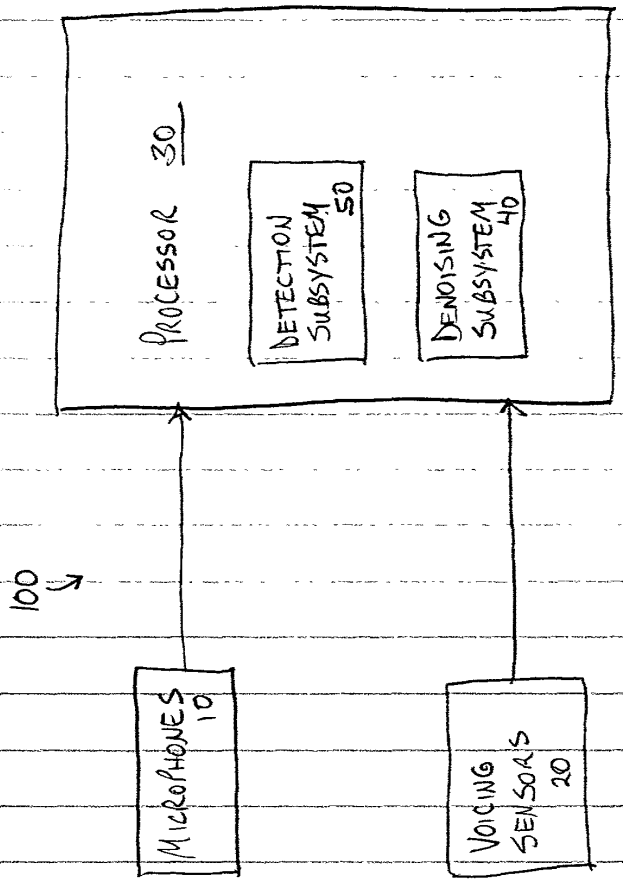

PROCESSOR 30
DETECTION SUBSYSTEM 50
DENOISING SUBSYSTEM 40
100
MICROPHONES 10
VOICING SENSORS 20

FIGURE 1

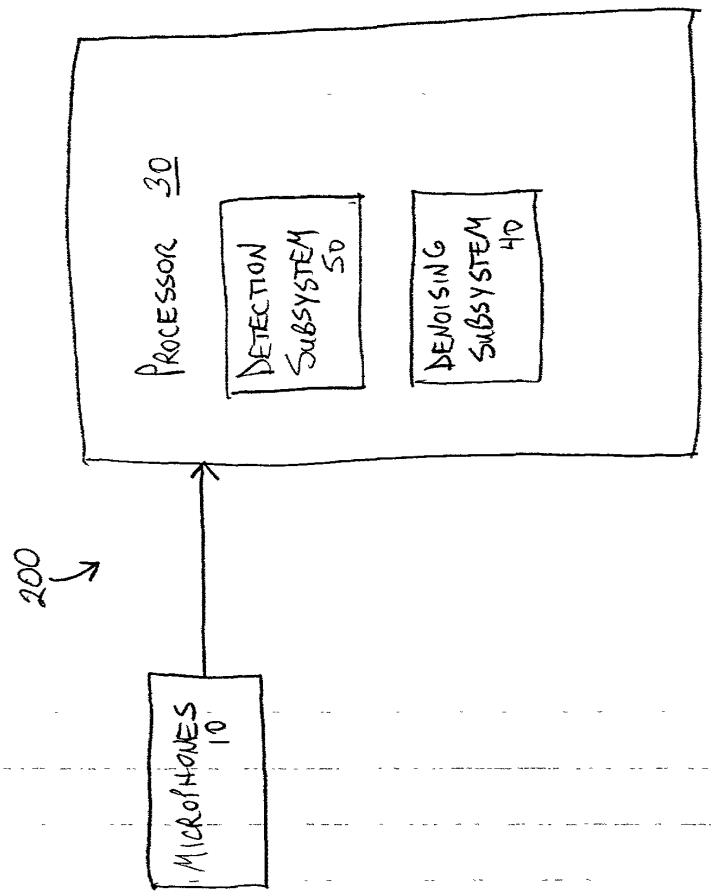

PROCESSOR 30
DETECTION SUBSYSTEM 50
DENOISING SUBSYSTEM 40
MICROPHONES 10
200

FIGURE 2

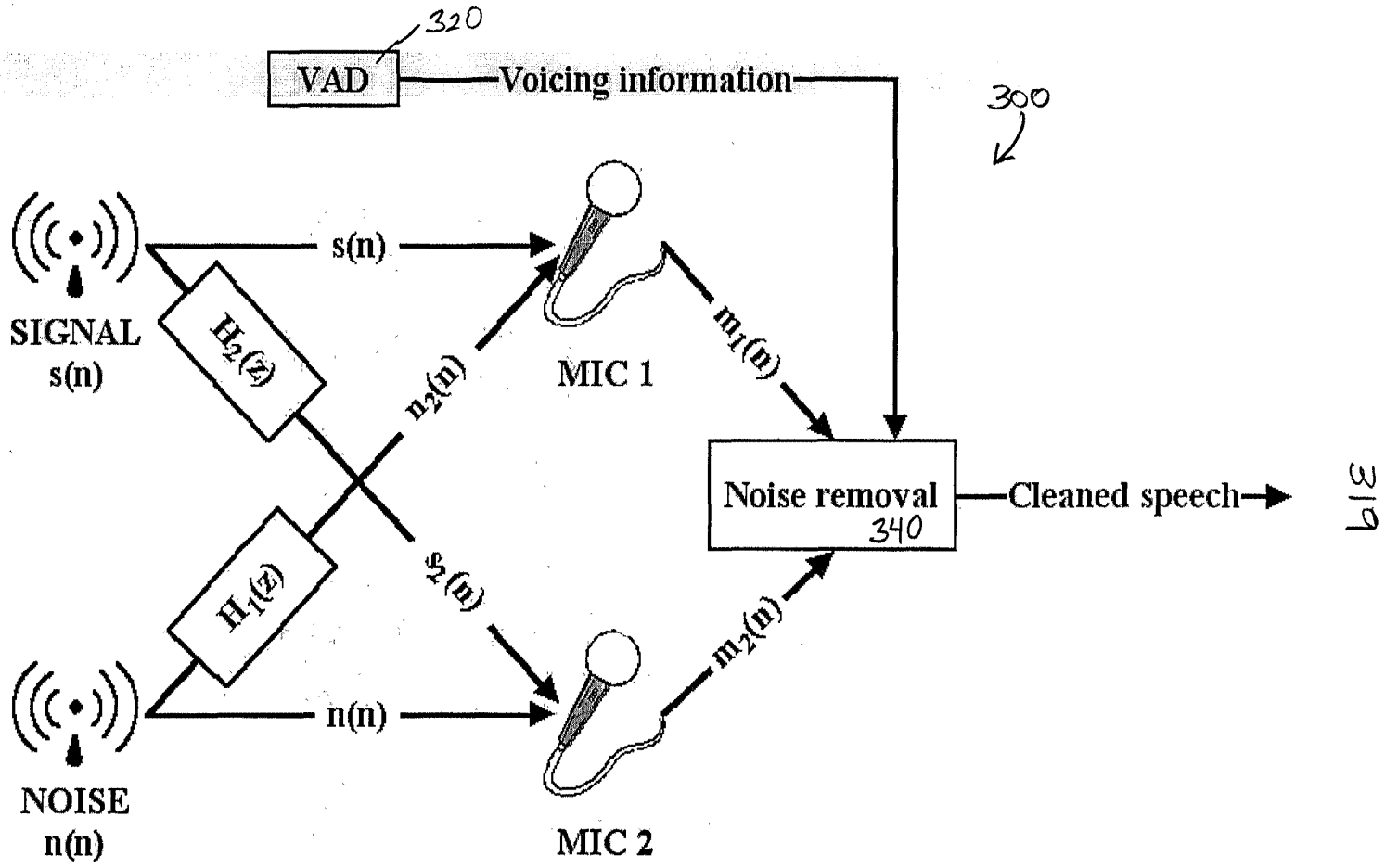

320
VAD
Voicing information
300
s(n)
SIGNAL
s(n)
$H_2(z)$
$n_2(n)$
MIC 1
$m_1(n)$
Noise removal
340
Cleaned speech
319
$H_1(z)$
$s_2(n)$
$m_2(n)$
n(n)
NOISE
n(n)
MIC 2

Figure 3

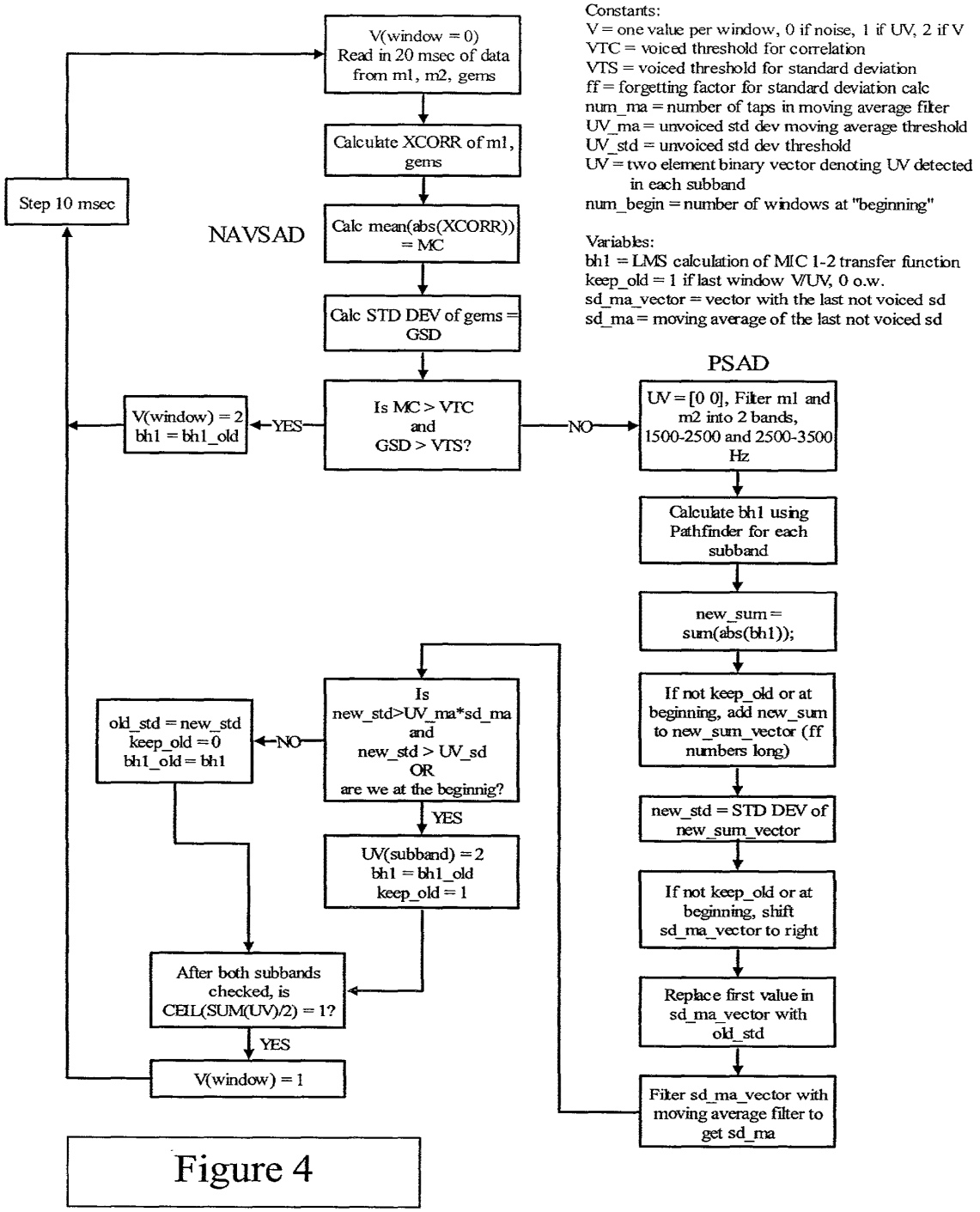

Constants:
V = one value per window, 0 if noise, 1 if UV, 2 if V
VTC = voiced threshold for correlation
VTS = voiced threshold for standard deviation
ff = forgetting factor for standard deviation calc
num_ma = number of taps in moving average filter
UV_ma = unvoiced std dev moving average threshold
UV_std = unvoiced std dev threshold
UV = two element binary vector denoting UV detected in each subband
num_begin = number of windows at "beginning"
Variables:
bh1 = LMS calculation of MIC 1-2 transfer function
keep_old = 1 if last window V/UV, 0 o.w.
sd_ma_vector = vector with the last not voiced sd
sd_ma = moving average of the last not voiced sd
NAVSAD
PSAD
V(window = 0)
Read in 20 msec of data from m1, m2, gems
Calculate XCORR of m1, gems
Calc mean(abs(XCORR)) = MC
Calc STD DEV of gems = GSD
Is MC > VTC and GSD > VTS?
YES
NO
V(window) = 2
bh1 = bh1_old
Step 10 msec
UV = [0 0], Filter m1 and m2 into 2 bands, 1500-2500 and 2500-3500 Hz
Calculate bh1 using Pathfinder for each subband
new_sum = sum(abs(bh1));
If not keep_old or at beginning, add new_sum to new_sum_vector (ff numbers long)
new_std = STD DEV of new_sum_vector
If not keep_old or at beginning, shift sd_ma_vector to right
Replace first value in sd_ma_vector with old_std
Filter sd_ma_vector with moving average filter to get sd_ma
Is new_std>UV_ma*sd_ma and new_std > UV_sd OR are we at the beginnig?
NO
YES
old_std = new_std
keep_old = 0
bh1_old = bh1
UV(subband) = 2
bh1 = bh1_old
keep_old = 1
After both subbands checked, is CEIL(SUM(UV)/2) = 1?
YES
V(window) = 1

Figure 4

Figure 5A

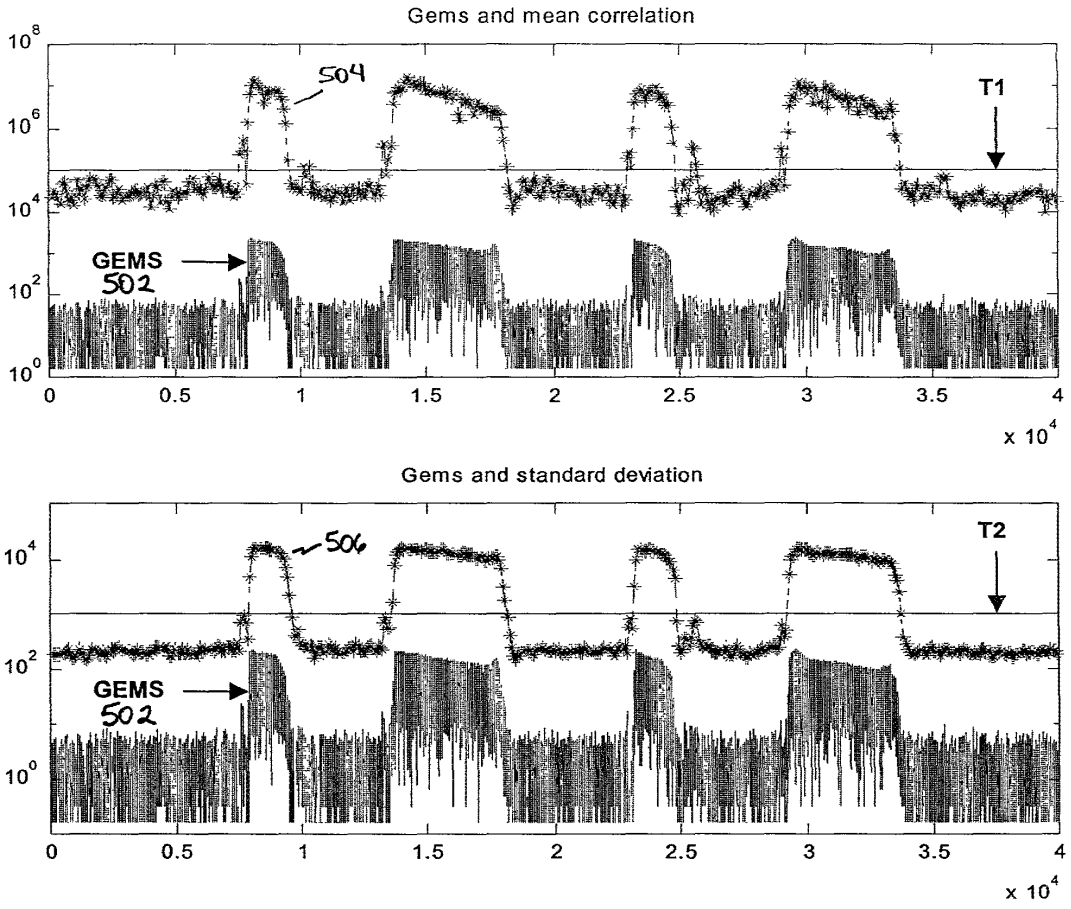


Figure 5B

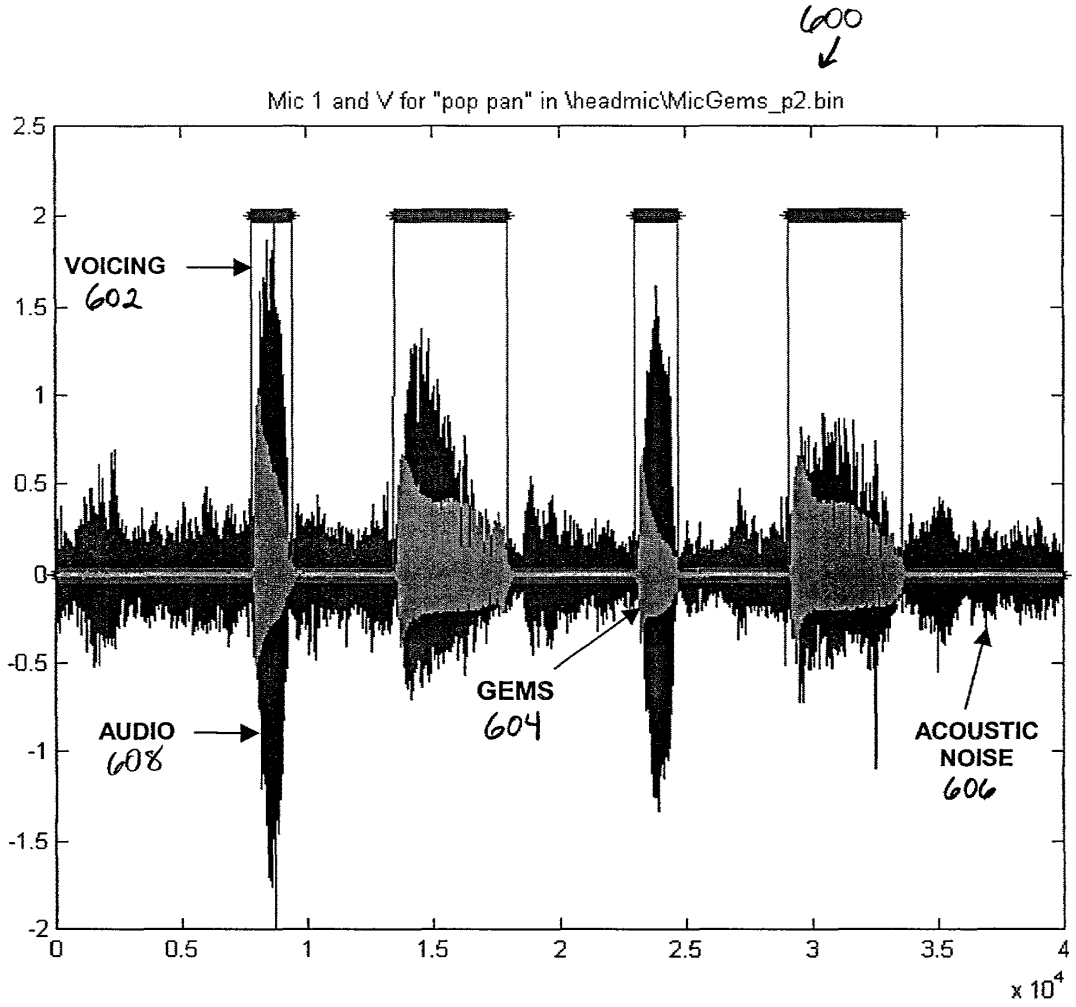

600
Mic 1 and V for "pop pan" in \headmic\MicGems_p2.bin
VOICING
602
AUDIO
608
GEMS
604
ACOUSTIC
NOISE
606
2.5
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.5
1
1.5
2
2.5
3
3.5
4
x 10^4

Figure 6

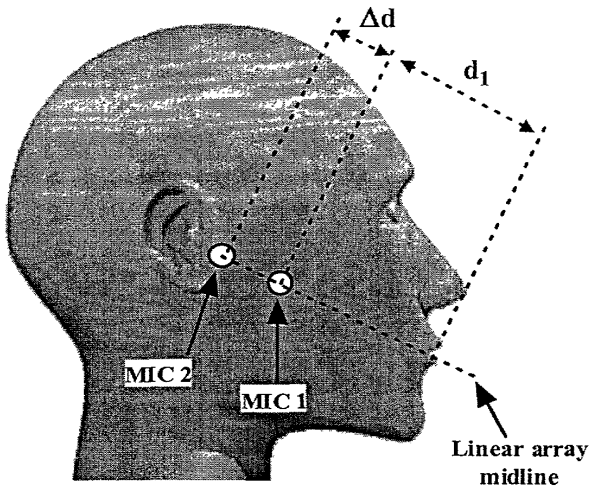


Figure 7

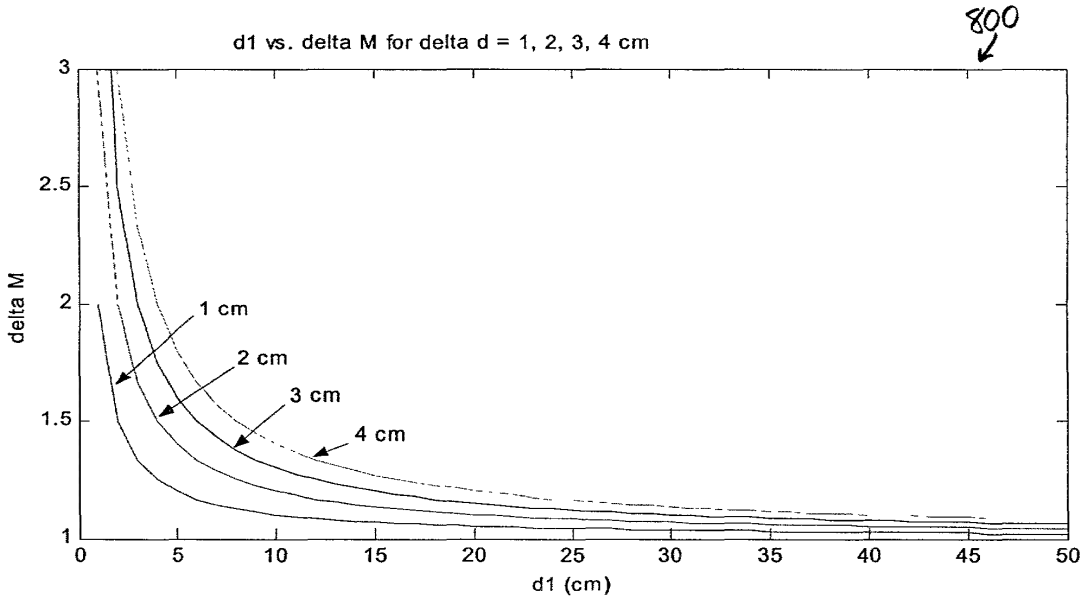


Figure 8

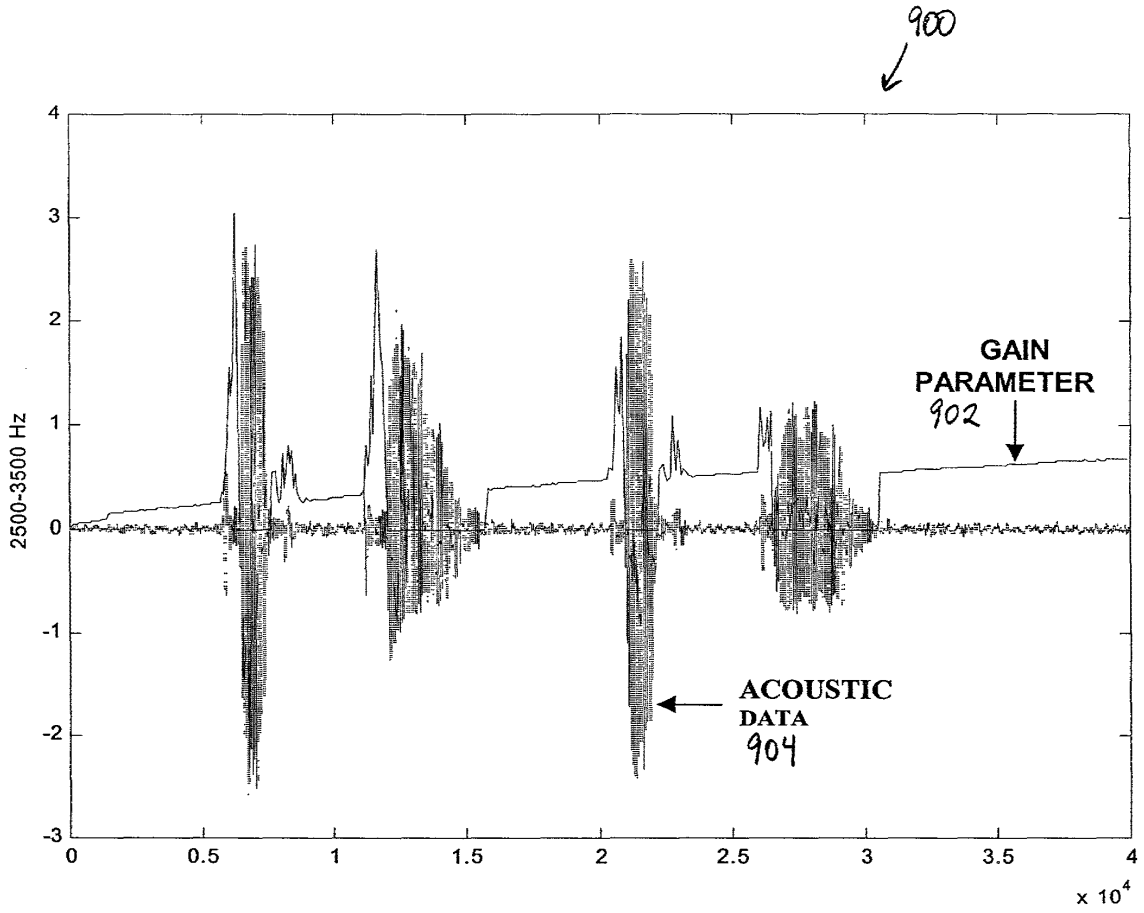


Figure 9

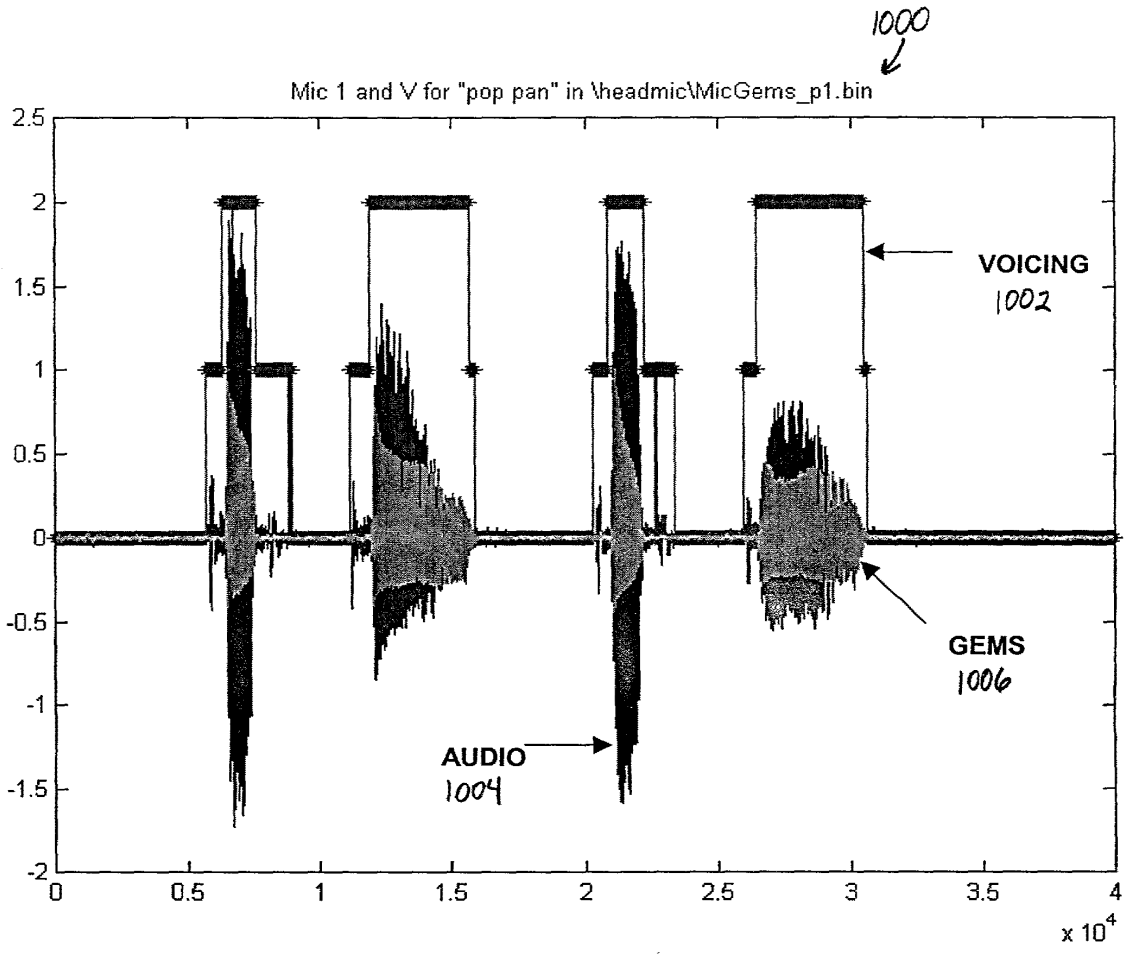

1000
Mic 1 and V for "pop pan" in \headmic\MicGems_p1.bin
VOICING
1002
GEMS
1006
AUDIO
1004
2.5
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.5
1
1.5
2
2.5
3
3.5
4
x 10^4

Figure 10

INVENTORSHIP DECLARATION BY SOLE INVENTOR

I HEREBY DECLARE THAT:

My residence, mailing address, and citizenship are stated next to my name in PART A on hereof.

I believe I am the original, first, and sole inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled

DETECTING VOICED AND UNVOICED SPEECH USING BOTH ACOUSTIC AND NONACOUSTIC SENSORS

the specification of which:

☒ is attached hereto.

☐ was filed on _____ as Application Serial No. _____ and was amended on _____ (if applicable).

I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56.

I hereby claim priority benefits under Title 35, United States Code §119 of any foreign application(s) for patent or inventor's certificate listed in PART B hereof and have also identified in PART B hereof any foreign application for patent or inventor's certificate having a filing date before that of the application of which priority is claimed.

I hereby claim the benefit under Title 35, United States Code, §119(e) and §120 of any United States application(s) listed in PART C hereof and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph by the first paragraph of Title 35, United States Code §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations §1.56 which occurred between the filing date of the prior application and any national or PCT international filing date of this application.

I hereby declare that all statements made herein of my knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

Please direct all telephone calls to Richard L. Gregory, Jr. at (650) 838-4408. Address all correspondence to:

Customer No. 22918
Perkins Coie LLP
101 Jefferson Drive
Menlo Park, California 94025-1114
(650) 838-4300

See page 2 attached, signed, and made a part hereof.

INVENTORSHIP DECLARATION BY SOLE INVENTOR

PART A: INVENTOR INFORMATION AND SIGNATURE

Full name of Sole inventor: Gregory C. Burnett

Citizenship: U.S.A. Mailing Address: 675 South H Street
Livermore, CA 94550

Residence (if different): _____

Inventor's Signature: _____ Date: _____

PART B: CLAIM TO PRIORITY OF FOREIGN APPLICATION(S) UNDER 35 U.S.C. 119(a-d)

Country	App. No.	Filing Date

PART C: CLAIM TO PRIORITY OF U.S. PROVISIONAL AND NONPROVISIONAL APPLICATION(S) UNDER 35 U.S.C. 119(e) AND 120

Serial No.	Filing Date	Status
60/294,383	5/30/01	☒ Pending
09/905,361	7/12/01	☒ Pending
60/335,100	10/30/01	☒ Pending
60/332,202	11/21/01	☒ Pending
09/990,847	11/21/01	☒ Pending
60/362,162	3/5/02	☒ Pending
60/362,103	3/5/02	☒ Pending
60/362,170	3/5/02	☒ Pending
60/361,981	3/5/02	☒ Pending
60/362,161	3/5/02	☒ Pending
60/368,209	3/27/02	☒ Pending
60/368,208	3/27/02	☒ Pending
60/368,343	3/27/02	☒ Pending

UNITED STATES PATENT AND TRADEMARK OFFICE

Commissioner for Patents
Washington, DC 20231
www.uspto.gov

Bib Data Sheet

CONFIRMATION NO. 9714

SERIAL NUMBER	FILING DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
10/159,770	05/30/2002 RULE	381	2654	ALPH.P009

APPLICANTS

Gregory C. Burnett, Livermore, CA;

**** CONTINUING DATA ***************************

This appln claims benefit of 60/294,383 05/30/2001
and claims benefit of 60/335,100 10/30/2001
and claims benefit of 60/332,202 11/21/2001
and claims benefit of 60/362,162 03/05/2002
and claims benefit of 60/362,103 03/05/2002
and claims benefit of 60/362,170 03/05/2002
and claims benefit of 60/361,981 03/05/2002
and claims benefit of 60/362,161 03/05/2002
and claims benefit of 60/368,209 03/27/2002
and claims benefit of 60/368,208 03/27/2002
and claims benefit of 60/368,343 03/27/2002

**** FOREIGN APPLICATIONS ********************

IF REQUIRED, FOREIGN FILING LICENSE GRANTED
** 07/09/2002

Foreign Priority claimed	STATE OR COUNTRY	SHEETS DRAWING	TOTAL CLAIMS	INDEPENDENT CLAIMS
☐ yes ☐ no 35 USC 119 (a-d) conditions met ☐ yes ☐ no ☐ Met after Allowance Verified and Acknowledged _____ Examiner's Signature Initials	CA	10	7	4

ADDRESS

22918

TITLE

Detecting voiced and unvoiced speech using both acoustic and nonacoustic sensors

FILING FEE RECEIVED		
954	FEES: Authority has been given in Paper No. _____ to charge/credit DEPOSIT ACCOUNT No. _____ for following:	☐ All Fees ☐ 1.16 Fees (Filing) ☐ 1.17 Fees (Processing Ext. of time) ☐ 1.18 Fees (Issue) ☐ Other _____ ☐ Credit

PATENT APPLICATION FEE DETERMINATION RECORD
Effective October 1, 2001

Application or Docket Number: 37212.8009.US00

CLAIMS AS FILED - PART I

	(Column 1)	(Column 2)
TOTAL CLAIMS	7	
FOR	NUMBER FILED	NUMBER EXTRA
TOTAL CHARGEABLE CLAIMS	7 minus 20=	* 0
INDEPENDENT CLAIMS	4 minus 3 =	* 1
MULTIPLE DEPENDENT CLAIM PRESENT		☐

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

SMALL ENTITY TYPE ☐		OR	OTHER THAN SMALL ENTITY	
RATE	FEE		RATE	FEE
BASIC FEE	370.00	OR	BASIC FEE	740.00
X$ 9=		OR	X$18=	
X42=		OR	X84=	84
+140=		OR	+280=	
TOTAL		OR	TOTAL	824

CLAIMS AS AMENDED - PART II

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
AMENDMENT A	Total	*	Minus	**	=
	Independent	*	Minus	***	=
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

SMALL ENTITY		OR	OTHER THAN SMALL ENTITY	
RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
AMENDMENT B	Total	*	Minus	**	=
	Independent	*	Minus	***	=
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

		OR		
RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA
AMENDMENT C	Total	*	Minus	**	=
	Independent	*	Minus	***	=
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM				☐

		OR		
RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
X$ 9=		OR	X$18=	
X42=		OR	X84=	
+140=		OR	+280=	
TOTAL ADDIT. FEE		OR	TOTAL ADDIT. 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.

PATENT APPLICATION SERIAL NO. _____

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE
FEE RECORD SHEET

06/06/2002 WABDELR1 00000030 502207 10159770

01 FC:101 740.00 CH
02 FC:102 84.00 CH

PTO-1556
(5/87)

*U.S. Government Printing Office: 2001 — 481-697/59173

MULTIPLE DEPENDENT CLAIM FEE CALCULATION SHEET
(FOR USE WITH FORM PTO-875)

SERIAL NO. 10/159,770 | FILING DATE

APPLICANT(S)

CLAIMS

	AS FILED		AFTER 1st AMENDMENT		AFTER 2nd AMENDMENT			*		*		*	
	IND.	DEP.	IND.	DEP.	IND.	DEP.		IND.	DEP.	IND.	DEP.	IND.	DEP.
1	1						51						
2	1						52						
3		1					53						
4		1					54						
5		1					55						
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7	1						57						
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46							96						
47							97						
48							98						
49							99						
50							100						
TOTAL IND.	4						TOTAL IND.						
TOTAL DEP.	3						TOTAL DEP.						
TOTAL CLAIMS	7						TOTAL CLAIMS						

PTO-1360 (3-78)

*MAY BE USED FOR ADDITIONAL CLAIMS OR AMENDMENTS

U.S. DEPARTMENT of COMMERCE
Patent and Trademark Office