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APPLICATION NO.	ISSUE DATE	PATENT NO.
17/215,804	20-JUN-23	11683643

Akerman LLP
777 South Flagler Drive Suite 1100 West Tower
West Palm Beach, FL 33401

EGRANT NOTIFICATION

Your electronic patent grant (eGrant) is now available, which can be accessed via Patent Center at https://patentcenter.uspto.gov

The electronic patent grant is the official patent grant under 35 U.S.C. 153. For more information, please visit https://www.uspto.gov/electronicgrants

Samsung et al. v. Techiya
IPR2024-01032
Exhibit 1004

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APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/215,804	06/20/2023	11683643	11472-39-5_CONT2-16247186	9372

150139 7590 05/31/2023

Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

ISSUE NOTIFICATION

The projected patent number and issue date are specified above. The patent will issue electronically. The electronically issued patent is the official patent grant pursuant to 35 U.S.C. § 153. The patent may be accessed on or after the issue date through Patent Center at https://patentcenter.uspto.gov/. The patent will be available in both the public and the private sides of Patent Center. Further assistance in electronically accessing the patent, or about Patent Center, is available by calling the Patent Electronic Business Center at 1-888-217-9197.

The USPTO is implementing electronic patent issuance with a transition period, during which period the USPTO will mail a ceremonial paper copy of the electronic patent grant to the correspondence address of record. Additional copies of the patent (i.e., certified and presentation copies) may be ordered for a fee from the USPTO's Certified Copy Center at https://certifiedcopycenter.uspto.gov/index.html. The Certified Copy Center may be reached at (800)972-6382.

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 104 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 Application Assistance Unit (AAU) of the Office of Patents Stakeholder Experience (OPSE), Stakeholder Support Division (SSD) at (571)-272-4200.

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

Steven Wayne Goldstein, Delray Beach, FL;
Marc Andre Boillot, Plantation, FL;
John Usher, Devon, UNITED KINGDOM;
Jason McIntosh, Sugar Hill, GA;

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

Staton Techiya LLC, Lighthouse Point, FL

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Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

PTO/SB/08a (01-22)
Approved for use through 05/31/2024. OMB 0651-0031
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/L.S/	1	10709339		2020-07-14	Hugh Lusted	
/L.S/	2	10970375		2021-04-06	Devu Manikantan	
/L.S/	3	10142332		~~2015-01-05~~ 11/2018	Sourabh Ravindran	
/L.S/	4	8851372		2014-10-07	Tiger T G Zhou	
/L.S/	5	9112701		2015-08-18	Akane Sano	
/L.S/	6	9013351		2015-04-21	James Park	
/L.S/	7	9684778		2017-06-20	Francis M. Tharappel	

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Change(s) applied to document, /T.W./ 5/24/2023

EFS Web 2.1.18

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99) Change(s) applied to document,	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

/T.D.R./
2/22/2023

/L.S/	4	H0877468	JP		1996-03-22	Keizaburo		☒
/L.S/	5	H10162283	JP		1998-06-19	Masahiro		☒
/L.S/	6	EP1519625	EP	A2	05/2005	Victorian		☐
/L.S/	7	2006037156	WO	A1	04/2006	Mejia et al.		☐
/L.S/	8	EP1401240	EP		03/2004	Jensen		☐
/L.S/	9	2008050583	WO		05/2008	Kitada		☐
/L.S/	10	2007092660	WO		08/2007	Moberly		☐
/L.S/	11	2004114722	WO		12/2004	Pedersen		☐
/L.S/	12	EP1385324	EP		01/2004	Pedersen		☐
/L.S/	13	EP1640972	EP		03/2006	Dijkstra		☐

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT2-16247186

/L.S/	24	20050123146		2005-06-09	Voix et al.		
/L.S/	25	20040125965		2004-07-01	Alberth Jr., et al.		
/L.S/	26	20060083395		2006-04-20	Allen et al.		
/L.S/	27	20110264447		2011-10-27	Visser et al.		
/L.S/	28	20100296668		2010-11-25	Lee et al.		
/L.S/	29	20110293103		2011-12-01	Park et al.		
/L.S/	30	20100061564		2010-03-11	Clemow et al.		

If you wish to add additional U.S. Published Application citation information please click the Add button. Add

FOREIGN PATENT DOCUMENTS Remove

Examiner Initial*	Cite No	Foreign Document Number[3]	Country Code[2]	Kind Code[4]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T[5]
Change(s) applied to document, /C.S./ 2/23/2023 /L.S/	1	EP1519625	EP	A2	03/2005	Victorian		
/L.S/	2	2006037156	WO	A1	04/2006	Mejia et al.		

If you wish to add additional Foreign Patent Document citation information please click the Add button Add

EFS Web 2.1.18

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

/L.S/	26	20060083395		2006-04-20	Allen et al.	
/L.S/	27	20110264447		2011-10-27	Visser et al.	
/L.S/	28	20100296668		2010-11-25	Lee et al.	
/L.S/	29	20110293103		2011-12-01	Park et al.	
/L.S/	30	20100061564		2010-03-11	Clemow et al.	
/L.S/	31	20050058313		2005-03-17	Victorian, Thomas A.	
/L.S/	32	20140122092		05/2014 ~~2019-03-19~~	Goldstein; Steven Wayne	
/L.S/	33	20090286515		2009-05-05	Othmer; Konstantin	
/L.S/	34	20060173563		2006-08-03	Borovitski; Michael D.	
/L.S/	35	20090076821		2009-03-19	Brenner; Vadim	
/L.S/	36	20050281423		2005-12-22	Armstrong	

Change(s) applied to document, /C.S/ 2/13/2023

EFS Web 2.1.18

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

/L.S/	27	20140163976		2014-06-12	Park	
/L.S/	28	20080137873		2008-06-12	Goldstein	
/L.S/	29	20090122996		2009-05-14	Klein	
/L.S/	30	20040086138		2004-05-06	Kuth	
/L.S/	31	20070021958		01/2007 ~~2008-01-25~~	Visser et al.	
Change(s) applied to document, /C.S./ 2/23/2023	32	20020111798		2002-08-15	Huang	

If you wish to add additional U.S. Published Application citation information please click the Add button. Add

FOREIGN PATENT DOCUMENTS Remove

Examiner Initial*	Cite No	Foreign Document Number[3]	Country Code[2] i	Kind Code[4]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T[5]
	1							

If you wish to add additional Foreign Patent Document citation information please click the Add button Add

NON-PATENT LITERATURE DOCUMENTS Remove

Examiner Initials*	Cite No	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, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]

Doc Code: MFEE.C.AD PTO/SB/47-EFS

Document Description: Maintenance Fee Address Change

FEE ADDRESS INDICATION FORM			
Application Number	17215804	Art Unit	2651
Filing Date	29-Mar-2021	Examiner Name	LUNSEE LAO
First Name Inventor	Steven Wayne Goldstein	Attorney Doc. Number	11472-39-5_CONT2-16247186

INSTRUCTIONS: In order for the fee address identified on this form to be effective, the issue fee must have been paid for the application listed on this form. Only an address represented by a customer number can be established as the fee address for maintenance fee purposes (hereafter, fee address). A fee address should be established when correspondence related to maintenance fees should be mailed to a different address than the correspondence address for the application. For more information on customer numbers, see the Manual of Patent Examining Procedure (MPEP) § 403.

Please recognize as the Fee Address under the provisions of 37 CFR 1.363 the address associated with:

Customer Number:	154598 Staton Techiya, LLC John Keady 9501 Jagged Creek Ct Delray Beach FL 33446 UNITED STATES (571) 606-1291 yasmin.seyal@techiya.co

Signature :

I certify, in accordance with 37 CFR 1.4(d)(4) that I am an attorney or agent registered to practice before the Patent and Trademark Office who has filed and has been granted power of attorney in this application.

Signature	/Peter A. Chiabotti/	**Date**	05-03-2023
Name	Peter A. Chiabotti	**Registration Number**	54603

Electronic Patent Application Fee Transmittal

Application Number:	17215804
Filing Date:	29-Mar-2021
Title of Invention:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Filer:	Peter Allen Chiabotti/Yoshi Hassan
Attorney Docket Number:	11472-39-5_CONT2-16247186

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
UTILITY APPL ISSUE FEE	2501	1	480	480
PUBL. FEE- EARLY, VOLUNTARY, OR NORMAL	1504	1	0	0
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**480**

Electronic Acknowledgement Receipt	
EFS ID:	47936885
Application Number:	17215804
International Application Number:	
Confirmation Number:	9372
Title of Invention:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	150139
Filer:	Peter Allen Chiabotti/Yoshi Hassan
Filer Authorized By:	Peter Allen Chiabotti
Attorney Docket Number:	11472-39-5_CONT2-16247186
Receipt Date:	03-MAY-2023
Filing Date:	29-MAR-2021
Time Stamp:	09:50:37
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	DA
Payment was successfully received in RAM	$480
RAM confirmation Number	E202353050359122
Deposit Account	
Authorized User	
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Issue Fee Payment (PTO-85B)	Web85b.pdf	60794 c15869e557982544482 95a3833fd7d7fb8273f83	no	2
Warnings:					
Information:					
2	Maintenance Fee Address Change	web85feeaddress.pdf	42515 cd829e886006ac178227882e3b5984e3584e4e57	no	1
Warnings:					
Information:					
3	Fee Worksheet (SB06)	fee-info.pdf	40430 3e183c9678e384a262ec319b1a333f02962e69ac	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	143739		

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.

Doc Code: IFEE PTOL/85B-EFS

Document Description: Issue Fee Payment (PTO-85B)

Issue Fee Transmittal Form

Application Number	Filing Date	First Named Inventor	Atty. Docket No.	Confirmation No.
17215804	29-Mar-2021	Steven Wayne Goldstein	11472-39-5_CONT2-162471	9372

TITLE OF INVENTION :

METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

Entity Status	Application Type	Art Unit	Class - Subclass	EXAMINER
Small	Utility under 35 USC 111(a)	2651	066000	LUNSEE LAO

Issue Fee Due	Publication Due	Total Fee(s) Due	Date Due	Prev. Paid Fee
$480	$0	$480	27-Jul-2023	$0

1.Change of Correspondence Address and/or Indication Of Fee Address (37 CFR 1.33 & 1.363)

Current Correspondence Address:	Current Indicated Fee Address :
150139 Akerman LLP 777 South Flagler Drive Suite 1100 West Tower West Palm Beach FL 33401 UNITED STATES 5616535000 ip@akerman.com	
☐ Change of correspondence address requested, system generated AIA/122-EFS form attached	☒ Fee Address indication requested, system generated SB/47-EFS form attached

2.Entity Status

Change in Entity Status

○ Applicant certifying micro entity status; system generated Micro Entity certification form attached. See 37 CFR 1.29.
Note: Absent a valid certification of micro entity status, issue fee payment in the micro entity amount will not be accepted at the risk of application abandonment. If this box is checked, you will be prompted to choose a micro entity status on the gross income basis (37 CFR 1.29(a)) or the institution of higher education basis (37 CFR 1.29(d)), and make the applicable certification online.

◉ Applicant asserting small entity status. See 37 CFR 1.27.
Note: If the application was previously under micro entity status, checking this box will be taken to be a notification of loss of entitlement to micro entity status.

○ Applicant changing to regular undiscounted fee status.
Note: Checking this box will be taken to be a notification of loss of entitlement to small or micro entity status, as applicable.

Doc Code: IFEE PTOL/85B-EFS

Document Description: Issue Fee Payment (PTO-85B)

3.The Following Fee(s) Are Submitted:

☒ Issue Fee	☐ I authorize USPTO to apply my previously paid issue fee to the current fees due
☐ Publication Fee	☐ The Director is hereby authorized to apply my previously paid issue fee to the current fee due and to charge deficient fees to Deposit Account Number _____
	☒ If **in addition to** the payment of the issue fee amount submitted with this form, there are any discrepancies in any amount(s) due, the Director is authorized to charge any deficiency, or credit any overpayment, to Deposit Account Number 500951 . The **issue fee must be submitted** with this form. **If payment of the issue fee does not accompany this form, checking this box and providing a deposit account number will NOT be effective to satisfy full payment of the fee(s) due.**

4.Firm and/or Attorney Names To Be Printed

NOTE: If no name is listed, no name will be printed
For printing on the patent front page, list to be displayed as entered

1. Akerman LLP
2. Peter A. Chiabotti
3.

5.Assignee Name(s) and Residence Data To Be Printed

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.

Name	City	State	Country	Category
Staton Techiya LLC	Delray Beach	FLORIDA	UNITED STATES	corporation

6.Signature

I certify, in accordance with 37 CFR 1.4(d)(4) that I am an attorney or agent registered to practice before the Patent and Trademark Office who has filed and has been granted power of attorney in this application. I also certify that this Fee(s) Transmittal form is being transmitted to the USPTO via EFS-WEB on the date indicated below.

Signature	/Peter A. Chiabotti/	**Date**	05-03-2023
Name	Peter A. Chiabotti	**Registration Number**	54603

WEB IFEE 1.0

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United States Patent and Trademark Office
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NOTICE OF ALLOWANCE AND FEE(S) DUE

150139 7590 04/27/2023

Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

EXAMINER
LAO, LUNSEE

ART UNIT	PAPER NUMBER
2651	

DATE MAILED: 04/27/2023

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/215,804	03/29/2021	Steven Wayne Goldstein	11472-39-5_CONT2-16247186	9372

TITLE OF INVENTION: METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0.00	$0.00	$480	07/27/2023

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

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

HOW TO REPLY TO THIS NOTICE:

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

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

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

For purposes of this notice, small entity fees are 40% the amount of undiscounted fees, and micro entity fees are 20% the amount of undiscounted fees.

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

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

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

PTOL-85 (Rev. 02/11)

PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), by mail or fax, or via EFS-Web.

By mail, send to: Mail Stop ISSUE FEE
Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313-1450

By fax, send to: (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 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. **Because electronic patent issuance may occur shortly after issue fee payment, any desired continuing application should preferably be filed prior to payment of this issue fee in order not to jeopardize copendency.**

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

150139 7590 04/27/2023
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

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 transmitted to the USPTO via EFS-Web or by facsimile to (571) 273-2885, on the date below.

(Typed or printed name)
(Signature)
(Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/215,804	03/29/2021	Steven Wayne Goldstein	11472-39-5_CONT2-16247186	9372

TITLE OF INVENTION: METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0.00	$0.00	$480	07/27/2023

EXAMINER	ART UNIT	CLASS-SUBCLASS
LAO, LUNSEE	2651	381-066000

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/AIA/122 or PTO/SB/122) attached.

❑ "Fee Address" indication (or "Fee Address" Indication form PTO/AIA/47 or 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 must have been previously recorded, or filed for recordation, as set forth in 37 CFR 3.11 and 37 CFR 3.81(a). 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. Fees submitted: ❑Issue Fee ❑Publication Fee (if required)

4b. Method of Payment: *(Please first reapply any previously paid fee shown above)*

❑ Electronic Payment via Patent Center or EFS-Web ❑ Enclosed check ❑ Non-electronic payment by credit card (Attach form PTO-2038)

❑ The Director is hereby authorized to charge the required fee(s), any deficiency, or credit any overpayment to Deposit Account No. _____

5. **Change in Entity Status** (from status indicated above)

❑ Applicant certifying micro entity status. See 37 CFR 1.29
NOTE: Absent a valid certification of Micro Entity Status (see forms PTO/SB/15A and 15B), issue fee payment in the micro entity amount will not be accepted at the risk of application abandonment.

❑ Applicant asserting small entity status. See 37 CFR 1.27
NOTE: If the application was previously under micro entity status, checking this box will be taken to be a notification of loss of entitlement to micro entity status.

❑ Applicant changing to regular undiscounted fee status.
NOTE: Checking this box will be taken to be a notification of loss of entitlement to small or micro entity status, as applicable.

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

Authorized Signature _____ Date _____

Typed or printed name _____ Registration No. _____

Exhibit 1004
Page 0017 of 1251

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.
17/215,804	03/29/2021	Steven Wayne Goldstein	11472-39-5_CONT2-16247186	9372

150139 7590 04/27/2023

Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

EXAMINER	
LAO, LUNSEE	
ART UNIT	**PAPER NUMBER**
2651	

DATE MAILED: 04/27/2023

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

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

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

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

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

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

The information collected by PTOL-85 Part B is required by 37 CFR 1.311. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 30 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, Virginia 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, Virginia 22313-1450. Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

Privacy Act Statement

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

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

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

Notice of Allowability	Application No. 17/215,804	Applicant(s) Goldstein et al.	
	Examiner LUNSEE LAO	Art Unit 2651	AIA (FITF) Status No

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

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

1. ☑ This communication is responsive to 04-07-2023.
   ☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on _____.

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

3. ☑ The allowed claim(s) is/are 1,3-6,9-13 and 15 . As a result of the allowed claim(s), you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information , please see **http://www.uspto.gov/patents/init_events/pph/index.jsp** or send an inquiry to **PPHfeedback@uspto.gov.**

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

   Certified copies:

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

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

   * Certified copies not received: _____ .

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

5. ☐ CORRECTED DRAWINGS (as "replacement sheets") must be submitted.
   ☐ including changes required by the attached Examiner's Amendment / Comment or in the Office action of Paper No./Mail Date _____ .

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

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

Attachment(s)

1. ☑ Notice of References Cited (PTO-892)
2. ☑ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date 04-07-2023.
3. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material _____.
4. ☐ Interview Summary (PTO-413), Paper No./Mail Date. _____.
5. ☐ Examiner's Amendment/Comment
6. ☑ Examiner's Statement of Reasons for Allowance
7. ☐ Other _____.

/LUN-SEE LAO/
Primary Examiner, Art Unit 2651

Notice of Pre-AIA or AIA Status

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

DETAILED ACTION

Introduction

2. This action responds to the amendment filed on 04-07-2023. Claims 1, 3-6, 9-13 and 15 have been amended and claims 2, 7, 8, and 14 cancelled and claim 15 has been added. Claims 1, 3-6, 9-13 and 15 are pending.

Allowable Subject Matter

3. Claims 1, 3-6, 9-13 and 15 are allowed over prior art record.

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

The amendment filed on 04-07-2023 have been carefully reviewed with updated search. Consequently, reasons for allowance of Claims 1, 3-6 9-13 and 15 are set forth in according to the applicant's amendment state on pages 6.

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

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Le Faucheur (US PAT.9,384,726) is cited to show other related the METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION.

EXAMINER'S COMMENT

6. Applicant should note that this application is co-pending with US number US 2021/0211801 and since it appears that this application may issue first, applicant should take care to address any double-patenting issues that may remain in the 17/211,814 application.

7. Any response to this action should be mailed to:

Mail Stop ____(explanation, e.g., Amendment or After-final, etc.)

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

Facsimile responses should be faxed to:
(571) 273-8300

Hand-delivered responses should be brought to:
Customer Service Window
Randolph Building
401 Dulany Street
Alexandria, VA 22314

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Lao,Lun-See whose telephone number is (571) 272-7501 The examiner can normally be reached on Monday-Friday from 8:00 to 5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Nguyen Duc M(SPE), can be reached on (571) 272-7503.

Any inquiry of a general nature or relating to the status of this application or proceeding

should be directed to the Technology Center 2600 whose telephone number is (571) 272-2600.

/LUN-SEE LAO/
Primary Examiner, Art Unit 2651
US Patent and Trademark Office
Knox
571-272-7501
Date 04-20-2023

Notice of References Cited	Application/Control No. 17/215,804	Applicant(s)/Patent Under Reexamination Goldstein et al.	
	Examiner LUNSEE LAO	Art Unit 2651	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	CPC Classification	US Classification
*	A	US-9384726-B2	07-2016	Le Faucheur; Laurent	G10K11/1783	1/1
	B					
	C					
	D					
	E					
	F					
	G					
	H					
	I					
	J					
	K					
	L					
	M					

FOREIGN PATENT DOCUMENTS

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

NON-PATENT DOCUMENTS

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

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

Index of Claims	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/215,804	Goldstein et al.
	Examiner	**Art Unit**
	LUNSEE LAO	2651

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

CLAIMS

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

CLAIM		DATE								
Final	Original	10/19/2022	02/03/2023	04/20/2023						
1	1	✓	=	=						
	2	✓	-	-						
2	3	✓	=	=						
3	4	✓	=	=						
4	5	✓	=	=						
5	6	✓	=	=						
	7	✓	-	-						
	8	✓	-	-						
6	9	✓	=	=						
7	10	✓	=	=						
8	11	✓	=	=						
9	12	✓	=	=						
10	13	✓	=	=						
	14	✓	-	-						
11	15		=	=						

Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/215,804	Goldstein et al.
	Examiner	Art Unit
	LUNSEE LAO	2651

CPC - Searched*		
Symbol	Date	Examiner
H04R1/1016 H04R25/02 H04R3/002 H04R25/453 H04R2460/01 H04R3/02 H04R2201/107 H04R2460/05 H04R3/005 H04R1/1041 H04R1/1083 H04R2460/15 H04M3/568 H04M9/08 G10K11/17819 G10K11/17823 G10K11/17881 G10K11/17885 G10K2210/1081	04/19/2023	L.S.

CPC Combination Sets - Searched*		
Symbol	Date	Examiner

US Classification - Searched*			
Class	Subclass	Date	Examiner
381	66, 56-58, 312, 314, 74, 92, 94.1-94.5	04/19/2023	L.S.

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

Search Notes		
Search Notes	Date	Examiner
PE2E	04/19/2023	L.S.
CHECK 101	04/19/2023	L.S.

Interference Search			
US Class/CPC Symbol	US Subclass/CPC Group	Date	Examiner
	UPDATE ABOVE SEARCHED	04/19/2023	L.S.

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/215,804	Goldstein et al.
	Examiner	Art Unit
	LUN-SEE LAO	2651

CPC						
Symbol					Type	Version
H04R	/	3	/	002	F	2013-01-01
H04R	/	25	/	02	I	2013-01-01
H04R	/	1	/	1016	I	2013-01-01

CPC Combination Sets								
Symbol					Type	Set	Ranking	Version
	/		/					

NONE		Total Claims Allowed:	
(Assistant Examiner)	(Date)	11	
/LUN-SEE LAO/ Primary Examiner, Art Unit 2651	19 April 2023	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	3

U.S. Patent and Trademark Office — Part of Paper No.: 20230419

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/215,804	Goldstein et al.
	Examiner	**Art Unit**
	LUN-SEE LAO	2651

INTERNATIONAL CLASSIFICATION					
CLAIMED					
H04R3/00	/	3	/	00	
H04R25/02	/	25	/	02	
H04R1/10	/	1	/	10	

NON-CLAIMED					
	/		/		

US ORIGINAL CLASSIFICATION	
CLASS	**SUBCLASS**
381	66

CROSS REFERENCES(S)						
CLASS	**SUBCLASS (ONE SUBCLASS PER BLOCK)**					

NONE (Assistant Examiner)	(Date)	**Total Claims Allowed:** 11	
/LUN-SEE LAO/ Primary Examiner, Art Unit 2651 (Primary Examiner)	19 April 2023 (Date)	O.G. Print Claim(s) 1	O.G. Print Figure 3

U.S. Patent and Trademark Office — Part of Paper No.: 20230419

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/215,804	Goldstein et al.
	Examiner	Art Unit
	LUN-SEE LAO	2651

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

CLAIMS

Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original	Final	Original
1	1	7	10												
	2	8	11												
2	3	9	12												
3	4	10	13												
4	5		14												
5	6	11	15												
	7														
	8														
6	9														

NONE		Total Claims Allowed:	
(Assistant Examiner)	(Date)	11	
/LUN-SEE LAO/ Primary Examiner, Art Unit 2651	19 April 2023	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	3

U.S. Patent and Trademark Office — Part of Paper No.: 20230419

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. 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 contains a valid OMB control number.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1					

If you wish to add additional U.S. Patent citation information please click the Add button. Add

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Examiner Initial*	Cite No	Publication Number	Kind Code[1]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/L.S/	1	20110187640		2011-08-04	Jacobsen et al.	
/L.S/	2	20100328224		2010-12-30	Kerr et al.	
/L.S/	3	20060064037		2006-03-23	Shalon et al.	
/L.S/	4	20030033152		2003-02-13	Cameron	
/L.S/	5	20050283369		2005-12-22	Clauser et al.	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

/L.S/	6	20160058378		2016-03-03	Wisby et al.	

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FOREIGN PATENT DOCUMENTS Remove

Examiner Initial*	Cite No	Foreign Document Number[3]	Country Code[2]	Kind Code[4]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T[5]
/L.S/	1	WO9326085	WO	A1	1993-12-23	Hohman		X
/L.S/	2	JP3353701	JP	B2	2002-12-03	Kondo		☐

If you wish to add additional Foreign Patent Document citation information please click the Add button Add

NON-PATENT LITERATURE DOCUMENTS Remove

Examiner Initials*	Cite No	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, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]
/L.S/	1	90/015,146, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., REQUEST FOR EX PARTE REEXAMINATION OF U.S. PATENT NO. 10,979,836	

If you wish to add additional non-patent literature document citation information please click the Add button Add

EXAMINER SIGNATURE

Examiner Signature	/LUN-SEE LAO/	Date Considered	04/20/2023

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

[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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 1 hour 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.**

Privacy Act Statement

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(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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/L.S/	1	WO9326085	WO	A1	1993-12-23	Hohman		X
/L.S/	2	JP3353701	JP	B2	2002-12-03	Kondo		☐

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	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

Examiner Initials*	Cite No	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, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]
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Examiner Signature	/LUN-SEE LAO/	Date Considered	04/20/2023

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[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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CERTIFICATION STATEMENT

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OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/L.S/	1	7444353		2008-10-28	Chen	
/L.S/	2	8018337		2011-09-13	Jones	
/L.S/	3	6738482		2004-09-23	Jaber	
/L.S/	4	7003099		2006-02-21	Zhang	
/L.S/	5	8477955		2013-07-02	Engle	
/L.S/	6	8270629		2012-09-18	Bothra	
/L.S/	7	7853031		2010-12-14	Hamacher	
/L.S/	8	5550923		1996-08-27	Hotvet	

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/L.S/	9	6647368		2003-11-11	Nemirovski	
/L.S/	10	7512245		2009-03-31	Rasmussen	
/L.S/	11	8189803		2012-05-29	Bergeron	
/L.S/	12	7936885		2011-05-03	Frank	
/L.S/	13	5251263		1993-10-05	Andrea	
/L.S/	14	9113240		2015-08-18	Ramakrishman	
/L.S/	15	7529379		2009-05-05	Zurek	

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/L.S/	1	20120170412		2012-07-05	Calhoun	
/L.S/	2	20080221880		2008-09-11	Cerra et al.	

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/L.S/	3	20060083387		2006-04-20	Emoto	
/L.S/	4	20050102142		2005-05-12	Soufflet	
/L.S/	5	20080159547		2008-07-03	Schuler	
/L.S/	6	20050227674		2005-10-13	Kopra	
/L.S/	7	20040133421		2004-07-08	Burnett	
/L.S/	8	20020193130		2002-12-19	Yang	
/L.S/	9	20050071158		2005-03-31	Byford	
/L.S/	10	20070092087		2007-04-26	Bothra	
/L.S/	11	20040202340		2004-10-14	Armstrong	
/L.S/	12	20050281422		2005-12-22	Armstrong	
/L.S/	13	20070223717		2007-09-27	Boersma	

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/L.S/	14	20030035551		2003-02-20	Light	
/L.S/	15	20080063228		2008-03-13	Mejia	
/L.S/	16	20070100637		2007-05-03	McCune	
/L.S/	17	20070253569		2007-11-01	Bose	
/L.S/	18	20070160243		2007-07-12	Dijkstra	
/L.S/	19	20080130908		2008-06-05	Cohen	
/L.S/	20	20070014423		2007-04-26	Darbut	

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Examiner Initials*	Cite No	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, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]
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Examiner Signature	/LUN-SEE LAO/	Date Considered	04/20/2023

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Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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/L.S/	1	4,533,795	A	1985-08-06	Baumhauer, Jr. et al.	
/L.S/	2	5,251,263		1993-10-05	Andrea et al.	
/L.S/	3	6,647,368	B2	2003-11-11	Nemirovski	
/L.S/	4	6,671,379	B2	2003-12-30	Nemirovski	
/L.S/	5	6,728,385	B2	2004-04-27	Kvaloy et al.	
/L.S/	6	6,754,359	B1	2004-06-22	Svean et al.	
/L.S/	7	7,502,484	B2	2009-03-10	Ngia et al.	
/L.S/	8	7,903,826	B2	2011-03-08	Boersma	

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/L.S/	9	7,953,241	B2	2011-05-31	Jorgensen et al.	
/L.S/	10	7,983,433	B2	2011-07-19	Nemirovski	
/L.S/	11	8,014,553	B2	2011-09-06	Radivojevic et al.	
/L.S/	12	8,150,084	B2	2012-04-03	Jessen et al.	
/L.S/	13	8,351,634	B2	2013-01-08	Khenkin	
/L.S/	14	9,135,797	B2	2015-09-15	Couper et al.	

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/L.S/	1	20010046304	A1	2001-11-29	Rast	
/L.S/	2	20030035551	A1	2003-02-20	Light et al.	
/L.S/	3	20040047486	A1	2004-03-11	Van Doom et al.	

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/L.S/	4	20060083388	A1	2006-04-20	Rothschild	
/L.S/	5	20060153394	A1	2006-07-13	Beasley	

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/L.S/	1	1519625	EP	A2	2005-03-30	Starkey Laboratories Inc		
/L.S/	2	1640972	EP	A1	2006-03-29	Sonova Holding AG		

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/L.S/	1	90/019,169, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., REQUEST FOR EX PARTE REEXAMINATION OF U.S. PATENT NO. 11,244,666	
/L.S/	2	Oshana, DSP Software Development Techniques for Embedded and Real-Time Systems, Introduction, pp. xi-xvii (2006)	
/L.S/	3	Mulgrew et al., Digital Signal Processing: Concepts and Applications, Introduction, pp. xxiii-xxvi (2nd ed. 2002)	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

EXAMINER SIGNATURE			
Examiner Signature	/LUN-SEE LAO/	Date Considered	04/20/2023

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Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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Bibliographic Data

Application No: 17/215,804

Foreign Priority claimed: ○ Yes ◉ No

35 USC 119 (a-d) conditions met: ☐ Yes ☑ No ☐ Met After Allowance

Verified and Acknowledged: /LUN-SEE LAO/ Examiner's Signature | Initials

Title: METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

FILING or 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
03/29/2021 RULE	381	2651	11472-39-5_CONT2-16247186

APPLICANTS

Staton Techiya LLC, Lighthouse Point, FL, UNITED STATES

INVENTORS

Steven Wayne Goldstein, Delray Beach, FL, UNITED STATES

Marc Andre Boillot, Plantation, FL, UNITED STATES

John Usher, Devon, UNITED KINGDOM

Jason McIntosh, Sugar Hill, GA, UNITED STATES

CONTINUING DATA

This application is a CIP of 16247186 01/14/2019 PAT 11057701

16247186 is a CON of 13956767 08/01/2013 PAT 10182289

13956767 is a CON of 12170171 07/09/2008 PAT 8526645

12170171 is a CIP of 12115349 05/05/2008 PAT 8081780

12115349 has PRO of 60916271 05/04/2007

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Attorney Docket Number	11472-39-5_CONT2-16247186

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/L.S/	1	5002151		1991-03-26	Oliveira; Robert J., et al.	
/L.S/	2	7801318		2010-09-21	Bartel; Roland	

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/L.S/	1	20070019817		2007-01-25	Siltmann; Simone	
/L.S/	2	20050096899		2005-05-05	Padhi; Kabi P. et al.	
/L.S/	3	20060188105		2006-08-24	Baskerville; Orval, et al.	
/L.S/	4	20010046304		2001-11-29	Rast; Rodger H.	

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/L.S/	1	Ronald M. Aarts, Roy Irwan, and Augustus J. E.. Janssen, Efficient Tracking of the Cross-Correlation Coefficient, IEEE Transactions on Speech and Audio Processing, Vol. 10, No. 6, September 2002.	
/L.S/	2	Robert Oshana, DSP Software Development Techniques for Embedded and Real-Time Systems, Embedded Technology Series, Elsevier Inc., 2006, ISBN-10: 0-7506-7759-7.	

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Doc description: Information Disclosure Statement (IDS) Filed

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/L.S/	1	20080111004	KR	A	2008-12-22	SONY ERICSSON MOBILE COMM AB		☒
/L.S/	2	2013501969	JP	A	2013-01-17	Georgiou et al.		☒
/L.S/	3	TWM568011U	TW	U	2018-10-01	MOBILE DEVICES INC		☒

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/L.S/	4	WO2008077981	WO	A1	2008-07-03	Farzin Tahmassebi		☒
/L.S/	5	ES2273616	ES	A1	2007-12-16	Farzin Tahmassebi		☒
/L.S/	6	CN105554610	CN	B	2019-01-04	Beijing Xiaoniao Tingting Technology Co., Ltd.		☒
/L.S/	7	EP2963647	EP	B1	2019-07-31	Harman International Industries, Incorporated		☐
/L.S/	8	CN203761556	CN	U	2014-08-06	FUNG SHING COMPANY LTD		☒
/L.S/	9	CN105637892	CN	A	2020-03-13	Bose Corp		☒
/L.S/	10	JP6389232	JP	B2	2018-09-12	Silas Logic, Inc.		☒

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6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

PE2E SEARCH - Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
L1	198	voice SAME activity SAME internal SAME level SAME background SAME noise	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 02:34 PM
L2	61	voice SAME activity SAME internal SAME level SAME background SAME noise	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/14 02:41 PM
L3	59	L2 AND ((H04R3/005 OR H04R1/1016 OR H04R25/02 OR H04R3/00 OR H04R1/1041 OR H04R1/1083 OR H04R2201/107 OR H04R2410/05 OR H04R25/505 OR H04R29/004 OR H04R2460/01 OR H04R3/02 OR H04R1/08 OR H04R2430/01 OR H04R25/50 OR H04R3/002 OR H04R2460/05 OR H04R3/04 OR H04R25/453 OR H04R2499/13 OR H04R1/10 OR H04R2420/07 OR H04R2499/11 OR H04R25/305 OR H04R5/04 OR H04R2460/07 OR H04R2460/15 OR G10L25/78 OR G10L2021/02087 OR G10L21/0264 OR G10L15/20 OR G10L19/00 OR	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/14 02:42 PM

		G10L15/22 OR G10L17/00 OR G10L2015/223 OR G10L25/06 OR G10L21/0208 OR H04N1/00411 OR H04N21/4312 OR H04W4/02 OR H04W4/021 OR H04W4/029 OR H04S1/002 OR H04S3/02 OR G10K11/17819 OR G10K11/17823 OR G10K11/17881 OR G10K11/17885 OR G10K2210/1081).cpc.)					
L4	10	generat$3 SAME voice SAME activity SAME (background OR ambient) SAME noise SAME mix$3 SAME ratio SAME ear SAME canal	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/14 02:45 PM
L5	10	L4 AND ((H04R1/1016 OR H04R25/02 OR H04R3/002 OR H04R25/453 OR H04R2460/01 OR H04R3/02 OR H04R2201/107 OR H04R2460/05 OR H04R3/005 OR H04R1/1041 OR H04R1/1083 OR H04R2460/15 OR H04M3/568 OR H04M9/08 OR G10K11/17819 OR G10K11/17823 OR G10K11/17881 OR G10K11/17885 OR G10K2210/1081).cpc.)	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/14 02:48 PM
L6	300	"8600085," "9584896," "9270244," "9936297,9508335," "10499139," "8798283," "9953626"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO;	OR	ON	ON	2022/10/14 03:51 PM

			JPO; DERWENT; IBM_TDB)				
L7	40	"11057701," "10182289,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 03:53 PM
L8	10	"11057701"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 03:55 PM
L9	77	"8526645," "8081780"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 03:58 PM
L10	18	"20170345406," "20140241553," "20070177741," "20080205664," "20110103606," "20170345406"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS,	OR	ON	ON	2022/10/14 04:02 PM

			IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L11	298	"6804638," "6163508," "6400652," "7433714," "7562020," "9135797," "9123343," "7756285"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:05 PM
L12	1016	"5903868," "3876843," "5208867," "6226389," "8493204," "6661901," "7430299," "6567524," "7107109," "4947440," "9037458"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:13 PM
L13	461	"4054749," "6298323," "6804643," "6163338," "6748238," "6021325," "8750295," "7920557," "20030265246"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:15 PM

L14	74	"20050288057," "20060195322," "20070043563," "20010046304," "20070189544," "20060092043," "20020106091," "20040109668," "20020118798," "20040042103," "20030161097"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:17 PM
L15	76	"20160104452," "20040203351," "20080165988," "20070291953," "20110096939," "20060067551," "20090024234," "20090010456," "20070086600," "20040190737," "20040196992"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:19 PM
L16	22	"20050123146," "20040125965," "20060083395," "20110264447," "20100296668,"20110293103	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:25 PM
L17	298	"6804638," "6163508," "6400652," "7433714," "7562020," "9135797," "9123343," "7756285"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO,	OR	ON	ON	2022/10/14 04:33 PM

			MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L18	1016	"5903868," "3876843," "5208867," "6226389," "8493204," "6661901," "7430299," "6567524," "7107109," "4947440," "9037458"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:36 PM
L19	461	"4054749," "6298323," "6804643," "6163338," "6748238," "6021325," "8750295," "7920557," "20030265246"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:37 PM
L20	383	"7177433," "9053697," "7903826," "8150044," "8577062," "8774433," "8855343," "9191740," "9628896," "5577511," "6056698"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:40 PM
L21	1127	"5317273," "8162846," "7280849," "8045840,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP,	OR	ON	ON	2022/10/14 04:48 PM

		"6028514," "6408272," "20030165246," "20050288057," "20060195322"	AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L22	79	"20070043563," "20010046304," "20070189544," "20060092043," "20020106091," "20040109668," "20020118798," "20040042103," "20030161097," "20160104452," "20040203351"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:50 PM
L23	76	"20080165988," "20070291953," "20110096939," "20060067551," "20090024234," "20090010456," "20070086600," "20040190737," "20040196992," "20050123146," "20040125965"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 04:52 PM
L24	64	"20060083395," "20110264447," "20100296668," "20110293103," "20100061564," "20050058313," "20140122092," "20090286515," "20060173563," "20090076821," "20050281423"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO,	OR	ON	ON	2022/10/14 04:54 PM

			RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L25	55	"20030130016," "20050207605," "20070143820," "20110055256," "20060182287," "20030198359," "20070086600," "20060287014," "20100119077"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 05:02 PM
L26	877	"6359993," "7072482," "7778434," "5524056," "6415034,8014553," "408849," "7209569"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 05:10 PM
L27	8	(US-20220823-$ | US-6415034-$ | US-6754359-$ | US-20050058313-$ | US-20050168824-$).DID.	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/14 05:13 PM
L28	511	"20030161097," "6728385," "6163338," "6804638," "20040042103,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU,	OR	ON	ON	2022/10/15 03:01 PM

		"20010046304"	CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L29	183	"6754359"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:03 PM
L30	29	"20070021958," "20080037801," "20020111798"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:07 PM
L31	6	"20070021958," "20080037801," "20020111798"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/15 03:08 PM
L32	8	"20060204014," "20050078838"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/15 03:08 PM
L33	877	"6359993," "7072482," "7778434," "5524056," "6415034,8014553," "408849," "7209569"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS,	OR	ON	ON	2022/10/15 03:46 PM

			IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L34	183	"6754359"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:49 PM
L35	20	(US-6021207-$ | US-6081732-$ | US-6118878-$ | US-5131032-$ | US-6169912-$ | US-6570985-$ | US-5692059-$ | US-6738482-$ | US-6760453-$ | US-5963901-$).DID.	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:52 PM
L36	873	"6754359," "6021207," "5267321," "7477756," "7450730," "6567524," "6728385," "8140325," "8983081," "5327506"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:55 PM

L37	559	"5933510," "8086093," "7039585," "6606598," "6005525," re3851, "7903825," "8014553," "8160261," "6639987," "5923624"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 03:58 PM
L38	436	"8160273," "7464029," "7983907," "8254591," "8718305," "8798278," "8917894," "8625818," "9491542," "8218784," "6118877"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 04:03 PM
L39	295	"7050592," "8401200," "9196247," "7574917," "5276740," "20030165246," "20080037801," "20060204014," "20050078838"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 04:05 PM
L40	43	"20070036377," "20020165719," "20060140425," "20030152359," "20070003090," "20030165319," "20070255435," "20040264938," "20060264176,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO,	OR	ON	ON	2022/10/15 04:08 PM

		"20040109579," "20060167687"	MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L41	82	"20060083390," "20050068171," "20050028212," "20080145032," "20020098878," "20060188075," "20140023203," "20060188105," "20020076057," "20110116643," "20150215701"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 04:10 PM
L42	44	"20140163976," "20080137873," "20090122996," "20040086138," "20070021958," "20020111798"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 04:14 PM
L43	298	"6804638," "6163508," "6400652," "7433714," "7562020," "9135797," "9123343," "7756285"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/15 05:11 PM
L44	1016	"5903868," "3876843," "5208867," "6226389,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP,	OR	ON	ON	2022/10/15 05:13 PM

		"8493204," "6661901," "7430299," "6567524," "7107109," "4947440," "9037458"	AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L45	461	"4054749," "6298323," "6804643," "6163338," "6748238," "6021325," "8750295," "7920557," "20030265246"	(US-PGPUB; USPAT; USOCR; FIT (AP, AT, AU, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, MC, MD, MY, NL, NO, NZ, OA, PH, PL, PT, RO, RS, RU, SE, SG, SI, SK, SU, TH, TN, TR, TW, UA, VN, WO); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 09:39 AM
L46	74	"20050288057," "20060195322," "20070043563," "20010046304," "20070189544," "20060092043," "20020106091," "20040109668," "20020118798," "20040042103," "20030161097"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 10:26 AM
L47	76	"20160104452," "20040203351," "20080165988," "20070291953," "20110096939," "20060067551," "20090024234," "20090010456," "20070086600," "20040190737," "20040196992"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO,	OR	ON	ON	2022/10/17 10:34 AM

			RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L48	25	"20050123146," "20040125965," "20060083395," "20110264447," "20100296668,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 10:40 AM
L49	10	"8045840"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 10:43 AM
L50	16	(US-4809262-$\|US-5259033-$|US-6304648-$\|US-6381572-$|US-6647368-$\|US-6754359-$|US-6870807-$\|US-5999828-$).DID.	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 10:48 AM
L51	530	"7236580," "7783054," "7986802," "8060366," "8275145," "8380521," "8401178," "8838184,"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU,	OR	ON	ON	2022/10/17 02:44 PM

		"7039195," "8081780," "6728385"	CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L52	1423	"6738482," "6081732," "6169912," "6570985," "7349353," "5131032," "6760453," "5963901," "6021207," "5692059," "6118878"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 02:50 PM
L53	648	"6631196," "7817803," "8027481," "5796819," "703608," "7004097," "6466666," "20080037801," "20010046304"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 02:53 PM
L54	58	"20030112947," "20040137969," "20050102133," "20060062395," "20070291953," "20080019539," "20090147966," "20120184337," "20130051543," "20140370838," "20150288823"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA,	OR	ON	ON	2022/10/17 03:00 PM

			VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L55	64	"20090010444," "20050058313," "20050069161," "20070189544," "20040202340," "20090034748," "20070036342," "20060067512"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/17 03:04 PM
L56	8	(US-20220823-$ | US-6415034-$ | US-6754359-$ | US-20050058313-$ | US-20050168824-$).DID.	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/18 10:09 AM
L57	31	"1385324"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/18 10:24 AM
L58	6348	Jens.inv. AND background AND noise	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB,	OR	ON	ON	2022/10/18 10:33 AM

			HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L59	375220	Jens Erik.inv. AND background AND noise	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/18 10:36 AM
L60	18	"6,754,359"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/18 01:09 PM
L61	183	"6754359"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT;	OR	ON	ON	2022/10/18 01:09 PM

			IBM_TDB)				
L62	35	"8526645"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/18 02:06 PM
L63	19	("5182774" | "8526645" | "9173045" | "9729957" | "9749732" | "20070098187" | "20080107287" | "20100002891" | "20110280411" | "20140254828" | "20140294191" | "20150010158" | "20150023515" | "20160232889" | "20170223443" | "20170245045" | "20180122400" | "20180288546" | "20190313196").pn. OR ("11206003").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/18 02:06 PM
L64	51	"8081780"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/18 02:09 PM
L65	141	("2850012" | "2876767" | "3110356" | "3505999" | "3602654" | "4006796" | "4029083" | "4060080" | "4834211" | "4896679" | "4913165" | "5131411" | "5333622" | "6256396" | "6368288" | "6368289" | "6440102" | "7171371" | "7756281" | "7779844" | "7817803" | "7822219" | "7886745" | "7913696" | "8018328" | "8047207" | "8081780" | "8111839" | "8142870" | "8150043" | "8150044" | "8170228" | "8194864" | "8194865" | "8199919" | "8208644"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/18 02:10 PM

		| "8208652" | "8213629" | "8213649" | "8221860" | "8221861" | "8229128" | "8251925" | "8311228" | "8312960" | "8315400" | "8319620" | "8326628" | "8326635" | "8437492" | "8447031" | "8499886" | "9479859" | "9485595" | "10917711" | "11102563" | "20020143242" | "20070270988" | "20080015463" | "20080031475" | "20080037797" | "20080046246" | "20080130906" | "20080137873" | "20080144840" | "20080144841" | "20080144842" | "20080176289" | "20080178088" | "20080181419" | "20080181442" | "20080205660" | "20080212787" | "20080219456" | "20080219486" | "20080240458" | "20080253583" | "20080267416" | "20080299339" | "20080311324" | "20090016541" | "20090016542" | "20090022294" | "20090022353" | "20090028356" | "20090034765" | "20090067661" | "20090071486" | "20090071487" | "20090101433" | "20090130423" | "20090146799" | "20090147966" | "20090154748" | "20090155518" | "20090192407" | "20090214072" | "20090220096" | "20090238374" | "20090238386" | "20090240497" | "20090245530" | "20090264161" |					

		"20090290721" | "20100002897" | "20100012420" | "20100033313" | "20100071707" | "20100074451" | "20100076793" | "20100135502" | "20100142715" | "20100142725" | "20100177918" | "20100241256" | "20100322454" | "20110079227" | "20110085689" | "20110115626" | "20110228963" | "20110235843" | "20110311079" | "20120101514" | "20120103346" | "20120123573" | "20120177209" | "20120177210" | "20120288104" | "20130035608" | "20130039518" | "20130098706" | "20130123919" | "20160057551" | "20190111456" | "20200314519" | "20210152924" | "20210243514").pn. OR ("11477560").urpn. AND (PGPB | USPT | USOC).dbnm.					
L66	98	"6415034"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/18 03:29 PM
L67	2	"20050168824"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/19 11:53 AM
L68	97	(audio adj3 content) SAME voice SAME	(US-PGPUB; USPAT; USOCR; FIT (AU, AP,	OR	ON	ON	2022/10/19 12:14 PM

		activity SAME threshold	AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L69	50	(audio adj3 content) SAME (voice adj3 activity) SAME threshold	(US-PGPUB; USPAT; USOCR; FIT (AP, AT, AU, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, MC, MD, MY, NL, NO, NZ, OA, PH, PL, PT, RO, RS, RU, SE, SG, SI, SK, SU, TH, TN, TR, TW, UA, VN, WO); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2022/10/19 12:47 PM
L70	21	(audio adj3 content) SAME (voice adj3 activity) SAME threshold	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/19 12:49 PM
L71	36	(audio adj3 content) SAME (voice SAME activity) SAME threshold	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/19 12:50 PM
L72	292	(audio adj3 content) SAME (voice adj3 activity)	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/19 12:51 PM
L73	274	L72 AND ((H04R3/005 OR H04R1/1041 OR H04R1/1016 OR H04R1/1083 OR H04R2420/07 OR H04R3/00 OR H04R25/505 OR H04R2410/05 OR H04R25/02 OR H04R3/002 OR H04R2201/107 OR H04R2430/01 OR H04R29/008 OR H04R2460/01 OR H04R2460/05 OR	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/10/19 01:03 PM

		H04R2460/15 OR H04R2499/13 OR H04R29/004 OR H04R1/08 OR H04R29/00 OR H04R25/554 OR H04R2225/55 OR H04R2430/03 OR H04R25/43 OR H04R25/50 OR H04R29/001 OR H04R3/02 OR H04R1/1075 OR H04R1/406 OR H04R3/04 OR H04R2420/01 OR H04R25/407 OR H04R25/453 OR H04R2499/11 OR H04R1/10 OR H04R2225/41 OR H04R2225/43 OR H04R2460/09 OR H04R25/40 OR H04R25/558 OR H04R25/604 OR H04R25/652 OR H04R1/028 OR H04R2201/401 OR H04R2225/61 OR H04R2460/13 OR H04R25/305 OR H04R25/65 OR H04R5/04 OR H04R1/02 OR H04R2225/023 OR H04R2225/025 OR H04R2225/39 OR H04R2430/20 OR H04R2430/21 OR H04R2460/11 OR H04R25/405 OR H04R25/48 OR H04R25/603 OR H04R1/1091 OR H04R2201/028 OR H04R2201/103 OR H04R2201/109 OR H04R2227/003 OR H04R2227/009 OR H04R25/30 OR H04R25/552 OR H04R27/00 OR H04R29/005 OR H04R5/027 OR G10L25/78 OR G10L19/09 OR G10L19/005 OR					

		G10L19/06 OR G10L2019/0011 OR G10L19/012 OR G10L19/002 OR G10L19/0212 OR G10L19/07 OR G10L19/083 OR G10L19/12 OR G10L19/22 OR G10L2019/0002 OR G10L2019/0016 OR G10L21/0208 OR G10L15/22 OR G10L2021/02087 OR G10L15/20 OR G10L21/0264 OR G10L21/02 OR G10L21/0364 OR G10L25/51 OR G10L15/00 OR G10L17/00 OR G10L19/00 OR G10L19/008 OR G10L2015/223 OR G10L15/30 OR G10L17/04 OR G10L21/0216 OR G10L21/0272 OR G10L15/26 OR G10L19/018 OR G10L2021/02082 OR G10L2021/02165 OR G10L2025/937 OR G10L25/84 OR G10L15/065 OR G10L19/04 OR G10L19/107 OR G10L19/26 OR G10L19/265 OR G10L2015/227 OR G10L2015/228 OR G10L2021/02166 OR G10L2021/02168 OR G10L21/0232 OR G10L25/03 OR G10L25/30 OR G10L13/027 OR G10L19/0204 OR G10L2015/088 OR G10L2025/932 OR G10L25/06 OR G10L25/21 OR G10L25/93 OR G10L15/005 OR G10L15/063 OR G10L15/16 OR G10L15/1815 OR G10L15/1822 OR					

		G10L17/08 OR G10L17/24 OR G10L19/032 OR G10L19/035 OR G10L19/167 OR G10L19/173 OR G10L19/24 OR G10L2015/025 OR G10L2019/0014 OR G10L2025/783 OR G10L21/00 OR G10L21/003 OR G10L21/0316 OR G10L21/0324 OR G10L21/0356 OR G10L21/10 OR G10L25/18 OR G10L25/24 OR G10L25/54 OR G10L25/57 OR G10L25/81 OR G10L15/01 OR G10L15/04 OR G10L15/08 OR G10L15/25 OR G10L17/18 OR G10L17/22 OR G10L19/0017 OR G10L19/20 OR G10L2025/786 OR G10L21/034 OR G10L25/60 OR H04M3/568 OR H04M9/08 OR H04M1/6008 OR H04M1/6066 OR H04M1/22 OR H04M1/72412 OR H04M1/82 OR H04M19/04 OR H04M19/048 OR H04M1/60 OR H04M7/0084 OR H04M9/082 OR H04M1/575 OR H04M1/724 OR H04M1/72469 OR H04M2201/41 OR H04M2203/6045 OR H04M2250/12 OR H04M3/567 OR H04M9/085 OR H04M1/0202 OR H04M1/03 OR H04M1/2535 OR H04M1/271 OR H04M1/6083 OR H04M2250/10 OR				

		H04M2250/52 OR H04M3/20 OR H04M3/428 OR H04M3/4285 OR H04M3/56 OR H04N5/23238 OR H04N21/4394 OR H04N5/2628 OR H04N21/44008 OR H04N5/2252 OR H04N5/23267 OR H04N21/8456 OR H04N5/23218 OR H04N13/111 OR H04N21/233 OR H04N21/43072 OR H04N21/4316 OR H04N21/47 OR H04N21/47205 OR H04N21/47217 OR H04N21/4884 OR H04N5/04 OR H04N5/23219 OR H04N5/23254 OR H04N5/76 OR H04N13/117 OR H04N13/359 OR H04N21/23418 OR H04N21/8547 OR H04N5/2251 OR H04N5/2254 OR H04N5/23216 OR H04N5/23232 OR H04N13/106 OR H04N13/239 OR H04N21/251 OR H04N21/42203 OR H04N21/435 OR H04N21/439 OR H04N21/44 OR H04N21/4415 OR H04N21/44227 OR H04N21/8549 OR H04N5/05 OR H04N5/10 OR H04N5/23222 OR H04N5/23258 OR H04N5/2327 OR H04N9/67 OR H04N9/802 OR H04N9/8205 OR H04N1/2133 OR H04N21/242 OR H04N21/43074 OR H04N21/8106 OR H04N21/84 OR H04N21/845 OR H04N21/854 OR					

		H04N5/2253 OR H04N5/23241 OR H04N5/23264 OR H04N5/232933 OR H04N5/23299 OR H04N5/783 OR H04N7/08 OR H04N7/0885 OR H04N9/806 OR H04N9/87 OR H04N13/229 OR H04N13/25 OR H04N13/344 OR H04N21/2335 OR H04N21/2662 OR H04N21/440281 OR H04N21/44209 OR H04N21/4666 OR H04N21/6587 OR H04N5/232 OR H04N5/23203 OR H04N5/247 OR H04N5/265 OR H04N5/272 OR H04L43/0829 OR H04L65/80 OR H04L12/66 OR H04L65/403 OR H04L1/0009 OR H04L1/0014 OR H04L1/007 OR H04L1/08 OR H04L43/00 OR H04L43/0847 OR H04L43/0852 OR H04L43/087 OR H04L43/16 OR H04L65/1083 OR H04L67/104 OR H04L69/10 OR H04L65/612 OR H04L45/74 OR H04L47/2441 OR H04L47/30 OR H04L47/34 OR H04L65/1069 OR H04L65/4015 OR H04L65/60 OR H04L65/75 OR H04L65/762 OR H04L67/01 OR H04L63/0428 OR H04W4/80 OR H04W72/1215 OR H04W72/1242 OR H04W88/06 OR H04W12/02 OR H04H20/28 OR					

		H04H20/423 OR H04H60/07).cpc.)					
L74	0	(third adj3 microphone$1) SAME ambient SAME background SAME noise SAME (reduction OR cancel$4) SAME mix$3 SAME audio SAME content	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/02/03 10:56 AM
L75	0	(third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3 SAME audio SAME content	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/02/03 10:56 AM
L76	0	(third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3 SAME content	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/02/03 10:57 AM
L77	3	(third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/02/03 10:57 AM
L78	3	((third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3).clm.	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)	OR	ON	ON	2023/02/03 02:56 PM
L79	92	"20110187640," "20100328224," "20060064037," "20030033152," "20050283369"	(US-PGPUB; USPAT; USOCR; FIT (AU, AP, AT, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DE, DK, EA, EE, EP, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV,	OR	ON	ON	2023/04/20 10:31 AM

			MA, OA, RU, SU, WO, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, SE, SG, SI, SK, TH, TN, TR, TW, UA, VN); FPRS; EPO; JPO; DERWENT; IBM_TDB)				
L80	67	"20110187640," "20100328224," "20060064037," "20030033152," "20050283369"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 10:31 AM
L81	72	"20110187640," "20100328224," "20060064037," "20030033152," "20050283369", "20160058378"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 10:32 AM
L82	264	"5002151," "7801318," "20070019817," "20050096899," "20060188105," "20010046304"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 10:40 AM
L83	436	"4533795," "6647368," "6671379," "6728385," "6754359," "7502484," "7903826"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 10:45 AM
L84	2037	"7953241," "7963433," "8014553," "8150064," "8351634," "9135797," "20010046304," "20030035551," "20040047486," "20060083388," "20060153394"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 10:50 AM
L85	172	"10709339," "10970375," "10142332," "8851372," "9112701," "9013351," "9684778"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 12:57 PM
L86	24	"20090085873," "20100316033," "20140089672," "20190038224"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 01:01 PM
L87	782	"7444353," "8018337," "6738482," "7003099," "8477955," "8270629," "7853031," "5550923"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 01:07 PM
L88	388	"6647368," "7512245," "8189803," "7936885," "5251263," "9113240," "7529379," "20120170412," "20080221880"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 02:42 PM

L89	82	"20060083387," "20050102142," "20080159547," "20050227674," "20040133421," "20020193130," "20050071158," "20070092087," "20040202340," "20050281422," "20070223717"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 02:54 PM
L90	23	"20030035551," "20080063228," "20070100637," "20070253569," "20070160243," "20080130908," "20070014423"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2023/04/20 03:01 PM

PE2E SEARCH - Search History (Interference)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
N1	0	((third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3 SAME content)	(US-PGPUB; USPAT)	OR	ON	ON	2023/02/03 02:51 PM
N2	0	((third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3 SAME content).clm.	(US-PGPUB; USPAT)	OR	ON	ON	2023/02/03 02:51 PM
N3	1	((third adj3 microphone$1) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3).clm.	(US-PGPUB; USPAT)	OR	ON	ON	2023/02/03 02:51 PM
N4	1	((third adj3 microphone) SAME (ambient OR background) SAME noise SAME (reduction OR cancel$4) SAME mix$3).clm.	(US-PGPUB; USPAT)	OR	ON	ON	2023/04/20 03:31 PM
N5	0	((third adj3 microphone) SAME (ambient OR background) SAME	(US-PGPUB; USPAT)	OR	ON	ON	2023/04/20 03:33 PM

		noise SAME (reduction OR cancel$4) SAME mix$3 SAME content).clm.					

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(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
/L.S/	1	10709339		2020-07-14	Hugh Lusted	
/L.S/	2	10970375		2021-04-06	Devu Manikantan	
/L.S/	3	10142332		2015-01-05	Sourabh Ravindran	
/L.S/	4	8851372		2014-10-07	Tiger T G Zhou	
/L.S/	5	9112701		2015-08-18	Akane Sano	
/L.S/	6	9013351		2015-04-21	James Park	
/L.S/	7	9684778		2017-06-20	Francis M. Tharappel	

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/L.S/	1	20090085873		2009-04-02	Betts; William L.	
/L.S/	2	20100316033		2010-12-16	Atwal; Peter	
/L.S/	3	20140089672		2014-03-27	Michael Edward Smith Luna	
/L.S/	4	20190038224		2019-02-07	Zhaonian Zhang	

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EXAMINER SIGNATURE			
Examiner Signature	/LUN-SEE LAO/	Date Considered	04/20/2023

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OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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	1	20080111004	KR	A	2008-12-22	SONY ERICSSON MOBILE COMM AB		☒
	2	2013501969	JP	A	2013-01-17	Georgiou et al.		☒
	3	TWM568011U	TW	U	2018-10-01	MOBILE DEVICES INC		☒

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

	4	WO2008077981	WO	A1	2008-07-03	Farzin Tahmassebi		☒
	5	ES2273616	ES	A1	2007-12-16	Farzin Tahmassebi		☒
	6	CN105554610	CN	B	2019-01-04	Beijing Xiaoniao Tingting Technology Co., Ltd.		☒
	7	EP2963647	EP	B1	2019-07-31	Harman International Industries, Incorporated		☐
	8	CN203761556	CN	U	2014-08-06	FUNG SHING COMPANY LTD		☒
	9	CN105637892	CN	A	2020-03-13	Bose Corp		☒
	10	JP6389232	JP	B2	2018-09-12	Silas Logic, Inc.		☒

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Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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	1	20110187640		2011-08-04	Jacobsen et al.	
	2	20100328224		2010-12-30	Kerr et al.	
	3	20060064037		2006-03-23	Shalon et al.	
	4	20030033152		2003-02-13	Cameron	
	5	20050283369		2005-12-22	Clauser et al.	

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	6	20160058378		2016-03-03	Wisby et al.			

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	1	WO9326085	WO	A1	1993-12-23	Hohman		X
	2	JP3353701	JP	B2	2002-12-03	Kondo		☐

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	1	90/015,146, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., REQUEST FOR EX PARTE REEXAMINATION OF U.S. PATENT NO. 10,979,836	

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	1	7444353		2008-10-28	Chen	
	2	8018337		2011-09-13	Jones	
	3	6738482		2004-09-23	Jaber	
	4	7003099		2006-02-21	Zhang	
	5	8477955		2013-07-02	Engle	
	6	8270629		2012-09-18	Bothra	
	7	7853031		2010-12-14	Hamacher	
	8	5550923		1996-08-27	Hotvet	

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	9	6647368		2003-11-11	Nemirovski	
	10	7512245		2009-03-31	Rasmussen	
	11	8189803		2012-05-29	Bergeron	
	12	7936885		2011-05-03	Frank	
	13	5251263		1993-10-05	Andrea	
	14	9113240		2015-08-18	Ramakrishman	
	15	7529379		2009-05-05	Zurek	

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	1	20120170412		2012-07-05	Calhoun	
	2	20080221880		2008-09-11	Cerra et al.	

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	3	20060083387		2006-04-20	Emoto	
	4	20050102142		2005-05-12	Soufflet	
	5	20080159547		2008-07-03	Schuler	
	6	20050227674		2005-10-13	Kopra	
	7	20040133421		2004-07-08	Burnett	
	8	20020193130		2002-12-19	Yang	
	9	20050071158		2005-03-31	Byford	
	10	20070092087		2007-04-26	Bothra	
	11	20040202340		2004-10-14	Armstrong	
	12	20050281422		2005-12-22	Armstrong	
	13	20070223717		2007-09-27	Boersma	

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	14	20030035551		2003-02-20	Light	
	15	20080063228		2008-03-13	Mejia	
	16	20070100637		2007-05-03	McCune	
	17	20070253569		2007-11-01	Bose	
	18	20070160243		2007-07-12	Dijkstra	
	19	20080130908		2008-06-05	Cohen	
	20	20070014423		2007-04-26	Darbut	

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CN105637892A Assisting conversation while listening to audio

Applicants: BOSE CORP

Inventors: BRIGGS DREW STONE,TAYLOR TRISTAN EDWARD

Classifications:
IPC ***H04R1/10;***

CPC **H04R1/1041 (EP,US); H04R1/1091 (US);** H04R1/1083 (EP,US); H04R2201/107 (EP,US); H04R2460/01 (EP,US);

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Assisting conversation while listening to audio

Abstract

A portable system for enhancing communication between users in proximity to each other while listening to a common audio source includes headsets with an electroacoustic transducer for providing sound to a respective user's ear, and a voice microphone for detecting sound of the respective user's voice and providing a microphone input signal, and an electronic device integral to the first headset and in communication with the second headset. The electronic device generates a side-tone signal based on the microphone input signal from the first headset, generates a voice output signal based on the microphone input signal from the first headset, receives a content input signal, combines the side-tone signal with the content input signal and a far-end voice signal associated with the second headset to generate a combined output signal, and provides the combined output signal to the first headset for output by the first headset's transducer.

CN105637892A Assisting conversation while listening to audio

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Assist in conversation while listening to audio

Technical field

The present disclosure relates to assisting in conversation while listening to music, and more particularly to allowing two or more earphone users in proximity to each other to listen to music or some other audio source while being able to easily speak and easily hear each other, thereby in audio content Dialogue naturally.

Background technique

It is very difficult to make a conversation while listening to some other audio source, such as discussing the performance while listening to the music performance at the same time. In particular, it is difficult for the speaker to hear his own voice, and the voice must be raised to a higher level than the comfort level that he can just hear, not to mention letting others hear them above the music. speak. The speaker may not be able to control how loud it is to allow others to hear what they are saying. Similarly, the person listening to music must try to listen to people and judge what they say. Even if the voice is improved, the comprehensibility and ease of listening are also affected. In addition, loud speech can interfere with other people in the neighborhood and reduce privacy.

In other contexts such as conversations in noisy environments, various solutions have been tried to solve these problems. Hearing aids for those with hearing impairment often have a pointing mode that attempts to amplify the speech of the person speaking to the user while rejecting the unwanted unwanted noise, but they are subject to limitations due to the microphone being located at the ear of the listening party. The resulting adverse signal to noise ratio. Moreover, hearing aids only provide the benefits of listening, and do not address the discomfort of trying to speak loudly in noisy, let alone coordinating with shared audio sources. Other communication systems, such as noise cancellation for pilots, headsets connected to walkie-talkies, may be very effective for their applications, but are limited to intercoms in front of the dashboard and are not suitable for social interaction by the average consumer. Or used in a mobile environment, or even not suitable for use in a flight environment - ie by commercial passengers.

Summary of the invention

In general, in one aspect, a portable system for enhancing communication between at least two users that are in close proximity to each other while listening to a common audio source includes: a first earphone and a second earphone, each earphone including An electroacoustic transducer that provides sound to the respective user's ear and a voice microphone that detects sound of the corresponding user's voice and provides a microphone input signal; and a first electronic device that is integrated with the first earphone and that communicates with the second earphone. The first electronic device generates a first side tone signal based on the microphone input signal from the first earphone, generates a first voice output signal based on the microphone input signal from the first earphone, receives the content input signal, and the first side tone signal and the content The input signal is combined with a first far-end speech signal associated with the second earphone to generate a first combined output signal, and the first combined output signal is provided to the first earphone for electro-acoustic transduction by the first earphone Output.

Embodiments may include one or more of the following in any combination. The first electronic device can scale the first side tone signal to control the level of the user's speech. The first electronic device can scale the first sidetone signal based at least in part on the detected sound level of ambient noise such that the user is unable to hear the sound level above the ambient noise without assistance Make a speech. The first electronic device can scale the first sidetone signal based at least in part on the detected sound level of ambient noise such that the user speaks at a level that may be masked by ambient noise. The first electronic device can scale the

first sidetone signal such that the user speaks at a level that is less than one meter from the user and that cannot be heard without assistance.

The first electronic device can be directly coupled to the second earphone, and the first electronic device can generate a second side tone signal based on the microphone input signal from the second earphone, and generate the first far end based on the microphone input signal from the second earphone a voice signal, combining the second side tone signal with the content input signal and the first voice output signal to generate a second combined output signal, and providing the second combined output signal to the second earphone for use by the second The electroacoustic transducer output of the earphone. The first electronic device can include the content input signal in the first by scaling the content input signal to be sufficiently lower than the first and second sidetone signals and the first and second far-end voice output signals at the sound level And the second combined output signal such that the sidetone signal and the far end speech signal remain understandable above the content signal. The step of scaling the content input signal may be performed only when one of the microphone input signals from at least one of the first or second earphones is above a threshold. The second electronic device can be integrated with the second earphone, the first electronic device communicates with the second earphone through the second electronic device, and the second electronic device can generate the second side tone signal based on the microphone input signal from the second earphone Generating a second voice output signal based on the microphone input signal from the second earphone, providing the second voice output signal to the first electronic device as the first far-end voice signal, and receiving the first voice output signal from the first electronic device Receiving a content input signal as a second far-end speech signal, combining the second flanking signal with the content input signal and the second far-end speech signal to generate a second combined output signal, and providing the second combined output signal To the second earphone for output by the electroacoustic transducer of the second earphone.

The first electronic device and the second electronic device can convert the content input signal to be sufficiently lower than the first and second sidetone signals and the first and second far-end voice output signals at the sound level The content input signal is included in the respective first and second combined output signals such that the sidetone signal and the far end speech signal remain understandable above the content signal. The step of scaling the content input signal may be performed by the first electronic device and the second electronic device when the microphone input signal from any of the first or second earphones may be above a threshold. The first earphone and the second earphone may each include a noise canceling circuit including a noise canceling microphone for providing an anti-noise signal to the corresponding electro-acoustic transducer based on the output of the noise canceling microphone, and An electronic device can provide a first combined output signal to the first earphone for output by the electroacoustic transducer of the first earphone in conjunction with the anti-noise signal provided by the noise canceling circuit of the first earphone. Both the first earphone and the second earphone may include a passive noise reduction structure. Generating the first side tone signal can include applying a frequency dependent gain to the microphone input signal from the first earphone. Generating the first sidetone signal can include filtering a microphone input signal from the first earpiece and applying a gain to the filtered signal. The first electronic device can include a source of the content input signal. The content input signal can be received wirelessly.

In general, in one aspect, an earphone includes: an electroacoustic transducer for providing sound to a user's ear; a voice microphone for detecting a voice of a user's voice and providing a microphone input signal; and an electronic device based on A microphone input signal from the earphone generates a sidetone signal, generates a voice output signal based on a microphone input signal from the earphone, receives a content input signal, receives a far end speech signal associated with another earphone, and the side tone signal The content input signal and the far-end speech signal are combined to generate a combined output signal, the combined output signal is output to the electro-acoustic transducer, and the speech output signal is output to another earphone.

Embodiments may include one or more of the following in any combination. The electronic device can scale the sidetone signal to control the level of the user's speech. The electronic device can scale the sidetone signal based at least in part on the detected sound level of ambient noise, thereby allowing the user to speak at a level that is unlikely to be heard above ambient noise without assistance. The electronic device can scale the sidetone signal based at least in part on the detected sound level of ambient noise such that the user speaks at a level that may be masked by ambient noise. The electronic device can scale the sidetone signal such that the user speaks at a level that is less than one meter from the user and that cannot be heard without assistance. The headset can include a source of the content input signal and can provide the content input signal to another headset. The electronic device can provide the content input signal to the additional earphone by combining the content input signal with the voice output signal. The electronic device can provide the content input signal to the additional earphone independently of outputting the voice output signal.

Advantages include allowing the user to discuss audio content such as music, movies, or other content that is shared without having to endeavor to listen to it over the content or above other background noise. Privacy is improved because users don't have to speak very loudly to be heard—so other people can hear them above the background noise. It also enables users to discuss shared audio content in a quiet environment without disturbing others or jeopardizing privacy, as they can speak softly without having to try to hear each other on top of shared content.

All of the examples and features mentioned above can be combined in any technically possible manner. Other features and advantages will be apparent from the description and claims.

DRAWINGS

Figures 1 and 2 show the configuration of the headset and electronic device used in the conversation.

Figure 3 shows the circuitry for implementing the apparatus of Figures 1 and 2.

Detailed ways

The system described herein allows two or more users to listen to a common audio source while conducting a conversation, such as recorded or streamed music or music from a movie, to name a few. Although the intent is that the conversation is related to music, the user may of course discuss anything they want to discuss. The goal of the system is to allow users to engage in their conversations without having to speak, hear each other or music, and be understood. We mentioned music, but obviously any audio content can be used. U.S. Patent Application Ser. This application describes a method for gaining by one or more of an outgoing speech signal and an incoming far-end speech signal for a filter and for a sidetone signal and for each of two or more earphone users. A portable system that manages and generally assists in conversation. Figures 1 and 2 are two users of the headsets 102 and 104 that have been reproduced from this application and which show conversations in progress. In Figure 1, two earphones are connected to a common electronic device 106, while in Figure 2, each earphone is connected to its own associated electronic device 108 or 110. In general, the electronic device can be integrated with the headset - embedded in the earbuds or inline with the cable. Alternatively, the electronic device can be a separate device, such as a mobile phone. Each earphone includes a microphone 105, which may be in the cable as shown, integrated into one or both earplugs, or on a long rod microphone that is supported from one ear.

Figure 3 illustrates additional features added to this application by Krisch. Each of the combined electronic and acoustic systems 202, 204 includes a voice microphone 206, a sidetone gain stage 208, a voice output gain stage 210, an attenuation module 212, and a summing node 214. The voice microphones detect their user's voice as voice audio inputs V1 and V2 and provide a microphone input signal 207. The microphone 206 also detects the ambient noises N1 and N2 and sends them to the gain stage, which is filtered according to the noise suppression capability of the microphone. The microphone is more sensitive to speech input than ambient noise due to noise suppression ratio M, so the microphone input signals are represented as V1+N1/M and V2+N2/M. Within those signals, N1/M and N2/M represent unwanted background noise. The different ambient noise signals N1 and N2 are shown entering the two systems, but depending on the distance between the user and the acoustic environment, these noises may be identical in effect. It is also possible that the ambient noise N3 and N4 at the user's ear, which is the same as N1 or N2, is attenuated by the attenuation module 212 in each circuit, and the attenuation module 212 represents the combined passive and active noise reduction capabilities of the earphone - if any if. Residual noise is shown as entering the output summing node, but in a practical implementation, the electronic signals are first summed and output by the output transducer, and the output of the transducer is in the user's ear canal with residual noise in acoustics Combine on. That is, the output node represents an output transducer that is combined with its acoustic environment. Out1 and Out2 represent the total audio output of the system, including the attenuated ambient noise.

The sidetone gain stage 208 applies a filter and gain to the microphone input signal and changes the shape and sound level of the voice signal to optimize it for use as the sidetone signal 209. When a person is unable to hear his own voice, such as while listening to other sounds, it will tend to speak more loudly. This has the effect that the speaker's voice becomes vocal. On the other hand, if a person is wearing a noise-isolated or noise-cancelling earphone, it will tend to speak at a comfortable, quieter sound level, but it will also be affected by the occlusion effect, which hinders nature, Comfortable speech. This occlusion effect occurs when the ear canal resonance and bone conduction lead to distortion and low frequency amplification, and makes the human voice sound

unnatural to themselves. The sidetone signal is a signal that is played back to the speaker's ear so that the speaker can hear his or her own voice. If the sidetone signal is properly scaled, the speaker will intuitively control the level of his voice to a comfortable level and be able to speak naturally. The sidetone filter within the gain stage 208 shapes the speech signal to compensate for the manner in which the occlusion effect changes its sound when the speaker's ear is plugged, such that the sidetone signal is in addition to the appropriate sound level. Make the user sound like the actual sound when they are not wearing headphones. We represent the sidetone filter as part of the frequency dependent sidetone gain Gs.

The microphone input signal 207 is also equalized and scaled by the speech output gain stage 210, applying a frequency dependent speech output gain G0 that combines speech output filtering. The speech output filtering and gain are selected such that the speech signal from the microphone of one earphone can be heard and understood by the user of the second earphone. Both the filtered and scaled speech output signals 211 are sent to additional headphones where they are combined with the filtered and scaled sidetone signal 209 and residual ambient noise in each earphone to produce a combined audio output Out1 or Out2. When discussing a headset, we can refer to the speech output signal 211 from another headset played by the earphone in question as the far-end speech signal. In some examples, instead of or in addition to filtering or amplifying the speech output signal, the incoming far-end speech signal can be filtered and amplified in each earphone.

In order to allow the user of the headset to hear and discuss the common audio signal, the side channel provides additional audio content C to the earphone. Gain stage 218 applies a frequency dependent gain Gc to content C from content source 216, which provides content input signal 220 and adds an additional item GcC to each audio output. As with other gain stages, the gain Gc can be frequency dependent in particular, or the input path can include a filter to shape the audio signal C in conjunction with the application of a gradual gain. The content can be received or generated by one of the headsets and transmitted to another headset, or the content can be received independently at both headsets. If the content is received at one earphone and transmitted to another earphone, the gain Gc can be applied to the two earphones at the earphone where the transmission is made, or it can be applied to the content signal received at each earphone, This allows for the changes and customizations shown in Krisch's application. The gain(s) Gc are designed with the speech signal and speech gain in mind, allowing the content to be heard in a level that does not mask the speech signal—distal and sidetones—to enable audio content I heard the voice on it. Providing a single content input signal to both headsets allows two users to listen to the same content while still being able to talk to each other. This can, for example, allow two users to share the same piece of music and discuss the music among them, with various gains allowing them to hear themselves and each other above the music. The gain can be adjusted automatically so that the music is attenuated to avoid masking the speech while any user is speaking, but returns to the normal listening level when no one is speaking. Figure 3 shows content source 216 outside of two electronic circuits 202 and 204. In some examples, the content source can be integrated into one of the circuits, or in an electronic device that encloses one of the circuits, and the content input signal 220 is coupled via a second electronic device from the first electronic device to the closed second circuit The output of the input of the two electronic devices is provided to other circuits.

In some examples, the user may want to talk softly, relying on the communication system to transmit its voice to the companion of the conversation at the appropriate level. In this case, the sidetone signal can be amplified, allowing the user to hear his voice at a normal speech level despite a soft voice. In order to conduct a conversation completely privately in a quiet environment, the side tone level can be set such that the user's voice can be detected by the microphone, but cannot be heard by an unassisted person one meter away. As discussed below, the precise sound level used will also be based on the sound level of the audio input such that the combined effect of the audio level and the side sound level results in the desired speech speech level. In a noisy environment, the user may need to speak at a louder sound level to be detected by the microphone so that the sidetone number is again properly scaled so that the combination of the sidetone level and the audio content level results in the user's dialogue to the dialog system. Providing a sufficient signal level to speak, but does not cause the user to try to be heard above the background noise. This has the added advantage that the user does not have to speak very loudly so that other users in the vicinity can also hear the conversation above the background noise, since the background noise will block the speech level, but the speech level can still be The microphone is detected.

To enhance the conversation, Kirsch's application assumes that the headset is attenuating, and if it is not sourced, it is at least passively attenuated. For comparison, for music sharing, it may be desirable for the headset to be un-attenuated or open. Open headphones provide minimal source of attenuation for ambient sound. In a quiet environment, some people think this improves the quality of music playback. When the present invention is employed with open headphones, various filters and gains will be changed. In particular, the user may not

need a sidetone signal at all because his own voice can naturally enter his or her ear and the ear canal is not blocked, so that there is no occlusion effect. However, the masking effect of audio content C still exists, so that a certain amount of sidetones may be desired to allow the user to speak at the appropriate level of sound above the audio content. This sidetone is also useful for controlling the user's speech relative to the sound level of any background noise. The speech output/distal speech signal gain is also modified to take into account the different acoustic effects of the open headphones. In general, the goals are still the same – allowing users to hear each other without having to go to the phone or trying to listen, while still listening to the audio content at a pleasant level.

In either case for attenuating or open headphones, the content gain Gc is chosen such that the audio content C is loud enough for the two users to enjoy while not being too loud so that other gains need to be increased to People's uncomfortable sound level in order to allow dialogue. This will usually be a lower level of sound than that used for simple audio playback. In some examples, the gain Gc is triggered by the user's speech and automatically switches between at least two levels, one for the conversation and the other for listening. Therefore, the content will "ducked" while the user is speaking but will not be completely muted and will return to its normal level after they stop speaking. In general, it will be desirable to make the whisper very fast, but the gain will rise more gently back to the listening level so that it does not rise and fall in every gap in the conversation.

Another application of the system described herein is to provide a dialogue channel between participants of a silent disco. In a silent disco, a large number of participants listen to the published audio signal through a personal wireless listening device, such as a wireless headset or a headset connected to the mobile phone. The system described herein can use a silent disco audio feed as the audio content source 216 while allowing subsets of participants to be connected to each other for dialogue in parallel with the music being shared.

Embodiments of the systems and methods described above include the steps of computer components and computer implementations that will be apparent to those skilled in the art. For example, those skilled in the art will appreciate that the steps embodied by the computer may be stored as computer executable instructions on a computer readable medium such as, for example, a floppy disk, a hard disk, an optical disk, a flash ROM, or a non- Volatile ROM and RAM. Moreover, it will be understood by those skilled in the art that the computer-executable instructions can be executed on various processors, such as a microprocessor, digital signal processor, gate array, etc. as an example. For ease of explanation, not every step or component of the systems and methods described above is described as being part of a computing system, but those skilled in the art will recognize that each step or component can have a corresponding computer system or Software component. Such computer systems and/or software can thus be implemented by describing their corresponding steps or components, that is, their function, and are within the scope of the present disclosure.

Various embodiments have been described. However, it will be understood that additional modifications may be made without departing from the scope of the inventions described herein, and other embodiments are therefore within the scope of the following claims.

CN105637892A Assisting conversation while listening to audio

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

1.A portable system for enhancing communication between at least two users that are in close proximity to each other while listening to a common audio source, including:

The first earphone and the second earphone, each earphone includes:

An electroacoustic transducer for providing sound to a corresponding user's ear, and

a voice microphone for detecting the voice of the corresponding user and providing a microphone input signal;

a first electronic device integrated with the first earphone and communicating with the second earphone, the first electronic device configured to:

Generating a first side tone signal based on the microphone input signal from the first earphone,

Generating a first voice output signal based on the microphone input signal from the first earphone,

Receiving a content input signal,

Combining the first side tone signal with the content input signal and a first far end speech signal associated with the second headset to generate a first combined output signal, and

The first combined output signal is provided to the first earphone for output by an electroacoustic transducer of the first earphone.

2.The system of claim 1 wherein said first electronic device scales said first side tone signal to control a sound level of said user speech.

3.The system of claim 2 wherein said first electronic device scales said first sidetone signal based at least in part on a detected sound level of ambient noise such that said user is in the absence of assistance It is impossible to speak at a sound level that is heard above the ambient noise.

4.The system of claim 2 wherein said first electronic device scales said first sidetone signal based at least in part on a detected sound level of ambient noise such that said user is likely to be shielded by ambient noise The voice level speaks.

5.The system of claim 2 wherein said first electronic device scales said first sidetone signal such that said user is in the absence of assistance at a distance greater than one meter from said user It is impossible to speak at the level of the sound that is heard.

6.The system of claim 1 wherein said first electronic device is directly coupled to said second earphone, and said first electronic device is further configured to:

Generating a second side tone signal based on the microphone input signal from the second earphone,

Generating the first far end speech signal based on the microphone input signal from the second earphone,

Combining the second sidetone signal with the content input signal and the first voice output signal to generate a second combined output signal, and

The second combined output signal is provided to the second earphone for output by an electroacoustic transducer of the second earphone.

7.The system of claim 6 wherein said first electronic device scales said content input signal to be sufficiently lower than said first and second sidetone signals and said first and second at a sound level A far end speech output signal is included to include the content input signal in the first and second combined output signals such that the sidetone signal and the far end speech signal remain understandable above the content signal.

8.The system of claim 7 wherein the step of scaling the content input signal is only when one of the microphone input signals from at least one of the first earphone or the second earphone is above a threshold Executed.

9.The system of claim 1 further comprising a second electronic device integrated with said second earphone,

Wherein the first electronic device communicates with the second earphone through the second electronic device, and

The second electronic device is configured to:

Generating a second side tone signal based on the microphone input signal from the second earphone,

Generating a second voice output signal based on the microphone input signal from the second earphone,

Providing the second voice output signal to the first electronic device as the first far-end voice signal,

Receiving, by the first electronic device, the first voice output signal as a second far-end voice signal,

Receiving the content input signal,

Combining the second side tone signal with the content input signal and the second far end speech signal to generate a second combined output signal, and

The second combined output signal is provided to the second earphone for output by an electroacoustic transducer of the second earphone.

10.The system of claim 9 wherein said first electronic device and said second electronic device both scale said content input signal to be sufficiently lower than said first and second sidetone signals and at said sound level Determining the first and second far-end speech output signals to include the content input signal in respective first and second combined output signals such that the sidetone signal and the far-end speech signal are in the content signal Keep it understandable.

11.The system of claim 10 wherein the step of scaling the content input signal is performed by the first electronic device and the second electronic device in either of the first earphone or the second earphone Executed when the microphone input signal is above a threshold.

12.The system of claim 1 wherein said first earphone and said second earphone each comprise a noise cancellation circuit, said noise cancellation circuit comprising a noise cancellation microphone, said noise cancellation microphone for canceling a microphone based on said noise The output provides an anti-noise signal to the corresponding electroacoustic transducer, and

The first electronic device is configured to provide the first combined output signal to the first earphone for combining with a noise canceling circuit of the first earphone by an electroacoustic transducer of the first earphone The anti-noise signals provided are output together.

13.The system of claim 1 wherein said first earphone and said second earphone each comprise a passive noise reduction structure.

14.The system of claim 1 wherein generating the first sidetone signal comprises applying a frequency dependent gain to the microphone input signal from the first earphone.

15.The system of claim 1 wherein generating the first sidetone signal comprises filtering the microphone input signal from the first earphone and applying a gain to the filtered signal.

16.The system of claim 1 wherein said first electronic device further comprises a source of said content input signal.

17.The system of claim 1 wherein said content input signal is received wirelessly.

18.A headset that includes:

An electroacoustic transducer for providing sound to a user's ear;

a voice microphone for detecting the sound of the user's voice and providing a microphone input signal;

Electronic equipment, configured to:

Generating a side tone signal based on the microphone input signal from the earphone,

Generating a voice output signal based on the microphone input signal from the earphone,

Receiving a content input signal,

Receiving a far end speech signal associated with another headset,

Combining the sidetone signal with the content input signal and the far end speech signal to generate a combined output signal,

Outputting the combined output signal to the electro-acoustic transducer, and

The voice output signal is output to the additional earphone.

19.The earphone of claim 18, wherein said electronic device scales said sidetone signal to control a sound level of a user speaking.

20.The earphone of claim 19 wherein said electronic device scales said sidetone signal based at least in part on a detected sound level of ambient noise such that said user is unlikely to be in the absence of assistance The level of sound heard above the surrounding noise is spoken.

21.The earphone according to claim 19, wherein said electronic device scales said sidetone signal based at least in part on a sound level of the detected ambient noise, thereby causing said user to speak at a sound level that may be shielded by ambient noise .

22.The earphone according to claim 19, wherein said electronic device scales said sidetone signal such that said user cannot be heard without assistance at a distance greater than one meter from said user Speak to the sound level.

23.The headset of claim 18, wherein the headset comprises a source of the content input signal, and wherein the electronic device is configured to provide the content input signal to the additional headset.

24.The earphone according to claim 23, wherein said electronic device provides said content input signal to said additional earphone by combining said content input signal with said voice output signal.

25.The earphone of claim 23, wherein the electronic device provides the content input signal to the additional earphone independently of outputting the voice output signal.

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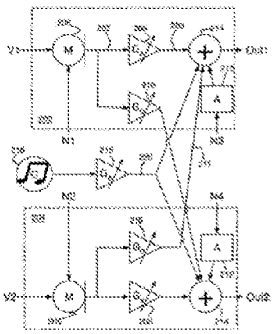
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(54)发明名称

在收听音频的同时辅助对话

(57)摘要

一种用于在收听共同音频源的同时提升互相接近的用户之间的通信的便携式系统，包括具有用于向相应用户的耳朵提供声音的电声换能器的耳机，和用于检测相应用户的语音的声音并且提供麦克风输入信号的语音麦克风，以及与第一耳机整合并且与第二耳机通信的电子设备。该电子设备基于来自第一耳机的麦克风输入信号生成侧音信号，基于来自第一耳机的麦克风输入信号生成话音输出信号，接收内容输入信号，将该侧音信号与该内容输入信号和与第二耳机相关联的远端语音信号进行组合以生成组合输出信号，并且将该组合输出信号提供至第一耳机以便由该第一耳机的换能器输出。



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1.一种用于在收听共同音频源的同时提升互相接近的至少两个用户之间的通信的便携式系统，包括：

第一耳机和第二耳机，每个耳机包括：

用于向相应用户的耳朵提供声音的电声换能器，和

用于检测所述相应用户的语音并且提供麦克风输入信号的语音麦克风；以及

与所述第一耳机整合并且与所述第二耳机通信的第一电子设备，所述第一电子设备被配置为：

基于来自所述第一耳机的所述麦克风输入信号，生成第一侧音信号，

基于来自所述第一耳机的所述麦克风输入信号，生成第一语音输出信号，

接收内容输入信号，

将所述第一侧音信号与所述内容输入信号和与所述第二耳机相关联的第一远端语音信号进行组合以生成第一组合输出信号，以及

将所述第一组合输出信号提供至所述第一耳机以用于由所述第一耳机的电声换能器输出。

2.根据权利要求1所述的系统，其中所述第一电子设备对所述第一侧音信号进行缩放以控制所述用户讲话的声级。

3.根据权利要求2所述的系统，其中所述第一电子设备至少部分基于检测到的周围噪声的声级来对所述第一侧音信号进行缩放，从而使得所述用户以在没有辅助的情况下不可能在所述周围噪声之上被听到的声级进行讲话。

4.根据权利要求2所述的系统，其中所述第一电子设备至少部分基于检测到的周围噪声的声级来对所述第一侧音信号进行缩放，从而使得所述用户以可能被周围噪声屏蔽的声级进行讲话。

5.根据权利要求2所述的系统，其中所述第一电子设备对所述第一侧音信号进行缩放，从而使得所述用户以在距所述用户大于一米的距离处在没有辅助的情况下不可能被听到的声级进行讲话。

6.根据权利要求1所述的系统，其中所述第一电子设备直接耦合至所述第二耳机，并且所述第一电子设备进一步被配置为：

基于来自所述第二耳机的所述麦克风输入信号，生成第二侧音信号，

基于来自所述第二耳机的所述麦克风输入信号，生成所述第一远端语音信号，

将所述第二侧音信号与所述内容输入信号和所述第一语音输出信号进行组合以生成第二组合输出信号，以及

将所述第二组合输出信号提供至所述第二耳机以用于由所述第二耳机的电声换能器输出。

7.根据权利要求6所述的系统，其中所述第一电子设备通过将所述内容输入信号缩放为在声级上充分低于所述第一和第二侧音信号以及所述第一和第二远端语音输出信号、来将所述内容输入信号包括在所述第一和第二组合输出信号中，从而使得所述侧音信号和远端语音信号在所述内容信号之上保持可理解。

8.根据权利要求7所述的系统，其中缩放所述内容输入信号的步骤仅在来自所述第一耳机或所述第二耳机中的至少一个耳机的所述麦克风输入信号之一高于阈值时才被执行。

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9.根据权利要求1所述的系统，进一步包括与所述第二耳机整合的第二电子设备，

其中所述第一电子设备通过所述第二电子设备而与所述第二耳机通信，并且

所述第二电子设备被配置为：

基于来自所述第二耳机的所述麦克风输入信号，生成第二侧音信号，

基于来自所述第二耳机的所述麦克风输入信号，生成第二语音输出信号，

将所述第二语音输出信号提供至所述第一电子设备以作为所述第一远端语音信号，

从所述第一电子设备接收所述第一语音输出信号以作为第二远端语音信号，

接收所述内容输入信号，

将所述第二侧音信号与所述内容输入信号和所述第二远端语音信号进行组合以生成第二组合输出信号，以及

将所述第二组合输出信号提供至所述第二耳机以用于由所述第二耳机的电声换能器输出。

10.根据权利要求9所述的系统，其中所述第一电子设备和第二电子设备通过均将所述内容输入信号缩放为在声级上充分低于所述第一和第二侧音信号以及所述第一和第二远端语音输出信号、来将所述内容输入信号包括在相应的第一和第二组合输出信号中，从而使得所述侧音信号和远端语音信号在所述内容信号之上保持可理解。

11.根据权利要求10所述的系统，其中缩放所述内容输入信号的步骤由所述第一电子设备和所述第二电子设备在来自所述第一耳机或所述第二耳机中的任一个耳机的所述麦克风输入信号高于阈值时执行。

12.根据权利要求1所述的系统，其中所述第一耳机和所述第二耳机均包括噪声消除电路，所述噪声消除电路包括噪声消除麦克风，所述噪声消除麦克风用于基于所述噪声消除麦克风的输出而向相应的电声换能器提供抗噪信号，并且

所述第一电子设备被配置为将所述第一组合输出信号提供至所述第一耳机以用于由所述第一耳机的电声换能器结合所述第一耳机的噪声消除电路所提供的所述抗噪信号一起输出。

13.根据权利要求1所述的系统，其中所述第一耳机和所述第二耳机均包括无源降噪结构。

14.根据权利要求1所述的系统，其中生成所述第一侧音信号包括向来自所述第一耳机的所述麦克风输入信号应用依赖于频率的增益。

15.根据权利要求1所述的系统，其中生成所述第一侧音信号包括对来自所述第一耳机的所述麦克风输入信号进行滤波并且向经滤波的信号应用增益。

16.根据权利要求1所述的系统，其中所述第一电子设备进一步包括所述内容输入信号的源。

17.根据权利要求1所述的系统，其中所述内容输入信号被无线接收。

18.一种耳机，包括：

电声换能器，用于向用户的耳朵提供声音；

语音麦克风，用于检测所述用户的语音的声音并且提供麦克风输入信号；以及

电子设备，被配置为：

基于来自所述耳机的所述麦克风输入信号生成侧音信号，

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基于来自所述耳机的所述麦克风输入信号生成语音输出信号，

接收内容输入信号，

接收与另外的耳机相关联的远端语音信号，

将所述侧音信号与所述内容输入信号和所述远端语音信号进行组合以生成组合输出信号，

将所述组合输出信号输出至所述电声换能器，以及

将所述语音输出信号输出至所述另外的耳机。

19.根据权利要求18所述的耳机，其中所述电子设备对所述侧音信号进行缩放以控制用户讲话的声级。

20.根据权利要求19所述的耳机，其中所述电子设备至少部分基于检测到的周围噪声的声级对所述侧音信号进行缩放，从而使得所述用户以在没有辅助的情况下不可能在所述周围噪声之上被听到的声级进行讲话。

21.根据权利要求19所述的耳机，其中所述电子设备至少部分基于检测到的周围噪声的声级对所述侧音信号进行缩放，从而使得所述用户以可能被周围噪声屏蔽的声级进行讲话。

22.根据权利要求19所述的耳机，其中所述电子设备对所述侧音信号进行缩放，从而使得所述用户以在距所述用户大于一米的距离处在没有辅助的情况下不可能被听到的声级进行讲话。

23.根据权利要求18所述的耳机，其中所述耳机包括所述内容输入信号的源，并且其中所述电子设备被配置为将所述内容输入信号提供至所述另外的耳机。

24.根据权利要求23所述的耳机，其中所述电子设备通过将所述内容输入信号与所述语音输出信号进行组合来将所述内容输入信号提供至所述另外的耳机。

25.根据权利要求23所述的耳机，其中所述电子设备独立于输出所述语音输出信号而将所述内容输入信号提供至所述另外的耳机。

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在收听音频的同时辅助对话

技术领域

[0001] 本公开涉及在收听音乐的同时辅助对话，尤其涉及允许相互邻近的两个或更多耳机用户收听音乐或一些其它音频源，而同时能够容易地讲话并且容易地互相听到，从而在音频内容之上自然地进行对话。

背景技术

[0002] 在收听一些其它音频源的同时进行对话会非常困难，诸如在同时收听音乐演出时对该演出进行讨论。特别地，讲话的人难以听到其自己的语音，并且必须将语音提高到比自己刚好能够听到的舒适声级更高的程度，更不用说要让其他人在音乐之上听到他们所讲。讲话者可能还难以掌控要讲多大声才能允许其他人听到他们所讲。同样，收听音乐的人必须竭力地听人讲，并且判断出说了什么。即使语音有所提高，可理解性和收听的容易度也受到影响。此外，大声讲话会干扰到邻近的其他人并且减少了隐私性。

[0003] 在诸如在嘈杂环境中进行对话的其它情境中，已经尝试了各种解决方案来解决这些问题。用于听力受损的那些人的助听器经常具有指向模式，其尝试将向用户讲话的人的语音放大同时拒绝所不希望听到的噪声，但是它们受到由于麦克风位于收听方的耳部的局限所导致的不良信噪比的影响。而且，助听器仅提供了收听方面的益处，而并没有解决在嘈杂中竭力大声讲话的不适，更不用说与共享音频源进行协调。诸如供飞行员使用的噪声消除、连接对讲机的耳机之类的其它通信系统对于它们的应用而言可能非常有效，但是被局限于仪表盘前的对讲，而并不适用于由一般消费者在社交或移动环境中使用、或者甚至并不适用于在飞行环境中—即由商业乘客—使用。

发明内容

[0004] 总体上，在一个方面，一种用于在收听共同音频源的同时提升互相接近的至少两个用户之间的通信的便携式系统包括：第一耳机和第二耳机，每个耳机包括用于向相应用户的耳朵提供声音的电声换能器和用于检测相应用户的语音的声音并且提供麦克风输入信号的语音麦克风；以及与第一耳机整合并且第二耳机进行通信的第一电子设备。第一电子设备基于来自第一耳机的麦克风输入信号生成第一侧音信号，基于来自第一耳机的麦克风输入信号生成第一语音输出信号，接收内容输入信号，将第一侧音信号与该内容输入信号和与第二耳机相关联的第一远端语音信号进行组合以生成第一组合输出信号，以及将第一组合输出信号提供至第一耳机以用于由第一耳机的电声换能器输出。

[0005] 实施方式可以以任意组合形式包括以下的一种或多种。第一电子设备可以对第一侧音信号进行缩放以控制用户讲话的声级。第一电子设备可以至少部分基于检测到的周围噪声的声级来对该第一侧音信号进行缩放，从而使得用户以在没有辅助的情况下不可能在周围噪声之上被听到的声级进行讲话。第一电子设备可以至少部分基于检测到的周围噪声的声级来对第一侧音信号进行缩放，从而使得用户以可能被周围噪声屏蔽的声级进行讲话。第一电子设备可以对第一侧音信号进行缩放，从而使得用户以在距该用户大于一米的

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距离处在没有辅助的情况下不可能被听到的声级进行讲话。

[0006] 第一电子设备可以直接耦合至第二耳机，并且第一电子设备可以基于来自该第二耳机的麦克风输入信号生成第二侧音信号，基于来自该第二耳机的麦克风输入信号生成第一远端语音信号，将第二侧音信号与内容输入信号和第一语音输出信号进行组合以生成第二组合输出信号，以及将该第二组合输出信号提供至该第二耳机以用于由该第二耳机的电声换能器输出。第一电子设备可以通过将该内容输入信号缩放为在声级上充分低于第一和第二侧音信号以及第一和第二远端语音输出信号、来将该内容输入信号包括在第一和第二组合输出信号中，从而使得该侧音信号和远端语音信号在内容信号之上保持可理解。缩放内容输入信号的步骤可以仅在来自第一或第二耳机中的至少一个耳机的麦克风输入信号之一高于阈值时才被执行。第二电子设备可以与第二耳机整合，第一电子设备通过第二电子设备而与第二耳机进行通信，并且该第二电子设备可以基于来自第二耳机的麦克风输入信号生成第二侧音信号，基于来自该第二耳机的麦克风输入信号生成第二语音输出信号，将第二语音输出信号提供至第一电子设备以作为第一远端语音信号，从第一电子设备接收第一语音输出信号以作为第二远端语音信号，接收内容输入信号，将第二侧音信号与内容输入信号和第二远端语音信号进行组合以生成第二组合输出信号，并且将该第二组合输出信号提供至第二耳机以用于由该第二耳机的电声换能器输出。

[0007] 第一电子设备和第二电子设备可以通过均将内容输入信号缩放为在声级上充分低于该第一和第二侧音信号以及该第一和第二远端语音输出信号、来将该内容输入信号包括在相应的第一和第二组合输出信号中，从而使得该侧音信号和远端语音信号在内容信号之上保持可理解。缩放该内容输入信号的步骤可以由第一电子设备和第二电子设备在来自该第一或第二耳机中的任一个耳机的麦克风输入信号可能高于阈值时执行。第一耳机和第二耳机均可以包括噪声消除电路，噪声消除电路包括噪声消除麦克风，噪声消除麦克风用于基于该噪声消除麦克风的输出而向相应的电声换能器提供抗噪信号，并且第一电子设备可以将第一组合输出信号提供至该第一耳机以便由该第一耳机的电声换能器结合该第一耳机的噪声消除电路所提供的抗噪信号一起输出。第一耳机和第二耳机均可以包括无源降噪结构。生成该第一侧音信号可以包括向来自该第一耳机的麦克风输入信号应用依赖于频率的增益。生成该第一侧音信号可以包括对来自该第一耳机的麦克风输入信号进行滤波并且向经滤波的信号应用增益。第一电子设备可以包括该内容输入信号的源。该内容输入信号可以被无线接收。

[0008] 总体上，在一个方面，一种耳机包括：电声换能器，用于向用户的耳朵提供声音；语音麦克风，用于检测用户的语音的声音并且提供麦克风输入信号；以及电子设备，其基于来自该耳机的麦克风输入信号生成侧音信号，基于来自该耳机的麦克风输入信号生成语音输出信号，接收内容输入信号，接收与另外的耳机相关联的远端语音信号，将该侧音信号与该内容输入信号和该远端语音信号进行组合以生成组合输出信号，将该组合输出信号输出至该电声换能器，并且将该语音输出信号输出至另外的耳机。

[0009] 实施方式可以以任意组合形式包括以下的一种或多种。该电子设备可以对该侧音信号进行缩放以控制用户讲话的声级。该电子设备可以至少部分基于检测到的周围噪声的声级对该侧音信号进行缩放，从而使得用户以在没有辅助的情况下不可能在周围噪声之上被听到的声级进行讲话。该电子设备可以至少部分基于检测到的周围噪声的声级对该侧音

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信号进行缩放，从而使得用户以可能被周围噪声屏蔽的声级进行讲话。该电子设备可以对该侧音信号进行缩放，从而使得用户以在距该用户大于一米的距离处在没有辅助的情况下不可能被听到的声级进行讲话。该耳机可以包括该内容输入信号的源，并且可以将该内容输入信号提供至另外的耳机。该电子设备可以通过将该内容输入信号与该语音输出信号进行组合来将该内容输入信号提供至另外的耳机。该电子设备可以独立于输出该语音输出信号而将该内容输入信号提供至另外的耳机。

[0010] 优势包括允许用户讨论所共享的诸如音乐、电影或其它内容的音频内容而不必竭力去听在该内容之上或者在其它背景噪声之上所听到的。由于用户并不必讲话很大声以使被听到——这样其他人也能够在背景噪声之上而听到他们，所以隐私得以改善。还使得用户能够在安静的环境中讨论所共享的音频内容，而并不打扰到其他人或者危及隐私，因为他们能够轻声讲话而不必竭力在所共享的内容之上听到彼此。

[0011] 以上所提到的所有示例和特征能够以任意技术上可能的方式进行组合。其它特征和优势将由于描述和权利要求而是显而易见的。

附图说明

[0012] 图1和2示出了在对话中所使用的耳机和电子设备的配置。

[0013] 图3示出了用于实施图1和2的设备的电路。

具体实施方式

[0014] 这里所描述的系统允许两个或更多用户收听共同的音频源、同时进行对话，举几个例子，上述音频源诸如所录制或流传输的音乐或者来自电影的音乐。虽然意图在于该对话与音乐相关，但是用户当然可能讨论他们想要讨论的任何事情。该系统的目标是允许用户进行他们的对话而不必竭力讲话、听到彼此或音乐、以及被理解。我们提到了音乐，但是显然可以使用任意的音频内容。Kathy Krisch和Steve Isabelle的、标题为“Assisting Conversation”、律师事务所卷号为N-13-133-US的美国专利申请No._____同时被提交并且通过引用全文结合于此。该申请描述了一种用于通过对滤波器以及应用于侧音信号以及针对两个或更多耳机用户中的每一个的外出语音信号和到来远端语音信号中的一个或多个的增益进行管理而总体上辅助对话的便携式系统。图1和2是从该申请所再现的并且示出了正在进行对话的耳机102和104的两个用户。在图1中，两个耳机连接至共同的电子设备106，而在图2中，每个耳机连接至其自己的相关联电子设备108或110。总体上，该电子设备可以与耳机进行整合——嵌入在耳塞之中或者与线缆内联。可替换地，该电子设备可以是单独的设备，诸如移动电话。每个耳机包括麦克风105，其可以如所示出的处于线缆之中、被整合到一个或两个耳塞之中、或者处于从一个耳朵获得支撑的长杆话筒上。

[0015] 图3示出了本申请为Krisch的申请所增加的附加特征。每个经组合的电子和声学系统202、204包括语音麦克风206、侧音增益级208、语音输出增益级210、衰减模块212和加总节点214。语音麦克风检测他们的用户的语音以作为语音音频输入V1和V2，并且提供麦克风输入信号207。麦克风206还检测到周围噪声N1和N2并且将它们送至增益级，根据麦克风的噪声抑制能力而被滤波。麦克风由于噪声抑制比M而对语音输入比周围噪声更为敏感，因此麦克风输入信号被表示为V1+N1/M和V2+N2/M。在那些信号内，N1/M和N2/M表示不想要的

背景噪声。不同的周围噪声信号N1和N2被示为进入两个系统，但是取决于用户与声学环境之间的距离，这些噪声在效果上可能是相同的。同样可能与N1或N2相同的、用户耳朵处的周围噪声N3和N4被每个电路中的衰减模块212进行衰减，衰减模块212表示耳机的经组合的无源和有源降噪能力—如果有的话。残余噪声被示为进入到输出加总节点，但是在实际实施方式中，电子信号首先被加总并且由输出换能器所输出，并且该换能器的输出在用户耳道内与残余噪声在声学上进行组合。也就是说，输出节点表示与其声学环境相结合的输出换能器。Out1和Out2表示该系统的总音频输出，包括经衰减的周围噪声。

[0016] 侧音增益级208对麦克风输入信号应用滤波器和增益并且改变语音信号的形状和声级从而对其进行优化以便用作侧音信号209。当人诸如在收听其它声音的同时而无法听到其自己的语音时，其将会趋向于更为大声地讲话。这样具有的影响是讲话者的语音变得声嘶力竭。另一方面，如果人正佩戴着噪声隔离或噪声消除耳机，则其将会趋向于以舒适的、更为安静的声级进行讲话，但是也将会受到闭塞效应的影响，这妨碍了自然、舒适的讲话。该闭塞效应在耳道共振和骨传导导致失真和低频放大时出现，并且使得人的语音他们自己听上去并不自然。侧音信号是针对讲话者的耳朵进行播放的信号，从而使得讲话者能够听到自己的语音。如果侧音信号被适当缩放，则讲话者将直观地将他的语音的声级控制在舒适的声级，并且能够自然地进行讲话。增益级208内的侧音滤波器对语音信号进行整形从而对该闭塞效应在讲话者的耳朵被塞住时改变其声音的方式作出补偿，从而使得除了处于适当声级之外，该侧音信号令用户听上去就像其没有佩戴耳机时的实际声音。我们将侧音滤波器表示为依赖于频率的侧音增益G_S的一部分。

[0017] 麦克风输入信号207也被语音输出增益级210进行均衡和缩放，应用结合了语音输出滤波的依赖于频率的语音输出增益G_O。该语音输出滤波和增益被选择为使得来自一个耳机的麦克风的语音信号能够被第二耳机的用户所听到并且可理解。经滤波器和缩放的语音输出信号211均被送至另外的耳机，它们在那里与每个耳机内经滤波和缩放的侧音信号209和残余周围噪音进行组合从而产生组合音频输出Out1或Out2。当讨论一个耳机时，我们可以将所考虑的耳机所播放的来自另外的耳机的语音输出信号211称之为远端语音信号。在一些示例中，取代对语音输出信号进行滤波和放大或者作为其的附加，到来的远端语音信号可以在每个耳机中进行滤波和放大。

[0018] 为了允许耳机的用户听到并讨论共同的音频信号，侧方通道向耳机提供了附加的音频内容C。增益级218向来自内容源216的内容C应用依赖于频率的增益G_C，这提供了内容输入信号220并且将附加项G_CC添加至每个音频输出。如利用其它增益级那样，增益G_C可以特别是依赖于频率的，或者输入路径可以包括滤波器以结合应用平缓增益来对音频信号C进行整形。该内容可以被耳机之一所接收或生成并且被传送至另外的耳机，或者该内容可以在两个耳机处被独立接收。如果该内容在一个耳机处被接收并且被传送至另外的耳机，则可以在进行传送的耳机处为两个耳机应用增益G_C，或者其可以被应用于在每个耳机处所接收的内容信号，这允许Krisch的申请中所示出的变化和定制。(多个)增益G_C在考虑语音信号和语音增益的情况下进行设计，从而允许内容以不屏蔽语音信号—远端和侧音的—的声级被听到，从而使得能够在音频内容之上听到语音。向两个耳机提供单个内容输入信号允许两个用户收听到相同的内容，同时还能够互相讲话。这例如能够允许两个用户分享同一段音乐，并且在他们之中对该音乐加以讨论，其中各种增益允许他们在该音乐之上听到自

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己和彼此。该增益可以自动调节，从而使得该音乐被衰减以避免在任一个用户正讲话时屏蔽了语音，但是当没有人讲话时返回至正常收听声级。图3示出了处于两个电子电路202和204之外的内容源216。在一些示例中，该内容源可以被整合到电路之一中、或者处于封闭这些电路之一的电子设备中，并且内容输入信号220经由来自第一电子设备的、耦合至封闭第二电路的第二电子设备的输入的输出，而被提供至其它电路。

[0019] 在一些示例中，用户可能想要轻声讲话，依赖于通信系统将其语音以适当声级传输至对话的同伴。在这种情况下，侧音信号可以被放大，从而使得用户尽管在轻声讲话的情况下也以正常的讲话声级听到其语音。为了在安静环境中完全私密地进行对话，侧音声级可以被设置为使得用户的语音能够被麦克风检测到、但是不可能被一米以外的未经辅助的人所听到。如以下所讨论的，所使用的精确声级还将基于音频输入的声级，从而音频声级和侧音声级的组合效应导致了所期望的讲话语音声级。在嘈杂环境中，用户可能需要以更为大声的声级进行讲话以便被麦克风所检测，从而侧音新号再次被适当缩放以使得侧音声级和音频内容声级的组合导致用户以向对话系统提供充分信号的声级进行讲话、但是并不会导致用户竭力要在背景噪声之上被听到。这样所具有的额外优势在于，用户不必很大声地进行讲话从而使得附近的其它用户也能够在背景噪声之上听到对话，因为背景噪声将会屏蔽讲话声级，但该讲话声级仍然能够被麦克风检测到。

[0020] 为了增强对话，Kirsch的申请假设耳机正进行衰减，如果不是有源地也至少是无源地衰减。作为比较，对于音乐共享，可能期望耳机是不衰减或是开放的。开放式耳机提供了对周围声音的最低程度的无源衰减。在安静环境中，某些人认为这改善了音乐播放的质量。当本发明随开放式耳机被采用时，将会对各种滤波器和增益作出改变。特别地，用户可能根本不需要侧音信号，因为他自己的语音能够自然地进入其耳朵，并且耳道并未被阻挡，从而并不存在闭塞效应。然而，音频内容C的屏蔽效果仍然存在，从而可能期望一定数量的侧音来允许用户以在音频内容之上的适当声级进行讲话。该侧音对于控制用户的语音相对于任何背景噪声的声级而言也仍然是有用的。语音输出/远端语音信号增益也被修改，从而考虑开放式耳机的不同声学效果。总体上，目标仍然是相同的——允许用户互相听到而并不必竭力去讲话或费劲去听，同时仍然以令人愉悦的声级听到音频内容。

[0021] 在针对衰减或开放式耳机的任一种情况下，内容增益G_c被选择为使得音频内容C足够大声以便由两个用户所享用，同时又不会过于大声从而使得需要将其它增益提高至令人不适的声级以便允许对话。这通常将是比简单音频播放所使用的更低的声级。在一些示例中，增益G_c由用户讲话所触发而在至少两个声级——一个用于对话而另一个则用于收听——之间自动切换。因此，内容将在用户正在讲话时“变小声(ducked)”但并不会完全静音，并且在他们停止讲话之后将会返回至它的正常声级。通常，将期望非常快速地进行变小声，但是增益将更为平缓地上升回到收听声级，从而其不会在对话中的每个间隙不断上升和下降。

[0022] 这里所描述的系统的另一种应用是在无声迪斯科(silent disco)的参与者之间提供对话通道。在无声迪斯科中，大量参与者通过个人无线收听设备、诸如无线头戴耳机或连接至移动电话的耳机来收听所发布的音频信号。这里所描述的系统可以使用无声迪斯科音频馈送作为音频内容源216，同时允许参与者的子集互相连接以便平行于所分享的音乐进行对话。

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[0023] 以上所描述的系统和方法的实施例包括对于本领域技术人员将清楚的计算机组件和计算机实施的步骤。例如，本领域技术人员将会理解的是，该计算机所实施的步骤可以作为计算机可执行指令存储在计算机可读介质上，上述计算机可读介质诸如举例为软盘、硬盘、光盘、闪存ROM、非易失性ROM和RAM。此外，本领域技术人员应当理解的是，该计算机可执行指令可以在各种处理器上执行，作为示例，上述处理器诸如微处理器、数字信号处理器、门阵列等。为了便于解释，并非以上所描述的系统和方法的每个步骤或部件都被描述为计算系统的一部分，但是本领域技术人员将会认识到，每个步骤或部件可以具有相对应的计算机系统或软件组件。这样的计算机系统和/或软件因此能够通过描述它们相对应的步骤或部件(也就是说它们的功能)而实现，并且处于本公开的范围之内。

[0024] 已经描述了多种实施方式。然而，将要理解的是，可以进行附加的修改而并不背离这里所描述的发明概念的范围，并且其它实施例因此处于以下权利要求的范围之内。

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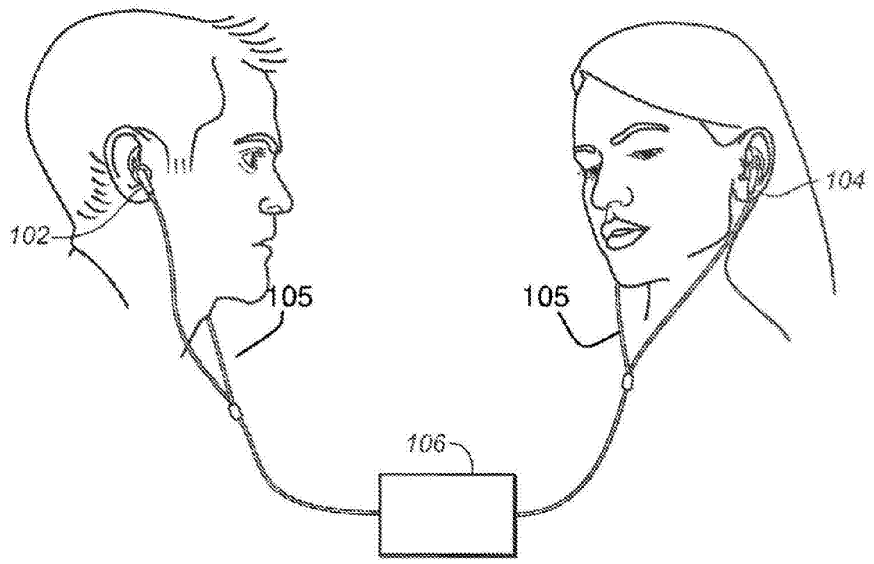


图1

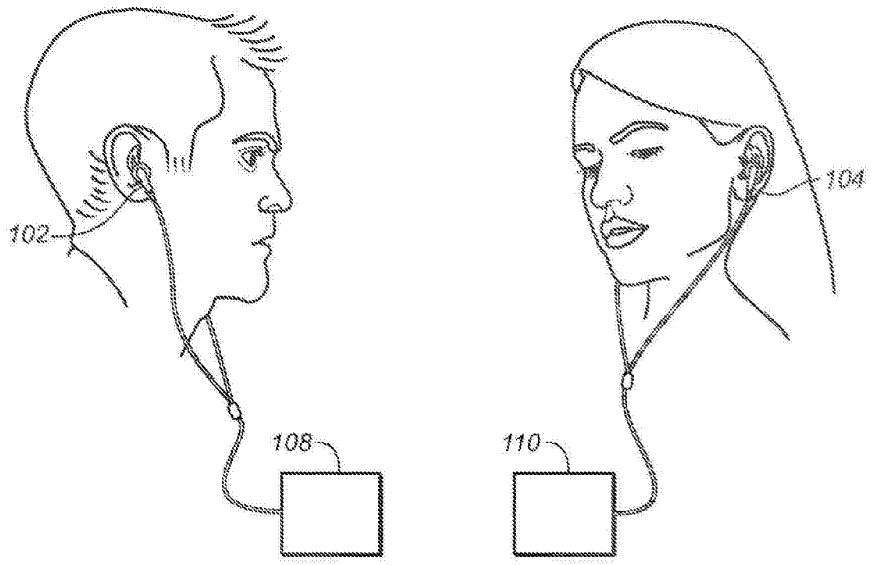


图2

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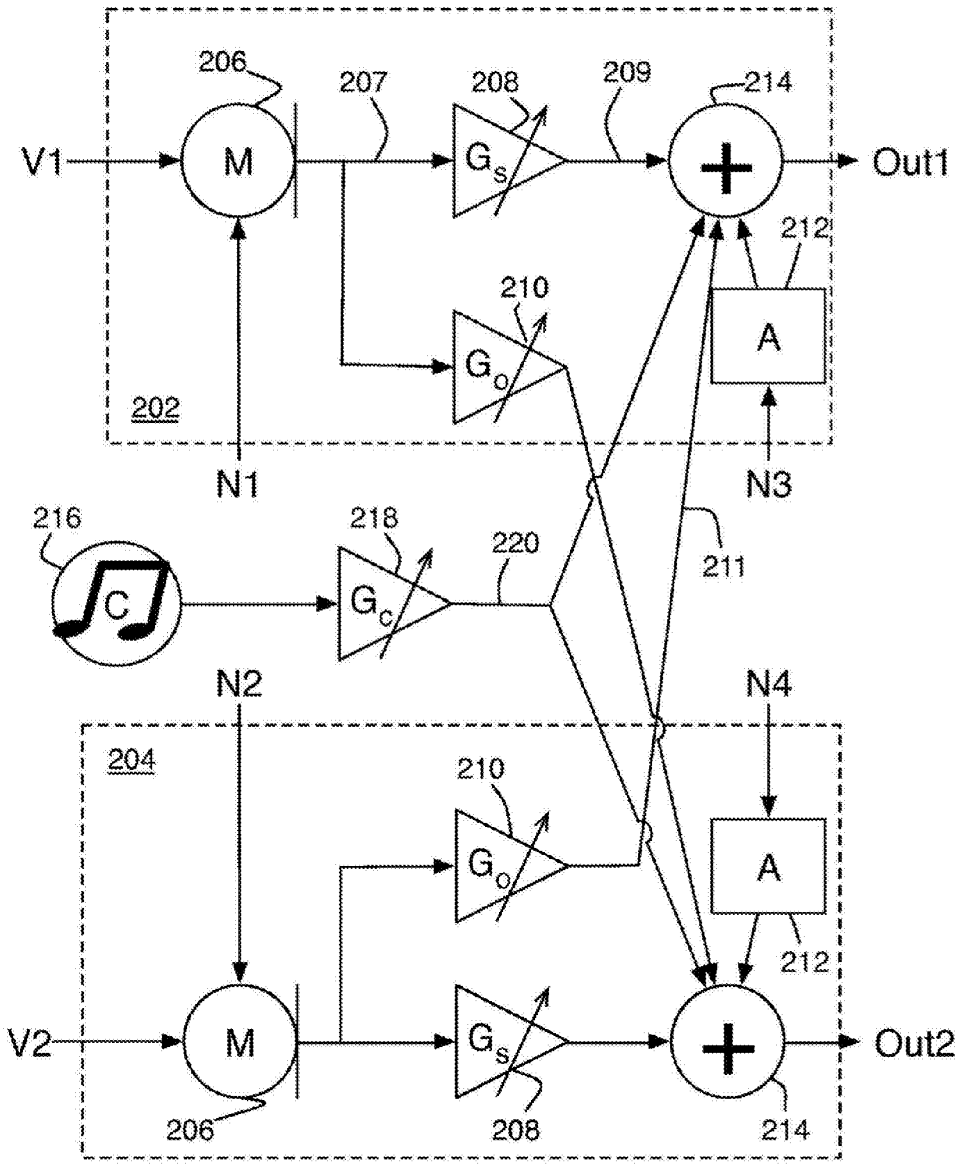

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211
C
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V2
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Out2
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208
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图3

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/215,804	CONFIRMATION NO. 9372
APPLICATION OF	:	STEVEN WAYNE GOLDSTEIN	
DATE FILED	:	03/29/2021	
EXAMINER	:	LAO, LUNSEE	
TC/A.U.	:	2651	
DOCKET	:	CONT2-16247186	
TITLE	:	Method and Device for in Ear Canal Echo Suppression	

Amendment After Allowance

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

Date: April 7, 2023

Sir:

The Office issued a Notice of Allowance on February 21, 2023. The following amendment to the claims is made with a Request for Continued Examination (RCE) prior to payment of the issue fee.

- **Amendments to the Claims** begin on page 2 of this paper; and
- **Remarks/Arguments** begin on page 6 of this paper.

AMENDMENTS TO THE CLAIMS

The following listing of claims will replace all prior versions, and listings, of claims in the application.

Listing of Claims:

1. (Currently Amended) ~~An earpiece configured to use multiple microphones for noise control~~ A device, comprising:

a first ambient microphone configured to generate a first ambient signal, wherein the first ambient microphone is part of an earpiece;

a second ambient microphone configured to generate a second ambient signal, wherein the second ambient microphone is part of the earpiece;

an ear canal microphone configured to generate an ear canal signal, wherein the ear canal microphone is part of the earpiece;

a speaker, wherein the speaker is part of the earpiece;

an eartip, configured to seal the earpiece between a first side and a second side of the earpiece;

a memory that stores instructions; and

a processor, wherein the processor is part of the earpiece, wherein the processor is operatively connected to the first ambient microphone, wherein the processor is operatively connected to the second ambient microphone, wherein the processor is operatively connected to the ear canal microphone, wherein the processor is operatively connected to the speaker, wherein the processor is operatively connected to the memory, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:

receiving a first microphone signal, wherein the first microphone signal is at least one of the first ambient signal or a modified first ambient signal;

receiving a second microphone signal, wherein the second microphone signal is at least one of the ear canal signal or a modified ear canal signal;

receiving a third microphone signal, wherein the third microphone signal is at least one of the second ambient signal or a modified second ambient signal;

generating a background noise reduction signal using either the first and second microphone signals or the second and third microphone signals;

mixing an audio content signal and the background noise reduction signal generating a mixed signal; and

sending a modified mixed signal to the speaker.

2. (Canceled).

3. (Currently Amended) The ~~earphone~~ device according to claim 1, wherein the operations further comprise:

generating an ambient signal from at least one of the first ambient signal or the second ambient signal or a combination of both.

4. (Currently Amended) The ~~earphone~~ device according to claim 3, wherein the operations further comprise:

sending a modified ambient signal to the speaker.

5. (Currently Amended) The ~~earphone~~ device according to claim 4, where the modified ambient signal is the ambient signal.

6. (Currently Amended) The ~~earphone~~ device according to claim 4, wherein the operations further comprise:

generating the modified ambient signal by filtering an ambient signal, wherein the ambient signal is generated from at least the first ambient sound signal or the second ambient sound signal.

7. (Canceled).

8. (Canceled).

9. (Currently Amended) The ~~earphone~~ device according to claim 4, wherein the operations further comprise:

generating a modified ambient signal by filtering the ambient signal and modifying an amplitude of at least one frequency of the ambient signal.

10. (Currently Amended) The ~~earphone~~ device according to claim 1, wherein the modified mixed signal is the mixed signal.

11. (Currently Amended) The ~~earphone~~ device according to claim 1, wherein the operations further comprise:

generating the modified mixed signal by mixing the mixed signal with an ambient signal, wherein the ambient signal is generated from at least one of the first ambient signal, the second ambient signal or a combination of both.

12.(Currently Amended) The ~~earphone~~ device according to claim 11, wherein the operations further comprise:

analyzing at least one of the ambient signal or the ear canal signal or a combination of both, to detect a user's voice.

13. (Currently Amended) The ~~earphone~~ device according to claim 12, wherein the operations further comprise:

reducing the audio content signal prior to generating the mixed signal if the user's voice is detected.

14. (Canceled).

15. (Currently Amended) The ~~earphone~~ device according to claim 12, wherein the operations further comprise:

reducing the ambient signal prior to generating the mixed signal if the user's voice is not detected.

REMARKS

The claims are amended herein with a Request for Continued Examination (RCE) to be consistent with Applicant's prior claim amendments submitted to the Office on January 6, 2023 in response to the October 25, 2022 Office Action. The amendment does not require an additional search, does not require more than a cursory review of the record, and do not involve materially added work on the part of the Office. No new matter is added.

Thus, please amend the patent application as set forth above and allow the patent application to proceed to issuance. Should the Examiner believe that anything further is necessary in order to place the application in better condition for allowance, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

No fees are believed due; however, the Commissioner is hereby authorized to charge any deficiency, or credit any overpayment, to Deposit Account No. 50-0951.

Respectfully submitted,

AKERMAN LLP

Dated: April 7, 2023

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(51) Int Cl.:
G10L 21/0364 *(2013.01)* ***H04S 7/00*** *(2006.01)*
H03G 9/00 *(2006.01)* ***H03G 9/02*** *(2006.01)*
H04R 1/10 *(2006.01)* ***H04R 3/00*** *(2006.01)*
G10L 25/84 *(2013.01)*

(54) APPROACH FOR PARTIALLY PRESERVING MUSIC IN THE PRESENCE OF INTELLIGIBLE SPEECH

ANSATZ ZUR TEILWEISEN KONSERVIERUNG VON MUSIK IN GEGENWART VON VERSTÄNDLICHER SPRACHE

APPROCHE POUR LA PRÉSERVATION PARTIELLE DE LA MUSIQUE EN PRÉSENCE DE PAROLES INTELLIGIBLES

(84) Designated Contracting States:
AL AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HR HU IE IS IT LI LT LU LV MC MK MT NL NO PL PT RO RS SE SI SK SM TR

(30) Priority: **09.06.2014 US 201462009851 P**

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(56) References cited:
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WO-A2-2014/022359
US-A1- 2007 189 544
US-A1- 2010 211 199

Note: Within nine months of the publication of the mention of the grant of the European patent in the European Patent Bulletin, any person may give notice to the European Patent Office of opposition to that patent, in accordance with the Implementing Regulations. Notice of opposition shall not be deemed to have been filed until the opposition fee has been paid. (Art. 99(1) European Patent Convention).

Printed by Jouve, 75001 PARIS (FR)

EP 2 963 647 B1

Description

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application claims the benefit of United States provisional patent application titled, "APPROACH FOR PARTIALLY PRESERVING MUSIC IN THE PRESENCE OF INTELLIGIBLE SPEECH," filed on June 9, 2014 and having Serial No. 62/009,851.

BACKGROUND

Field of the Embodiments of the Present Disclosure

[0002] Embodiments of the present disclosure relate generally to audio signal processing and, more specifically, to an approach for partially preserving music in the presence of intelligible speech.

Description of the Related Art

[0003] Headphones, earphones, earbuds, and other personal listening devices are commonly used by individuals who desire to listen to a particular type of audio source, such as music, speech, or movie soundtracks, without disturbing other people in the nearby vicinity. These types of audio sources are referred to herein as "entertainment" signals, and each is typically characterized as an audio signal that is present over a sustained period of time. Typically, these personal listening devices include an audio plug for insertion into an audio output of an audio playback device. The audio plug connects to a cable that carries the audio signal from the audio playback device to the personal listening device. In order to provide high quality audio, such personal listening devices include speaker components that typically cover the entire ear or completely seal the ear canal. The personal listening device is designed to provide a good acoustic seal, thereby reducing audio signal leakage and improving the quality of the listener's experience, particularly with respect to bass response.

[0004] Document US 2007/0189544 A1 discloses a method of adjusting an output of a media player comprising capturing an ambient audio signal, processing the ambient audio signal to determine whether one or more characteristic forms are present within the ambient audio signal, and reducing an output of a media player from a first volume to a second volume of the one or more characteristic forms are present within the ambient audio signal. The characteristic forms may be, for example, a name or personal identifier of a user of the media player, the voice of a user of the media player, or an alarm or siren.

[0005] Document WO 2011/112382 A1 discloses a method and system for filtering a multi-channel audio signal having a speech channel and at least one non-speech channel, to improve intelligibility of speech determined by the signal. In typical embodiments, the method includes steps of determining at least one attenuation control value indicative of a measure of similarity between speech-related content determined by the speech channel and speech-related content determined by the non-speech channel, and attenuating the non-speech channel in response to the at least one attenuation control value. Typically, the attenuating step includes scaling of a raw attenuation control signal (e.g. ducking gain control signal) for the non-speech channel in response to the at least one attenuation control value.

[0006] Document WO 2014/022359 A2 discloses an earphone system and method for automatically directing ambient sound to an earphone device. An ambient microphone signal from an ambient microphone proximate a sound isolating earphone or headset device is directed to a receiver within an earphone device according to mixing circuitry. The mixing circuitry is controlled by voice activity of the earphone device wearer. This enables hands-free operation of an earphone system to allow the earphone device wearer to maintain situation awareness with the surrounding environment. During detected voice activity, incoming audio content is attenuated while ambient sound is increased and provided to the earphone device. User voice activity is detected by analysis of at least one of an ear canal microphone signal or an ambient sound microphone signal.

[0007] One drawback of the personal listening devices described above is that, because the devices form a good acoustic seal with the ear, the ability of the listener to hear environmental sound is substantially reduced. As a result, the listener may be unable to hear certain important sounds from the environment, such as the sound of an oncoming vehicle, human speech, or an alarm. These types of important sounds emanating from the environment are referred to herein as "priority" or "alert" signals, and each is typically characterized as an audio signal that is intermittent, acting as an interruption.

[0008] One solution to the above problem is to acoustically or electronically mix or combine environmental sound, including any alert signal present in the environmental sound, detected by one or more microphones with the entertainment signal. The listener is then able to hear both the entertainment signal and the environmental sound, including any alert signal. One drawback with such solutions, though, is that the alert signal may be unintelligible (or may even be inaudible) if the entertainment signal is too loud relative to the alert signal.

[0009] As the foregoing illustrates, more effective techniques for providing entertainment signals and alert signals to users of listening devices would be useful.

SUMMARY

[0010] One or more embodiments set forth an audio processing system for a listening device that includes an input device, a voice activity detector and a ratio-based attenuator. The input device is configured to receive a first audio signal emanating from an environment and

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including a signal of interest. The voice activity detector is configured to receive a first control signal from a voice separator, determine that the signal of interest is present in the first audio signal by determining that the first control signal exceeds a ducker threshold, and generate a second control signal in response to the first audio signal. The ratio-based attenuator is configured to receive the second control signal from the voice activity detector. The ratio-based attenuator is further configured to determine whether a first signal level associated with the first audio signal exceeds a second signal level associated with a second audio signal received from an audio playback device by at least a target difference. The ratio-based attenuator is further configured to, if the first signal level exceeds the second signal level by at least a target difference, then maintain an audio level of the second audio signal. The ratio-based attenuator is further configured to if the first signal level does not exceed the second signal level by at least the target difference, then cause the audio level of the second audio signal to be adjusted from a first value to a second value, wherein, at the second value, the first signal level exceeds the second signal level by at least the target difference.

[0011] Other embodiments include, without limitation, a computer readable medium including instructions for performing one or more aspects of the disclosed techniques, as well as a method for performing one or more aspects of the disclosed techniques.

[0012] At least one advantage of the disclosed approach is that a listener who uses the disclosed listening device equipped with the audio processing system 100 hears a high-quality audio signal from an audio playback device plus certain alert signals from the environmental sound, while, at the same time, other sounds from the environmental sound are suppressed relative to the alert signal.

BRIEF DESCRIPTION OF THE DRAWINGS

[0013] So that the manner in which the above recited features of embodiments of the invention can be understood in detail, a more particular description of the invention, briefly summarized above, may be had by reference to embodiments, some of which are illustrated in the appended drawings. It is to be noted, however, that the appended drawings illustrate only typical embodiments of this invention and are therefore not to be considered limiting of its scope, for the invention may admit to other equally effective embodiments.

Figure 1 illustrates an audio processing system configured to implement one or more aspects of the various embodiments;

Figure 2 illustrates an exemplary audio ducking function performed by the audio processing system of Figure 1, according to various embodiments;

Figure 3 sets forth an exemplary set of waveforms that illustrates the audio ducking function performed by the audio processing system of Figure 1, according to various embodiments; and

Figures 4A-4B set forth a flow diagram of method steps for processing entertainment signals and alert signals, according to various embodiments.

DETAILED DESCRIPTION

[0014] In the following description, numerous specific details are set forth to provide a more thorough understanding of certain specific embodiments. However, it will be apparent to one of skill in the art that other embodiments may be practiced without one or more of these specific details or with additional specific details.

[0015] As described herein, an audio processing system in a listening device determines when an alert signal is present in the environmental sound and, in response, intelligently attenuates or ducks the audio level of an entertainment signal to maintain a voice-to-music ratio (VMR) at or near a target value. Stated another way, the audio processing system maintains a relatively constant alert signal level to entertainment signal level ratio (AER). Although the disclosed techniques are described in the context of a personal listening device, the techniques may also be applied to other audio systems, including, without limitation, a public address system. In general, the entertainment signal level corresponds to the volume setting on a playback device as set by a listener. The alert signal level corresponds to the strength of a signal of interest, such as speech, present in the environment. In one example, and without limitation, a low alert signal level could result from a soft-spoken server at a restaurant announcing that the table is ready. If the server's announcement is quieter than the voices of patrons, other servers, or the maître d', the entertainment signal is more heavily attenuated so that the server's voice can be heard. In another example, and without limitation, a low alert signal level could result from a fellow cyclist speaking at some distance away from the listener wearing earbuds that include microphones. Because sound levels are reduced over space due to acoustic spreading, a cyclist who is 20 feet away from the listener sounds quieter than a cyclist who is 6 feet away. As a result, the amount of ducking applied to the entertainment signal increases as the strength of the entertainment signal increases. However, the amount of ducking applied to the entertainment signal decreases as the strength of the alert signal increases.

[0016] The audio processing system applies intelligent ducking when an alert signal is actually present in the environment audio and is sufficiently loud. Therefore, if there is no alert signal or if a signal of interest, referred to herein as a potential alert signal, is at a negligible level, then audio processing system does not apply ducking to the entertainment signal. As a result, low-level, steady-

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state background sounds, such as distant, quiet, conversations at a large distance, are prevented from engaging the ducking system. This audio processing system is now described in greater detail.

System Overview

[0017] Figure 1 illustrates an audio processing system 100 configured to implement one or more aspects of the various embodiments. As shown, audio processing system 100 includes, without limitation, sound environment processor 130, voice separator 135, ambience separator 140, microphone root-mean square (RMS) detector 145, audio input RMS detector 150, voice activity detector 155, ratio-based attenuator 165, intelligent system monitor 160, attack release logic 170, voltage controlled amplifier 175, and mixer 180.

[0018] Sound environment processor 130 receives a microphone (mic) input signal 105 from one or more microphones that capture sound emanating from the environment. In some embodiments, sound environment processor 130 receives sound emanating from the environment electronically rather than via one or more microphones. Sound environment processor 130 performs noise reduction on the microphone input. Sound environment processor 130 cleans and enhances the microphone input by removing one or more noise signals, including, without limitation, microphone (mic) hiss, steady-state noise, very low frequency sounds (such as traffic din), and other low-level, steady-state sounds, while leaving intact any potential alert signal. In general, a low-level sound is a sound with a signal level that is below a threshold of loudness. In some embodiments, a gate may be used to remove such low-level signals from the input signal before transmitting the processed signal as an output. In general, a steady-state sound is a sound with a signal amplitude that remains relatively constant over time, in contrast to a transient sound with a signal amplitude that changes over time. In one example, and without limitation, the sound of an idling car could be considered a steady-state sound while the sound of an accelerating car or a car with a revving engine would not be considered a steady-state sound. In another example, and without limitation, the sound of operatic singing could be considered a steady-state sound while the sound of speech would not be considered a steady-state sound. In yet another example, and without limitation, the sound of very slow, symphonic music could be considered a steady-state sound while the sound of relatively faster, percussive music would not be considered a steady-state sound. The potential alert signal includes sounds that are not low-level, steady-state sound, such as human speech or an automobile horn. Sound environment processor 130 transmits a high-fidelity noise-reduced signal to mixer 180. Sound environment processor 130 also transmits a control signal to voice separator 135, ambience separator 140, and microphone RMS detector 145. In some embodiments, each of the control signals transmitted to voice separator 135, ambience separator 140, and microphone RMS detector 145 may be identical copies of the same signal. The control signal may be a lower fidelity version of the noise-reduced audio that is transmitted to mixer 180.

[0019] Voice separator 135 receives a control signal from sound environment processor 130. Voice separator 135 extracts any potential alert signal, such as speech, that is present in the received control signal. In some embodiments, voice separator 135 performs this function by applying a bandpass filter followed by an RMS detector to detect voice signals and other transient signals. In some embodiments, voice separator 135 may determine that audio energy in the range of 500 Hz to 3,000 Hz is considered to be a potential alert signal. In these embodiments, the output from voice separator 135 may be the RMS level of the audio energy in the range of 500 Hz to 3,000 Hz. If the RMS level of the audio in this frequency range exceeds a given threshold, voice separator 135 may determine that a potential alert signal is present. Voice separator 135 transmits a control signal to voice activity detector 155.

[0020] Ambience separator 140 is an optional module that receives a control signal from sound environment processor 130. Ambience separator 140 compares the RMS level of the microphone input against a threshold level to detect ambient signals, such as traffic noise, construction noise and other background noise, that pass through sound environment processor 130. In some embodiments, ambience separator 140 performs this function via a feature extraction approach. Ambience separator 140 transmits a control signal to voice activity detector 155.

[0021] Microphone RMS detector 145 receives a control signal from voice separator 135. Microphone RMS detector 145 detects the root mean square (RMS) level of the processed microphone input signal received from voice separator 135. Microphone RMS detector 145 calculates the energy of the processed microphone input signal in the log (dB units) domain. In some embodiments, microphone RMS detector 145 may employ an alternative signal level measurement technique other than detecting the RMS level of the signal. In one example, and without limitation, microphone RMS detector 145 employs a more sophisticated psychoacoustic signal level measurement technique. Microphone RMS detector 145 transmits a control signal, corresponding to the detected microphone RMS level, to ratio-based attenuator 165.

[0022] Audio input RMS detector 150 receives an entertainment signal from an audio playback device via audio input 110. Audio input RMS detector 150 detects the root mean square (RMS) level of the entertainment signal received via audio input 110. Audio input RMS detector 150 calculates the energy of the entertainment signal in the log (dB units) domain. In some embodiments, audio RMS detector 150 may employ an alternative signal level measurement technique other than detecting the RMS

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level of the signal. In one example, and without limitation, audio RMS detector 150 employs a more sophisticated psychoacoustic signal level measurement technique. Audio input RMS detector 150 transmits a control signal, corresponding to the calculated entertainment signal input RMS level, to ratio-based attenuator 165.

[0023] Voice activity detector 155 receives control signals from voice separator 135, ambience separator 140, and intelligent system monitor 160. Voice activity detector 155 analyzes the received control signals from voice separator 135 and ambience separator 140 and compares these two control signals against a ducker threshold 115 to detect the presence of an alert signal and, if an alert signal is present, whether or not to duck the entertainment audio received via audio input 110. In some embodiments, the ducker threshold 115 may be set via a user control. Voice activity detector 155 determines that an alert signal is present when the control signal received from voice separator 135 exceeds the ducker threshold 115. When ambience separator 140 is present, voice activity detector 155 further determines that an alert signal is present when the control signal received from voice separator 135 includes an alert signal that ambience separator 140 does not identify as an ambient signal. As a result, voice activity detector 155 reduces the likelihood that louder, low-frequency, steady-state sounds are misclassified as alert signals.

[0024] In some embodiments, if voice activity detector 155 determines that an alert signal is present, voice activity detector 155 may calculate a control signal based on the ratio of the signal from voice separator 135 versus the ducker threshold 115 If voice activity detector 155 determines that no alert signal is present, then voice activity detector 155 may set this control signal to 1, that is, unity gain, to inhibit ducking of the audio input 110. As further discussed herein, when voice activity detector 155 determines that an alert signal is no longer present, voice activity 155 continues to apply ducking during a ducking hold signal received from intelligent system monitor 160. Voice activity detector 155 transmits a control signal to intelligent system monitor 160 and to ratio-based attenuator 165.

[0025] Intelligent system monitor 160 receives a control signal from voice activity detector 155. Intelligent system monitor 160 determines the duration of ducking and the speed of ducking, based on the received control signal. As one example, and without limitation, human speech could vary in volume, where some portions of the human speech could be louder while other portions of the human speech are quieter. In addition, no human speech would be present between words or during short pauses. In such cases, audio ducking could be applied and removed multiple times during speech. Consequently, the entertainment signal present on audio input 110 would be ducked and released from ducking in rapid succession, resulting in an unpleasant listener experience. Intelligent system monitor 160 could effectively apply a hold time to maintain a ducking function for a threshold duration of time after an alert signal is no longer present at the microphone input 105. As a result, ducking would be maintained over short gaps, such as between words or short pauses, but ducking would be released when a relatively long pause is detected.

[0026] In some embodiments, intelligent system monitor 160 may determine that duration of ducking may be zero, resulting in no ducking. Intelligent system monitor 160 generates a control signal that identifies when ducking hold time should be applied. Intelligent system monitor 160 transmits the control signal to voice activity detector 155 and attack/release logic 170. Intelligent system monitor 160 optionally transmits the control signal to ratio-based attenuator 165.

[0027] Ratio-based attenuator 165 receives RMS audio levels from microphone RMS detector 145 and audio input RMS detector 150. Ratio-based attenuator 165 also receives control signals from voice activity detector 155 and optionally from intelligent system monitor 160. Ratio-based attenuator 165 determines the amount of gain reduction to be applied to the entertainment signal received via audio input 110. Ratio-based attenuator 165 receives a voice-to-music ratio (VMR) 120 that specifies a target level by which the audio input 110 is ducked relative to the microphone input 105. In some embodiments, the VMR 120 may be set via a user control. For example, and without limitation, if the VMR 120 is set to -30 dB and ducking is applied, then ratio-based attenuator 165 sets the attenuation of the signal received at audio input 110 so that the level of audio input 110 is at least -30 dB relative to the microphone input 105. If the control signal from intelligent system monitor 160 indicates that ducking hold time is applied, then ratio-based attenuator 165 maintains the most recent ducking level until the ducking hold is removed. During ducking hold time, ratio-based attenuator 165 neither increases nor decreases the ducking level. Ratio-based attenuator 165 generates a control signal that indicates the amount that the audio input 110 signal is to be ducked. In general, ratio-based attenuator 165 modifies the level of the entertainment signal via attenuation, rather than amplification, so the gain of the entertainment signal does not exceed unity. Ratio-based attenuator 165 transmits this control signal to attack/release logic 170.

[0028] Attack/release logic 170 receives control signals from intelligent system monitor 160 and ratio-based attenuator 165. Based on these control signals, attack/release logic 170 applies a smoothing function to the gain reduction resulting from ducking in order to prevent undesirable modulation of the entertainment signal received via audio input 110. In some embodiments, attack/release logic 170 may employ asymmetric attack and release profiles, such as a relatively fast attack profile and a relatively slower release profile. Attack/release logic 170 applies a fast attack to quickly duck the audio input 110 signal when a new alert signal is detected in order to avoid cutting off the beginning of the alert signal and, thereby, preserve the intelligibility of speech. Attack/re-

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lease logic 170 applies a relatively slower release to avoid abrupt changes to the entertainment signal when an alert signal is no longer detected.

[0029] Additionally, in some embodiments, attack/release logic 170 may apply a hysteresis logic to prevent undesirable modulation of the entertainment signal when the ducker is engaged. Via this hysteresis logic, attack/release logic 170 employs a slower attack/slower release profile when audio ducking is near the target ducking level, based on the control signal received from intelligent system monitor 160. This control signal indicates whether an alert signal has recently appeared or disappeared in the microphone input 105 signal, or whether the signal of interest has been present for a threshold duration of time. If an alert signal has recently appeared or disappeared, then attack/release logic 170 applies a fast attack/slow release profile. If, on the other hand, an alert signal has been present for a threshold period of time, then attack/release logic 170 applies a slower attack/slower release profile. The slower attack/slower release profile still responds to changes in RMS levels received from the microphone input 105 and audio input 110, but at a slower response rate relative to the fast attack/slow release profile. Attack/release logic 170 transmits a smoothed gain control signal to voltage controlled amplifier 175.

[0030] Voltage controlled amplifier 175 receives a smoothed gain control signal from attack/release logic 170. Voltage controlled amplifier 175 receives the entertainment signal via the audio input 110. Voltage controlled amplifier 175 applies the smoothed gain control signal to the entertainment signal to generate a ducked entertainment signal. When audio ducking is applied, voltage controlled amplifier 175 applies more attenuation to the entertainment signal when the entertainment signal received via the audio input 110 is relatively louder or when the alert signal received via the microphone input 105 is relatively quieter. When no audio ducking is applied, voltage controlled amplifier 175 passes the entertainment signal received via the audio input 110 without attenuation. Voltage controlled amplifier 175 passes the resulting ducked entertainment signal to mixer 180.

[0031] Mixer 180 receives noise-reduced microphone audio from sound environment processor 130 and the ducked entertainment signal from voltage controlled amplifier 175. Mixer 180 mixes or combines the two received audio signals to generate an output signal. In some embodiments, mixer 180 may perform a simple additive or multiplicative mix on the two audio signals. In other embodiments, mixer 180 may perform any technically feasible mix or combine function on the two audio signals. Mixer 180 transmits the output signal to one or more speakers via audio output 125.

[0032] In various embodiments, audio processing system 100 may be implemented as a state machine, a central processing unit (CPU), digital signal processor (DSP), a microcontroller, an application-specific integrated circuit (ASIC), or any device or structure configured to process data and execute software applications. In some embodiments, one or more of the blocks illustrated in Figure 1 may be implemented with discrete analog or digital circuitry. In one example, and without limitation, the voltage controlled amplifier 175 and mixer 180 could be implemented with analog components. Audio processing system 100 may be associated with a mobile computer, dash-mounted device, personal digital assistant, mobile phone, wearable device, mobile device or any other device suitable for practicing one or more embodiments of the present invention. Audio processing system 100 may be configured to implement an intelligent audio processing system 100, as further described herein. This audio processing system 100 may be implemented in any technically feasible form factor, including, without limitation, headphones or ear buds with situational awareness (e.g via microphones), broadcast systems, interruptible foldback (IFB) systems, or paging systems.

Operations of the Audio Processing System

[0033] Figure 2 illustrates an exemplary audio ducking function 200 performed by the audio processing system 100 of Figure 1, according to various embodiments. As shown, and without limitation, the audio ducking function 200 is superimposed onto a 3-axis coordinate system, where the three axes include a speech level axis 210, a music level axis 220, and a desired music gain axis 230.

[0034] The speech level axis 210 includes discrete levels for negligible speech 212, quieter speech 214, normal speech 216, and yelling 218. Similarly, the music level axis 220 includes discrete levels for fortissimo (very loud) music 222, average music 224, pianissimo (very soft) music 226, and none (no music) 228. Although discrete levels are indicated on the speech level axis 210 and on the music level axis 220, both of these axes are continuous and include values between the discrete levels as shown. The desired music gain axis 230 indicates a multiplier to apply to the incoming music signal based on the speech level, as shown on the speech level axis 210, and on the music level, as shown on the music level axis 220. A value of 1 on the desired music gain axis 230 indicates that the incoming music is multiplied by 1, a value of 0.9 on the desired music gain axis 230 indicates that the incoming music is multiplied by 0.9, value of 0.8 on the desired music gain axis 230 indicates that the incoming music is multiplied by 0.8, and so on. As shown, A "unity" gain of 1.0 (dark red) indicates no attenuation of the input music signal, and a "null" gain of 0.0 (dark blue) indicates heavy attenuation (e.g. 30dB) of the input music signal.

[0035] Following the line segment 240(0), corresponding to a negligible speech level, the desired music gain is 1 regardless of the music level. Line segment 240(0) indicates that there is no attenuation of the music signal when the speech level is negligible. Therefore, the listener hears only the music signal.

[0036] Following the line segment 240(1), correspond-

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ing to a quieter speech level, the desired music gain is approximately 0.18 when the music level is fortissimo. The desired music gain increases as the music level decreases until the desired music gain is 1 when there is no music signal. Line segment 240(1) indicates that a fortissimo music signal is significantly attenuated to ensure that a quieter speech level is audible over very loud music. Less attenuation is warranted for lower music levels, and no attenuation is needed when no music signal is present, such as during the silent gaps between songs. In one embodiment, when no music signal is present, the prior ducking level may be maintained for a threshold period of time in order to prevent a change in ducking level between songs.

[0037] Following the line segment 240(2), corresponding to a normal speech level, the desired music gain is approximately 0.32 when the music level is fortissimo. The desired music gain increases as the music level decreases until the desired music gain is 1 when there is no music signal. Line segment 240(2) indicates that a fortissimo music signal is moderately attenuated to ensure that a normal speech level is audible over very loud music. Less attenuation is warranted for lower music levels, and no attenuation is needed when no music signal is present.

[0038] Following the line segment 240(3), corresponding to a very loud speech level, such as yelling, the desired music gain is approximately 0.50 when the music level is fortissimo. The desired music gain increases as the music level decreases until the desired music gain is 1 when there is no music signal. Line segment 240(3) indicates that a fortissimo music signal is lightly attenuated to ensure that a loud level speech is audible over very loud music. Less attenuation is warranted for lower music levels, and no attenuation is needed when no music signal is present.

[0039] The audio ducking function 200 illustrated in Figure 2 bases the desired music gain on both the speech level and the music level in order to adapt to various combinations of music and speech levels. In general, the desired music gain increases as the music level decreases. The desired music gain remains constant as the speech level increases from negligible until quieter speech is detected. The desired music level then sharply decreases to a relatively low level and then gradually increases as the speech level increases from quieter speech to very loud speech.

[0040] Figure 3 sets forth an exemplary set of waveforms 300 that illustrates the audio ducking function performed by the audio processing system 100 of Figure 1, according to various embodiments. As shown, and without limitation, the set of waveforms 300 includes an input speech waveform 310, an input music waveform 320, and an output music waveform 330. These three waveforms are superimposed onto a 2-axis coordinate system. The vertical axis 340 indicates audio voltage level in dBu, a root mean square (RMS) measurement of voltage referenced to approximately 0.775 V RMS. The horizontal axis 350 indicates time in seconds.

[0041] The input music level, as shown by input music waveform 320 begins at -16 dBu during the time interval t=0 to t=30. The input music level reduces to -35 dBu during the time interval t=30 to t=60. The input music level reduces further to -75 dBu during the time interval t=60 to t=90.

[0042] During the time interval t=0 to t=5, the input speech waveform 310 indicates that no speech (approximately -90 dBu input level) is present. During this period, no ducking occurs, and the level of the output music waveform 330 remains at the level of the input music waveform 320. During the time period t=5 to t=10, the input speech waveform 310 indicates that speech is present at a level of -15 dBu. The input music level is ducked to a level that is 11 dB below the speech level. Because the input music level is at -16 dBu (below the input speech level), the level of the output music waveform 330 is ducked 10 dB to -26 dBu. During the time period t=10 to t=20, no speech (approximately -90 dBu input level) is present, and the ducking function is released, allowing the output music level to return to the -16 dBu level. During the time period t=20 to t=25, the input speech waveform 310 indicates that speech is present at a quieter level of -27 dBu. The input music level is ducked 22dB to a level that is 11 dB below the speech level, or -38 dBu.

[0043] During the time interval t=25 to t=35, the input speech waveform 310 indicates that no speech (approximately -90 dBu input level) is present. During this period, no ducking occurs, and the level of the output music waveform 330 returns to the level of the input music waveform 320. At time t=30, the input music level, and, correspondingly, the output music level, is reduced to -35 dBu. During the time period t=35 to t=40, the input speech waveform 310 indicates that speech is present at a level of -15 dBu. Because the input music level is already more than 11 dB below the speech level, no ducking occurs. During the time period t=40 to t=50, no speech (approximately -90 dBu input level) is present, and the output music level remains at the input music level of -35 dBu. During the time period t=50 to t=55, the input speech waveform 310 indicates that speech is present at a quieter level of -27 dBu. The input music level is ducked to a level that is 11 dB below the speech level. Because the input music level is at -35 dBu (below the input speech level), the level of the output music waveform 330 is ducked 3 dB to -38 dBu.

[0044] During the time interval t=55 to t=65, the input speech waveform 310 indicates that no speech (approximately -90 dBu input level) is present. During this period, no ducking occurs, and the level of the output music waveform 330 returns to the level of the input music waveform 320. At time t=60, the input music level, and, correspondingly, the output music level, is reduced to -75 dBu. During the time period t=65 to t=70, the input speech waveform 310 indicates that speech is present at a level of -15 dBu. Because the input music level is already more

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than 11 dB below the speech level, no ducking occurs. During the time period t=70 to t=80, no speech (approximately -90 dBu input level) is present, and the output music level remains at the input music level of -75 dBu. During the time period t=80 to t=85, the input speech waveform 310 indicates that speech is present at a quieter level of -27 dBu. Again, the input music level is already more than 11 dB below the speech level, and no ducking occurs. Finally, during the time period t=85 to t=90, no speech (approximately -90 dBu input level) is present, and the output music level remains at the input music level of -75 dBu.

[0045] Note that the output music waveform 330 indicates a fast attack time when applying ducking at t=5, t=20, and t=50, as illustrated by the relatively short fall time. The output music waveform 330 indicates a slower release time when releasing ducking at t=10, t=25, and t=55, as illustrated by the longer rise time. The rapid attack function allows the speech signal to pass through the listening device quickly to avoid cutting the initial portion of the speech signal. The long release function allows the audio ducking to release slowly, providing a less jarring experience for the listener and allowing for short gaps and pauses during speech before fully releasing the ducking function.

[0046] Figures 4A-4B set forth a flow diagram of method steps for processing entertainment signals and alert signals, according to various embodiments. Although the method steps are described in conjunction with the systems of Figures 1-3, persons skilled in the art will understand that any system configured to perform the method steps, in any order, is within the scope of the present disclosure.

[0047] As shown, a method 400 begins at step 402, where sound environment processor 130 receives environmental sound via a microphone input 105 and an entertainment signal via an audio input 110. The environmental sound represents the sound present in the environment of a listener. The entertainment signal represents music or other audio selected by the listener, such as music from an audio playback device. At step 404, sound environment processor 130 performs noise reduction on the environmental sound to remove low-level, steady-state sounds from the microphone input 105 signal, such as traffic noise and low-level microphone hiss. At step 406, the microphone RMS detector 145 calculates the RMS level of the voice separated sound received via the voice separator 135. At step 408, audio input RMS detector 150 calculates the RMS level of the entertainment signal received via the audio input 110.

[0048] At step 410, voice separator 135 detects that a potential alert signal, such as human speech, is present in the microphone input signal. In some embodiments, the voice separator 135 may perform this function by applying a first bandpass filter to the microphone input signal to generate a filtered microphone input signal. The voice separator 135 may calculate the RMS level associated with the filtered microphone input signal. If this RMS exceeds a threshold level, then an alert signal may be present in the microphone input signal.

[0049] At step 412, ambience separator 140 detects that ambient noise, such as background construction noise or background conversation noise, is present in the microphone input signal. Such ambient sound may still be present in the microphone input signal if the ambient noise is too loud or irregular to be filtered by noise reduction. In some embodiments, the ambience separator 140 may perform this function by applying a second bandpass filter to the microphone input signal to generate a filtered microphone input signal. The ambience separator 140 may calculate the RMS level associated with the filtered microphone input signal. The first and second bandpass filters may have the same frequency response, but the response time for ambient RMS detection may be slower than the response time for microphone input RMS detection.

[0050] At step 414, voice activity detector 155 determines whether the RMS level of the potential alert signal, such as voice or human speech, exceeds the RMS level of the ambient sound. If the RMS level of the signal of interest does not exceed the RMS level of the ambient sound, then the method 400 proceeds to step 402, described above. If, on the other hand, the RMS level of the signal of interest exceeds the RMS level of the ambient sound, then the method 400 proceeds to step 414, where the voice activity detector 155 determines whether the RMS level of the potential alert signal, such as voice or human speech, exceeds a ducker threshold level. If the RMS level of the signal of interest does not exceed the ducker threshold level, then the method 400 proceeds to step 402, described above. If, on the other hand, the RMS level of the signal of interest exceeds the ducker threshold level, then the method 400 proceeds to step 418, where ratio-based attenuator 165 determines whether the RMS level of the microphone input 105 exceeds the RMS level of the audio input 110 by at least a target difference, such as a voice-to-music ratio (VMR). In some embodiments, the target difference may be a fixed value. In other embodiments, the target difference may be user adjustable, or may depend on the entertainment signal level or on the alert signal level. If the RMS level of the microphone input 105 exceeds the RMS level of the audio input 110 by at least the VMR, then the method 400 proceeds to step 402, described above. If, on the other hand, the RMS level of the microphone input 105 does not exceed the RMS level of the audio input 110 by at least a target difference, then the method 400 proceeds to step 420, where ratio-based attenuator 165 calculates a reduction amount for the audio input 110 such that the RMS level of the microphone input 105 equals the RMS level of the audio input by the target difference.

[0051] At step 422, attack/release logic 170 applies an attack/release profile to the audio input 110. In some embodiments, attack/release logic 170 may apply a fast attack/slow release profile when presence or removal of an alert signal is first detected. Attack/release logic 170

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may apply a slower attack/slower release profile when the alert signal has been present for a minimum duration of time. At step 424, voltage controlled amplifier 175 attenuates the audio input 110 based on the level of ducking determined in the prior steps. At step 426, mixer 180 combines the noise-reduced microphone input and the attenuated audio input.

[0052] At step 428, intelligent system monitor 160 determines that the alert signal is no longer present in the microphone input 105. At step 430, intelligent system monitor 160 maintains the audio level of the audio input 110 at the attenuated level for a minimum duration of time. At step 432, after expiration of the minimum duration of time, intelligent system monitor 160 causes the audio level of the audio input 110 to increase to the pre-adjusted level. The method 400 then proceeds to step 402, described above. In various embodiments, the steps of method 400 may be performed in a continuous loop until certain events occur, such as powering down a device that includes the audio processing system 100.

[0053] In sum, an audio processing system 100 combines environmental sound with an entertainment signal, such as music from an audio playback device, allowing the listener to continue to hear the entertainment signal at a reduced level relative to the environmental sound when an alert signal, such as speech, is present in the environmental sound. The audio processing system 100 attenuates or ducks the entertainment signal only enough to ensure that the alert signal is intelligible without attenuating the entertainment signal more than necessary, based on the levels of the entertainment signal and the alert signal. The audio processing system 100 receives a target voice-to-music (VMR) ratio, typically via a user control. The audio processing system 100 attenuates the entertainment signal just enough to maintain this target VMR ratio. The audio processing system 100 intelligently automatically detects the presence of an alert signal in noisy, loud, and complex sound environments, without end-user input or intervention.

[0054] At least one advantage of the approach described herein is that a listener who uses the disclosed listening device equipped with the audio processing system 100 hears a high-quality audio signal from an audio playback device plus certain alert signals from the environmental sound, while, at the same time, other sounds from the environmental sound are suppressed relative to the alert signal. The entertainment signal is reduced only when an alert signal is present in the environment sound and only to the degree necessary to hear the alert signal. As a result, the listener hears alert signals from the environment along with an improved entertainment signal relative to prior approaches, leading to a better quality audio experience for the listener.

[0055] The descriptions of the various embodiments have been presented for purposes of illustration, but are not intended to be exhaustive or limited to the embodiments disclosed. Many modifications and variations will be apparent to those of ordinary skill in the art without departing from the scope of the described embodiments. The scope of the invention is defined by the appended claims.

[0056] Aspects of the present embodiments may be embodied as a system, method or computer program product. Accordingly, aspects of the present disclosure may take the form of an entirely hardware embodiment, an entirely software embodiment (including firmware, resident software, micro-code, etc.) or an embodiment combining software and hardware aspects that may all generally be referred to herein as a "circuit," "module" or "system." Furthermore, aspects of the present disclosure may take the form of a computer program product embodied in one or more computer readable medium(s) having computer readable program code embodied thereon.

[0057] Any combination of one or more computer readable medium(s) may be utilized. The computer readable medium may be a computer readable signal medium or a computer readable storage medium. A computer readable storage medium may be, for example, but not limited to, an electronic, magnetic, optical, electromagnetic, infrared, or semiconductor system, apparatus, or device, or any suitable combination of the foregoing. More specific examples (a non-exhaustive list) of the computer readable storage medium would include the following: an electrical connection having one or more wires, a portable computer diskette, a hard disk, a random access memory (RAM), a read-only memory (ROM), an erasable programmable read-only memory (EPROM or Flash memory), an optical fiber, a portable compact disc read-only memory (CD-ROM), an optical storage device, a magnetic storage device, or any suitable combination of the foregoing. In the context of this document, a computer readable storage medium may be any tangible medium that can contain, or store a program for use by or in connection with an instruction execution system, apparatus, or device.

[0058] Aspects of the present disclosure are described above with reference to flowchart illustrations and/or block diagrams of methods, apparatus (systems) and computer program products according to embodiments of the disclosure. It will be understood that each block of the flowchart illustrations and/or block diagrams, and combinations of blocks in the flowchart illustrations and/or block diagrams, can be implemented by computer program instructions. These computer program instructions may be provided to a processor of a general purpose computer, special purpose computer, or other programmable data processing apparatus to produce a machine, such that the instructions, which execute via the processor of the computer or other programmable data processing apparatus, enable the implementation of the functions/acts specified in the flowchart and/or block diagram block or blocks. Such processors may be, without limitation, general purpose processors, special-purpose processors, application-specific processors, or field-programmable

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[0059] The flowchart and block diagrams in the Figures illustrate the architecture, functionality, and operation of possible implementations of systems, methods and computer program products according to various embodiments of the present disclosure. In this regard, each block in the flowchart or block diagrams may represent a module, segment, or portion of code, which comprises one or more executable instructions for implementing the specified logical function(s). It should also be noted that, in some alternative implementations, the functions noted in the block may occur out of the order noted in the figures. For example, and without limitation, two blocks shown in succession may, in fact, be executed substantially concurrently, or the blocks may sometimes be executed in the reverse order, depending upon the functionality involved. It will also be noted that each block of the block diagrams and/or flowchart illustration, and combinations of blocks in the block diagrams and/or flowchart illustration, can be implemented by special purpose hardware-based systems that perform the specified functions or acts, or combinations of special purpose hardware and computer instructions.

[0060] While the preceding is directed to embodiments of the present disclosure, other and further embodiments of the disclosure may be devised without departing from the basic scope thereof, and the scope thereof is determined by the claims that follow.

Claims

1. An audio processing system (100), comprising:

   an input device configured to receive a first audio signal (105) emanating from an environment and including a signal of interest;
   a voice activity detector (155) configured to:

   receive a first control signal from a voice separator (135),
   determine that the signal of interest is present in the first audio signal (105) by determining that the first control signal exceeds a ducker threshold, and
   generate a second control signal in response to the first audio signal (105); and

   a ratio-based attenuator (165) configured to:

   receive the second control signal from the voice activity detector (155),
   determine whether a first signal level associated with the first audio signal (105) exceeds a second signal level associated with a second audio signal (110) received from an audio playback device by at least a target difference, and
   if the first signal level exceeds the second signal level by at least a target difference, then maintain an audio level of the second audio signal, and
   if the first signal level does not exceed the second signal level by at least the target difference, then cause the audio level of the second audio signal (110) to be adjusted from a first value to a second value, wherein, at the second value, the first signal level exceeds the second signal level by at least the target difference.

2. The audio processing system (100) of claim 1, further comprising a sound environment analyzer (130) configured to perform one or more noise reduction operations on the first audio signal (105) to remove a noise signal from the first audio signal (105).

3. The audio processing system (100) of claim 1 or 2, further comprising the voice separator (135) configured to:

   apply a first bandpass filter to the first audio signal (105) to generate a first filtered audio signal;
   calculate a third signal level associated with the first filtered audio signal; and
   generate the first control signal based on the third signal level.

4. The audio processing system (100) of claim 3, further comprising an ambience separator (140) configured to:

   apply a second bandpass filter to the first audio signal (105) to generate a second filtered audio signal;
   calculate a fourth signal level associated with the second filtered audio signal; and
   generate a third control signal based on the fourth signal level.

5. The audio processing system (100) of claim 4, wherein the first bandpass filter and the second bandpass filter have substantially the same frequency range.

6. The audio processing system (100) of claim 4, wherein the voice activity detector (155) is further configured to:

   receive the third control signal from the ambience separator (140); and
   further determine that the signal of interest is present in the first audio signal (105) by determining that the first control signal also exceeds the third control signal.

7. The audio processing system (100) of any of claims

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1 - 6, further comprising:

a first signal level detector configured to calculate the first signal level; and
a second signal level detector configured to calculate the second signal level.

8. The audio processing system (100) of any of claims 1-7, wherein the ratio-based attenuator (165) is further configured to generate a fourth control signal that indicates an amount that the second audio level is to be adjusted, and further comprising a voltage controlled amplifier (175) configured to attenuate the second audio signal (110) based on the fourth control signal to generate a third audio signal.

9. The audio processing system (100) of claim 8, further comprising at least one of:

an intelligent system monitor (160) configured to determine that the audio level of the second audio signal (110) has been adjusted, subsequently, determine that the signal of interest is no longer present in the first audio signal (105), maintain the audio level of the second audio signal (110) at the second level for a threshold duration of time, and, after expiration of the threshold duration of time, cause the audio level of the second audio signal (110) to increase to the first level;
attack/release logic configured to apply a first attack/release profile to the second audio signal (110) when presence or removal of the signal of interest is first detected and apply a second attack/release profile to the second audio signal (110) when the presence of the signal of interest has been detected for a threshold duration of time;
a mixer (180) configured to combine the first audio signal (105) with the third audio signal to generate an output signal (125) and transmit the output signal (125) to a speaker.

10. A method for processing entertainment signals and alert signals, the method comprising:

receiving a first audio signal (105) emanating from an environment and including a signal of interest;
determining that a signal of interest is present in the first audio signal (105) by determining that a first control signal associated with the first audio signal (105) exceeds a ducker threshold;
determining whether a first signal level associated with the first audio signal (105) exceeds a second signal level associated with a second audio signal (110) received from an audio playback device by at least a target difference, and
if the first signal level exceeds the second signal level by at least a target difference, then maintaining an audio level of the second audio signal (110), and
if the first signal level does not exceed the second signal level by at least the target difference, then causing the audio level of the second audio signal (110) to be adjusted from a first value to a second value, wherein, at the second value, the first signal level exceeds the second signal level by at least the target difference.

11. The method of claim 10, further comprising:

applying a first bandpass filter to the first audio signal (105) to generate a first filtered audio signal;
calculating a third signal level associated with the first filtered audio signal;
generating the first control signal based on the third signal level;
applying a second bandpass filter to the first audio signal (105) to generate a second filtered audio signal;
calculating a fourth signal level associated with the second filtered audio signal;
generating a second control signal based on the fourth signal level; and
further determining that the signal of interest is present in the first audio signal (105) by determining that the first control signal also exceeds the second control signal.

12. The method of claim 11, wherein the first bandpass filter and the second bandpass filter have substantially the same frequency range.

13. The method of any of claims 10 - 12, wherein the first signal level exceeds the second signal level by at least the target difference, and further comprising:

determining that the second signal level has increased from a third value to a fourth value;
determining that, at the fourth value, the first signal level does not exceed the second signal level by at least the target difference; and
causing the audio level of the second audio signal to be adjusted from a fifth value to a sixth value, wherein, at the sixth value, the first signal level exceeds the second signal level by at least the target difference;

or wherein the first signal level does not exceed the second signal level by at least the target difference, and further comprising:

determining that the first signal level has increased from a third value to a fourth value;

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determining that, at the fourth value, the first signal level exceeds the second signal level by at least the target difference; and
causing the audio level of the second audio signal (110) to be adjusted from the second value to the first value.

14. The method of any of claims 10 - 12, wherein the first signal level does not exceed the second signal level by at least the target difference, and further comprising:

determining that the first signal level has decreased from a third value to a fourth value; and
causing the audio level of the second audio signal (110) to be adjusted from the second value to a fifth value, wherein, at the fifth value, the first signal level exceeds the second signal level by at least the target difference.

15. A computer-readable storage medium including instructions that, when executed by a processor, cause the processor to process entertainment signals and alert signals, by performing the method of any of claims 10 - 14.

Patentansprüche

1. Audioverarbeitungssystem (100), umfassend:

eine Eingabevorrichtung, die dazu konfiguriert ist, ein erstes Audiosignal (105) zu empfangen, das von einer Umgebung ausgeht und ein Signal von Interesse beinhaltet;
einen Sprachaktivitätsdetektor (155), der zu Folgendem konfiguriert ist:

Empfangen eines ersten Steuersignals von einem Sprachseparator (135),
Bestimmen, dass das Signal von Interesse in dem ersten Audiosignal (105) vorhanden ist, indem bestimmt wird, dass das erste Steuersignal einen Duckerschwellenwert überschreitet, und
Erzeugen eines zweiten Steuersignals als Reaktion auf das erste Audiosignal (105); und

einen verhältnisbasierten Dämpfer (165), der zu Folgendem konfiguriert ist:

Empfangen des zweiten Steuersignals von dem Sprachaktivitätsdetektor (155),
Bestimmen, ob ein dem ersten Audiosignal (105) zugeordneter erster Signalpegel einen zweiten Signalpegel, der einem von einer Audiowiedergabevorrichtung empfangenen zweiten Audiosignal (110) zugeordnet ist, um mindestens eine Zieldifferenz überschreitet, und
wenn der erste Signalpegel den zweiten Signalpegel um mindestens eine Zieldifferenz überschreitet, Beibehalten eines Audiopegels des zweiten Audiosignals, und
wenn der erste Signalpegel den zweiten Signalpegel nicht um mindestens die Zieldifferenz überschreitet, Veranlassen, dass der Audiopegel des zweiten Audiosignals (110) von einem ersten Wert auf einen zweiten Wert eingestellt wird, wobei der erste Signalpegel bei dem zweiten Wert den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet.

2. Audioverarbeitungssystem (100) nach Anspruch 1, ferner umfassend einen Klangumgebungsanalysator (130), der dazu konfiguriert ist, einen oder mehrere Geräuschreduzierungsvorgänge an dem ersten Audiosignal (105) durchzuführen, um ein Geräuschsignal aus dem ersten Audiosignal (105) zu entfernen.

3. Audioverarbeitungssystem (100) nach Anspruch 1 oder 2, ferner umfassend den Sprachseparator (135), der zu Folgendem konfiguriert ist:

Anwenden eines ersten Bandpassfilters auf das erste Audiosignal (105), um ein erstes gefiltertes Audiosignal zu erzeugen;
Berechnen eines dritten Signalpegels, der dem ersten gefilterten Audiosignal zugeordnet ist; und
Erzeugen des ersten Steuersignals basierend auf dem dritten Signalpegel.

4. Audioverarbeitungssystem (100) nach Anspruch 3, ferner umfassend einen Umgebungsseparator (140), der zu Folgendem konfiguriert ist:

Anwenden eines zweiten Bandpassfilters auf das erste Audiosignal (105), um ein zweites gefiltertes Audiosignal zu erzeugen;
Berechnen eines vierten Signalpegels, der dem zweiten gefilterten Audiosignal zugeordnet ist; und
Erzeugen eines dritten Steuersignals basierend auf dem vierten Signalpegel.

5. Audioverarbeitungssystem (100) nach Anspruch 4, wobei das erste Bandpassfilter und das zweite Bandpassfilter im Wesentlichen den gleichen Frequenzbereich aufweisen.

6. Audioverarbeitungssystem (100) nach Anspruch 4, wobei der Sprachaktivitätsdetektor (155) ferner zu

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Folgendem konfiguriert ist:

Empfangen des dritten Steuersignals von dem Umgebungsseparator (140); und
ferner Bestimmen, dass das Signal von Interesse in dem ersten Audiosignal (105) vorhanden ist, indem bestimmt wird, dass das erste Steuersignal auch das dritte Steuersignal überschreitet.

7. Audioverarbeitungssystem (100) nach einem der Ansprüche 1-6, ferner umfassend:

einen ersten Signalpegeldetektor, der zum Berechnen des ersten Signalpegels konfiguriert ist; und
einen zweiten Signalpegeldetektor, der zum Berechnen des zweiten Signalpegels konfiguriert ist.

8. Audioverarbeitungssystem (100) nach einem der Ansprüche 1-7, wobei der verhältnisbasierte Dämpfer (165) ferner dazu konfiguriert ist, ein viertes Steuersignal zu erzeugen, das einen Betrag angibt, um den der zweite Audiopegel einzustellen ist, und ferner umfassend einen spannungsgesteuerten Verstärker (175), der dazu konfiguriert ist, das zweite Audiosignal (110) basierend auf dem vierten Steuersignal zu dämpfen, um ein drittes Audiosignal zu erzeugen.

9. Audioverarbeitungssystem (100) nach Anspruch 8, ferner umfassend mindestens eines von:

einer intelligenten Systemüberwachungseinrichtung (160), die dazu konfiguriert ist, zu bestimmen, dass der Audiopegel des zweiten Audiosignals (110) eingestellt wurde, anschließend zu bestimmen, dass das Signal von Interesse nicht länger in dem ersten Audiosignal (105) vorhanden ist, den Audiopegel des zweiten Audiosignals (110) bei dem zweiten Pegel für eine Schwellenzeitdauer beizubehalten und nach Ablauf der Schwellenzeitdauer zu veranlassen, dass der Audiopegel des zweiten Audiosignals (110) auf den ersten Pegel erhöht wird;
einer Angriffs-/Freigabelogik, die dazu konfiguriert ist, ein erstes Angriffs-/Freigabeprofil auf das zweite Audiosignal (110) anzuwenden, wenn das Vorhandensein oder die Entfernung des Signals von Interesse zuerst detektiert wird, und ein zweites Angriffs-/Freigabeprofil auf das zweite Audiosignal (110) anzuwenden, wenn das Vorhandensein des Signals von Interesse für eine Schwellenzeitdauer detektiert wurde;
einem Mischer (180), der dazu konfiguriert ist, das erste Audiosignal (105) mit dem dritten Audiosignal zu kombinieren, um ein Ausgabesignal (125) zu erzeugen und das Ausgabesignal (125) an einen Lautsprecher zu übertragen.

10. Verfahren zum Verarbeiten von Entertainmentsignalen und Alarmsignalen, wobei das Verfahren Folgendes umfasst:

Empfangen eines ersten Audiosignals (105), das von einer Umgebung ausgeht und ein Signal von Interesse beinhaltet;
Bestimmen, dass ein Signal von Interesse in dem ersten Audiosignal (105) vorhanden ist, indem bestimmt wird, dass ein dem ersten Audiosignal (105) zugeordnetes erstes Steuersignal einen Duckerschwellenwert überschreitet;
Bestimmen, ob ein dem ersten Audiosignal (105) zugeordneter erster Signalpegel einen zweiten Signalpegel, der einem von einer Audiowiedergabevorrichtung empfangenen zweiten Audiosignal (110) zugeordnet ist, um mindestens eine Zieldifferenz überschreitet, und
wenn der erste Signalpegel den zweiten Signalpegel um mindestens eine Zieldifferenz überschreitet, Beibehalten eines Audiopegels des zweiten Audiosignals (110), und
wenn der erste Signalpegel den zweiten Signalpegel nicht um mindestens die Zieldifferenz überschreitet, Veranlassen, dass der Audiopegel des zweiten Audiosignals (110) von einem ersten Wert auf einen zweiten Wert eingestellt wird, wobei der erste Signalpegel bei dem zweiten Wert den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet.

11. Verfahren nach Anspruch 10, ferner umfassend:

Anwenden eines ersten Bandpassfilters auf das erste Audiosignal (105), um ein erstes gefiltertes Audiosignal zu erzeugen;
Berechnen eines dritten Signalpegels, der dem ersten gefilterten Audiosignal zugeordnet ist;
Erzeugen des ersten Steuersignals basierend auf dem dritten Signalpegel;
Anwenden eines zweiten Bandpassfilters auf das erste Audiosignal (105), um ein zweites gefiltertes Audiosignal zu erzeugen;
Berechnen eines vierten Signalpegels, der dem zweiten gefilterten Audiosignal zugeordnet ist;
Erzeugen eines zweiten Steuersignals basierend auf dem vierten Signalpegel; und
ferner Bestimmen, dass das Signal von Interesse in dem ersten Audiosignal (105) vorhanden ist, indem bestimmt wird, dass das erste Steuersignal auch das zweite Steuersignal überschreitet.

12. Verfahren nach Anspruch 11, wobei das erste Bandpassfilter und das zweite Bandpassfilter im Wesent-

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lichen den gleichen Frequenzbereich aufweisen.

13. Verfahren nach einem der Ansprüche 10-12, wobei der erste Signalpegel den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet, und ferner umfassend:

Bestimmen, dass sich der zweite Signalpegel von einem dritten Wert auf einen vierten Wert erhöht hat;
Bestimmen, dass der erste Signalpegel bei dem vierten Wert den zweiten Signalpegel nicht um mindestens die Zieldifferenz überschreitet; und
Veranlassen, dass der Audiopegel des zweiten Audiosignals von einem fünften Wert auf einen sechsten Wert eingestellt wird, wobei der erste Signalpegel bei dem sechsten Wert den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet;
oder wobei der erste Signalpegel den zweiten Signalpegel nicht um mindestens die Zieldifferenz überschreitet, und ferner Folgendes umfassend:

Bestimmen, dass sich der erste Signalpegel von einem dritten Wert auf einen vierten Wert erhöht hat;
Bestimmen, dass der erste Signalpegel bei dem vierten Wert den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet; und
Veranlassen, dass der Audiopegel des zweiten Audiosignals (110) von dem zweiten Wert auf den ersten Wert eingestellt wird.

14. Verfahren nach einem der Ansprüche 10-12, wobei der erste Signalpegel den zweiten Signalpegel nicht um mindestens die Zieldifferenz überschreitet, und ferner umfassend:

Bestimmen, dass sich der erste Signalpegel von einem dritten Wert auf einen vierten Wert verringert hat; und
Veranlassen, dass der Audiopegel des zweiten Audiosignals (110) von dem zweiten Wert auf einen fünften Wert eingestellt wird, wobei der erste Signalpegel bei dem fünften Wert den zweiten Signalpegel um mindestens die Zieldifferenz überschreitet.

15. Computerlesbares Speichermedium, das Anweisungen beinhaltet, die bei Ausführung durch einen Prozessor den Prozessor zur Verarbeitung von Entertainmentsignalen und Alarmsignalen veranlassen, indem das Verfahren nach einem der Ansprüche 10-14 durchgeführt wird.

Revendications

1. Système de traitement audio (100) comprenant :

un dispositif d'entrée configuré pour recevoir un premier signal audio (105) provenant d'un environnement et comportant un signal d'intérêt ;
un détecteur d'activité vocale (155) configuré pour :

recevoir un premier signal de commande d'un séparateur de voix (135),
déterminer que le signal d'intérêt est présent dans le premier signal audio (105) en déterminant que le premier signal de commande dépasse un seuil de Ducker, et
générer un deuxième signal de commande en réponse au premier signal audio (105) ; et

un atténuateur basé sur un rapport (165) configuré pour :

recevoir le deuxième signal de commande du détecteur d'activité vocale (155),
déterminer si un premier niveau de signal associé au premier signal audio (105) dépasse un deuxième niveau de signal associé à un deuxième signal audio (110) reçu d'un dispositif de lecture audio par au moins une différence cible et
si le premier niveau de signal dépasse le deuxième niveau de signal par au moins une différence cible, maintenir alors un niveau audio du deuxième signal audio et
si le premier niveau de signal ne dépasse pas le deuxième niveau de signal par au moins la différence cible, provoquer alors le réglage du niveau audio du deuxième signal audio (110) d'une première valeur à une deuxième valeur, dans lequel, à la deuxième valeur, le premier niveau de signal dépasse le deuxième niveau de signal par au moins la différence cible.

2. Système de traitement audio (100) selon la revendication 1, comprenant en outre un analyseur d'environnement sonore (130) configuré pour effectuer une ou plusieurs opérations de réduction de bruit sur le premier signal audio (105) afin de supprimer un signal de bruit du premier signal audio (105).

3. Système de traitement audio (100) selon la revendication 1 ou 2, comprenant en outre le séparateur de voix (135) configuré pour :

appliquer un premier filtre passe-bande au premier signal audio (105) pour générer un premier

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signal audio filtré ;
calculer un troisième niveau de signal associé au premier signal audio filtré ; et
générer le premier signal de commande sur la base du troisième niveau de signal.

4. Système de traitement audio (100) selon la revendication 3, comprenant en outre un séparateur d'ambiance (140) configuré pour :

appliquer un second filtre passe-bande au premier signal audio (105) pour générer un second signal audio filtré ;
calculer un quatrième niveau de signal associé au second signal audio filtré ; et
générer un troisième signal de commande sur la base du quatrième niveau de signal.

5. Système de traitement audio (100) selon la revendication 4, dans lequel le premier filtre passe-bande et le second filtre passe-bande ont sensiblement la même gamme de fréquences.

6. Système de traitement audio (100) selon la revendication 4, dans lequel le détecteur d'activité vocale (155) est en outre configuré pour :

recevoir le troisième signal de commande provenant du séparateur d'ambiance (140) ; et
déterminer en outre que le signal d'intérêt est présent dans le premier signal audio (105) en déterminant que le premier signal de commande dépasse également le troisième signal de commande.

7. Système de traitement audio (100) selon l'une quelconque des revendications 1 à 6, comprenant en outre :

un premier détecteur de niveau de signal configuré pour calculer le premier niveau de signal ; et
un second détecteur de niveau de signal configuré pour calculer le deuxième niveau de signal.

8. Système de traitement audio (100) selon l'une quelconque des revendications 1 à 7, dans lequel l'atténuateur basé sur un rapport (165) est en outre configuré pour générer un quatrième signal de commande qui indique une quantité que le deuxième niveau audio doit régler, et comprenant en outre un amplificateur commandé en tension (175) configuré pour atténuer le deuxième signal audio (110) sur la base du quatrième signal de commande afin de générer un troisième signal audio.

9. Système de traitement audio (100) selon la revendication 8, comprenant en outre au moins l'un :

d'un moniteur de système intelligent (160) configuré pour déterminer que le niveau audio du deuxième signal audio (110) a été réglé, puis pour déterminer que le signal d'intérêt n'est plus présent dans le premier signal audio (105), pour maintenir le niveau audio du deuxième signal audio (110) au deuxième niveau pour une durée seuil, et, après l'expiration de la durée seuil, pour faire monter le niveau audio du deuxième signal audio (110) jusqu'au premier niveau ;
d'une logique d'attaque/libération configurée pour appliquer un premier profil d'attaque/libération au deuxième signal audio (110) lorsque la présence ou la suppression du signal d'intérêt est d'abord détectée et pour appliquer un second profil d'attaque/libération au deuxième signal audio (110) lorsque la présence du signal d'intérêt a été détectée pendant une durée limite ;
d'un mélangeur (180) configuré pour combiner le premier signal audio (105) avec le troisième signal audio pour générer un signal de sortie (125) et transmettre le signal de sortie (125) à un haut-parleur.

10. Procédé de traitement de signaux de divertissement et de signaux d'alerte, le procédé comprenant :

la réception d'un premier signal audio (105) provenant d'un environnement et comportant un signal d'intérêt ;
le fait de déterminer qu'un signal d'intérêt est présent dans le premier signal audio (105) en déterminant qu'un premier signal de commande associé au premier signal audio (105) dépasse un seuil de Ducker ;
le fait de déterminer si un premier niveau de signal associé au premier signal audio (105) dépasse un deuxième niveau de signal associé à un deuxième signal audio (110) reçu d'un dispositif de lecture audio d'au moins une différence cible, et
si le premier niveau de signal dépasse le deuxième niveau de signal d'au moins une différence cible, alors le maintien d'un niveau audio du deuxième signal audio (110), et
si le premier niveau de signal ne dépasse pas le deuxième niveau de signal d'au moins la différence cible, alors le fait d'amener le niveau audio du deuxième signal audio (110) à être réglé d'une première valeur à une deuxième valeur, dans lequel, à la deuxième valeur, le premier niveau de signal dépasse le deuxième niveau de signal d'au moins la différence visée.

11. Procédé selon la revendication 10, comprenant en outre :

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l'application d'un premier filtre passe-bande au premier signal audio (105) pour générer un premier signal audio filtré ;
le calcul d'un troisième niveau de signal associé au premier signal audio filtré ;
la génération du premier signal de commande sur la base du troisième niveau de signal ;
l'application d'un second filtre passe-bande au premier signal audio (105) pour générer un second signal audio filtré ;
le calcul d'un quatrième niveau de signal associé au deuxième signal audio filtré ;
la génération d'un deuxième signal de commande sur la base du quatrième niveau de signal ; et
la détermination en outre que le signal d'intérêt est présent dans le premier signal audio (105) en déterminant que le premier signal de commande dépasse également le deuxième signal de commande.

12. Procédé selon la revendication 11, dans lequel le premier filtre passe-bande et le second filtre passe-bande ont sensiblement la même gamme de fréquences.

13. Procédé selon l'une quelconque des revendications 10 à 12, dans lequel le premier niveau de signal dépasse le deuxième niveau de signal d'au moins la différence cible, et comprenant en outre :

le fait de déterminer que le deuxième niveau de signal a augmenté d'une troisième valeur à une quatrième valeur ;
le fait de déterminer que, à la quatrième valeur, le premier niveau de signal ne dépasse pas le deuxième niveau de signal d'au moins la différence cible ; et
le fait d'amener le niveau audio du deuxième signal audio à être réglé d'une cinquième valeur à une sixième valeur, dans lequel, à la sixième valeur, le premier niveau de signal dépasse le deuxième niveau de signal d'au moins la différence cible ;
ou dans lequel le premier niveau de signal ne dépasse pas le deuxième niveau de signal d'au moins la différence cible, et comprenant en outre :

le fait de déterminer que le premier niveau de signal a augmenté d'une troisième valeur à une quatrième valeur ;
le fait de déterminer que, à la quatrième valeur, le premier niveau de signal dépasse le deuxième niveau de signal d'au moins la différence cible ; et
le fait d'amener le réglage du niveau audio du deuxième signal audio (110) à être réglé de la deuxième valeur à la première valeur.

14. Procédé selon l'une quelconque des revendications 10 à 12, dans lequel le premier niveau de signal ne dépasse pas le deuxième niveau de signal d'au moins la différence cible, et comprenant en outre :

le fait de déterminer que le premier niveau de signal a diminué d'une troisième valeur à une quatrième valeur ; et
le fait d'amener le niveau audio du deuxième signal audio (110) à être réglé de la deuxième valeur à une cinquième valeur, dans lequel, à la cinquième valeur, le premier niveau de signal dépasse le deuxième niveau de signal d'au moins la différence cible.

15. Support de stockage lisible par ordinateur comportant des instructions qui, lorsqu'elles sont exécutées par un processeur, amènent le processeur à traiter des signaux de divertissement et des signaux d'alerte en exécutant le procédé selon l'une quelconque des revendications 10 à 14.

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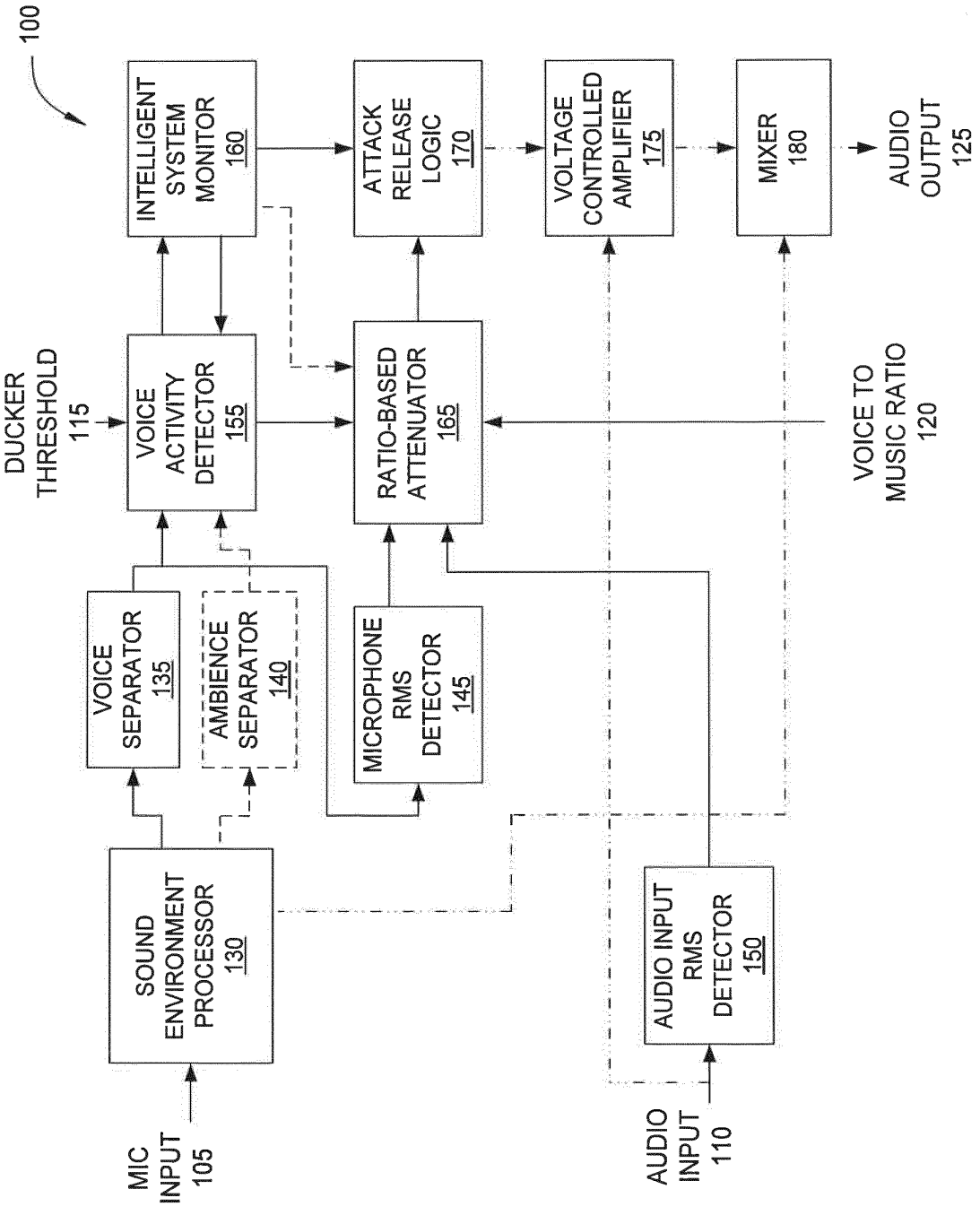

100
INTELLIGENT SYSTEM MONITOR 160
ATTACK RELEASE LOGIC 170
VOLTAGE CONTROLLED AMPLIFIER 175
MIXER 180
AUDIO OUTPUT 125
DUCKER THRESHOLD 115
VOICE ACTIVITY DETECTOR 155
RATIO-BASED ATTENUATOR 165
VOICE TO MUSIC RATIO 120
VOICE SEPARATOR 135
AMBIENCE SEPARATOR 140
MICROPHONE RMS DETECTOR 145
SOUND ENVIRONMENT PROCESSOR 130
AUDIO INPUT RMS DETECTOR 150
MIC INPUT 105
AUDIO INPUT 110

Figure 1

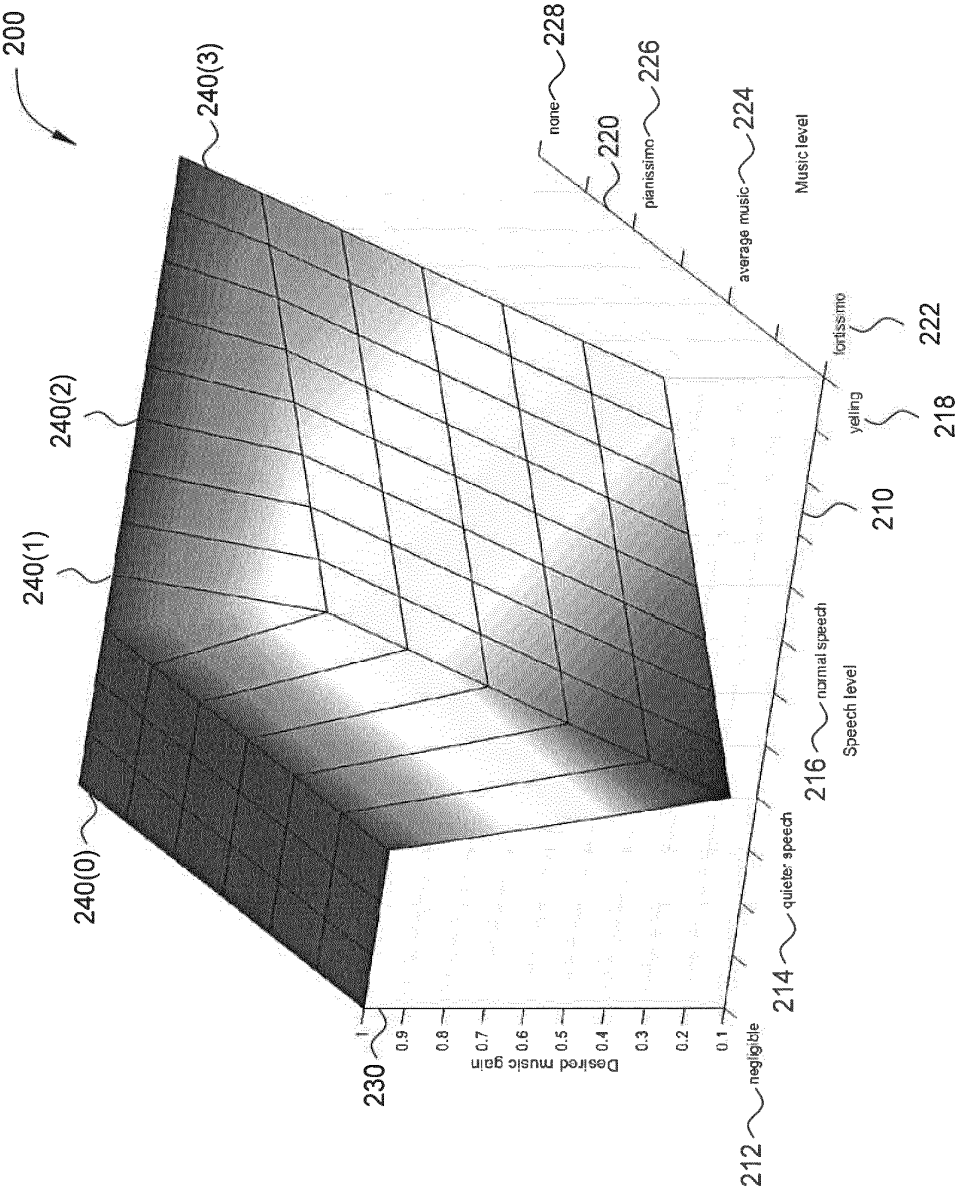


Figure 2

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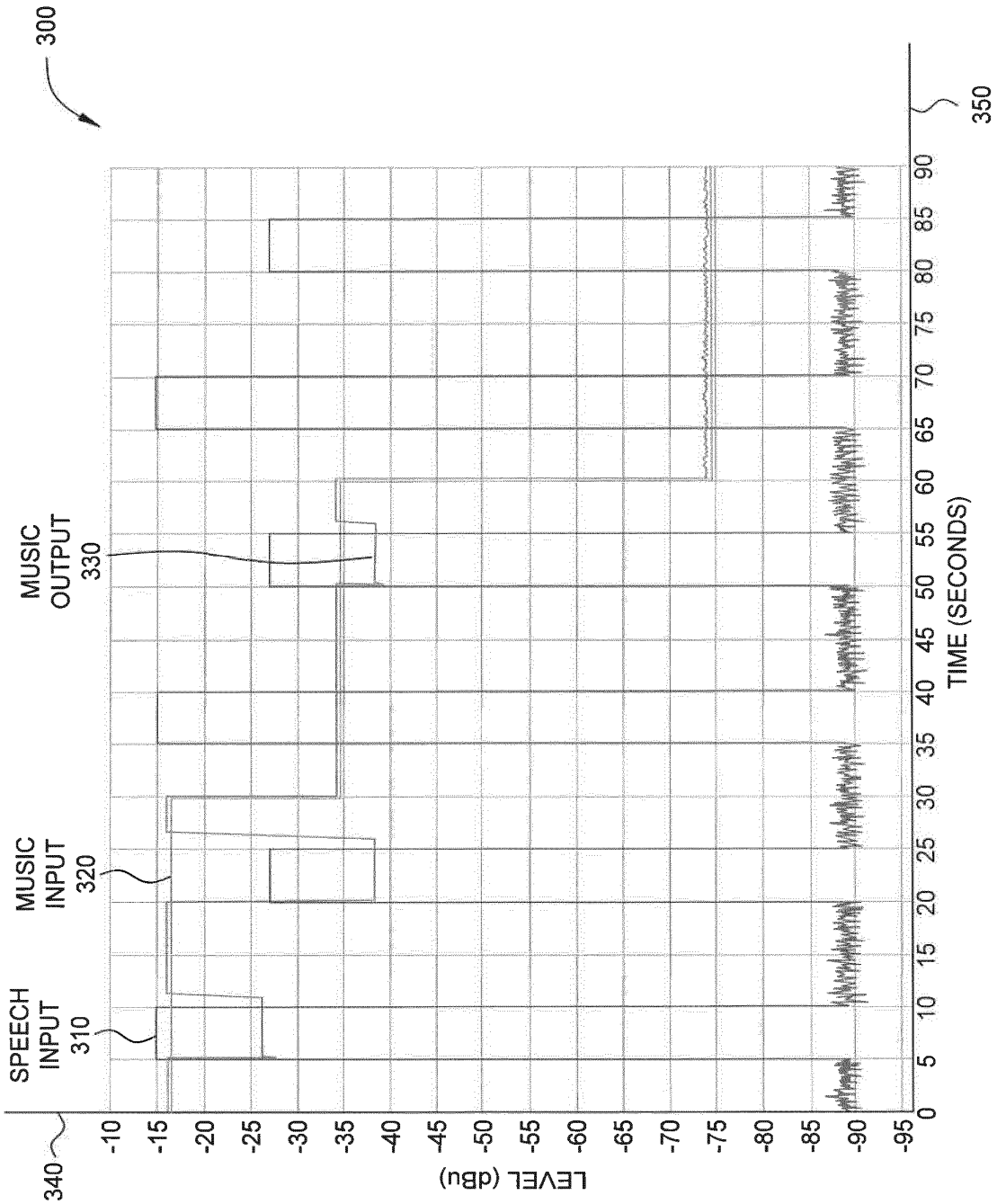


Figure 3

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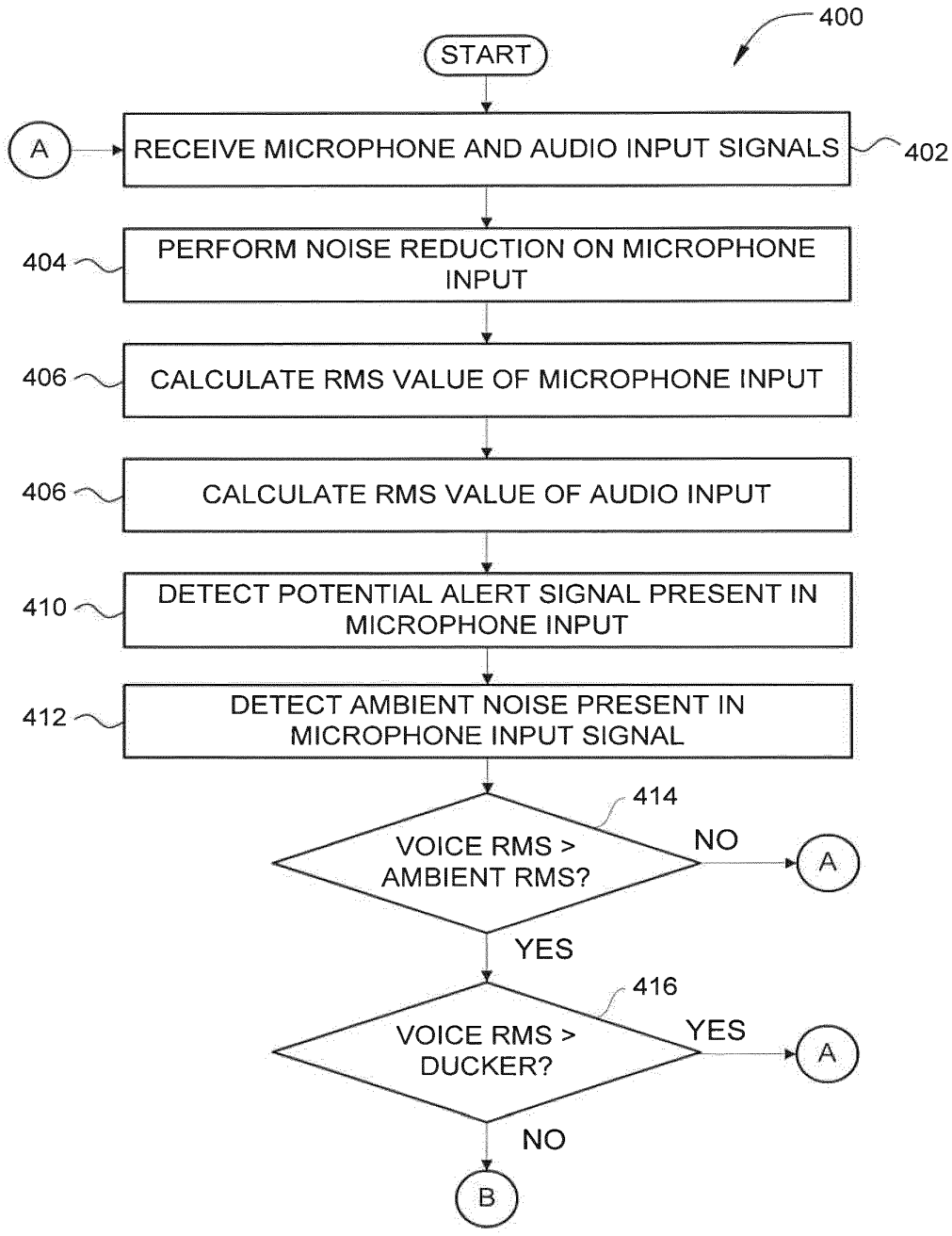

400
START
A
RECEIVE MICROPHONE AND AUDIO INPUT SIGNALS
402
404
PERFORM NOISE REDUCTION ON MICROPHONE INPUT
406
CALCULATE RMS VALUE OF MICROPHONE INPUT
406
CALCULATE RMS VALUE OF AUDIO INPUT
410
DETECT POTENTIAL ALERT SIGNAL PRESENT IN MICROPHONE INPUT
412
DETECT AMBIENT NOISE PRESENT IN MICROPHONE INPUT SIGNAL
414
VOICE RMS > AMBIENT RMS?
NO
A
YES
416
VOICE RMS > DUCKER?
YES
A
NO
B

Figure 4A

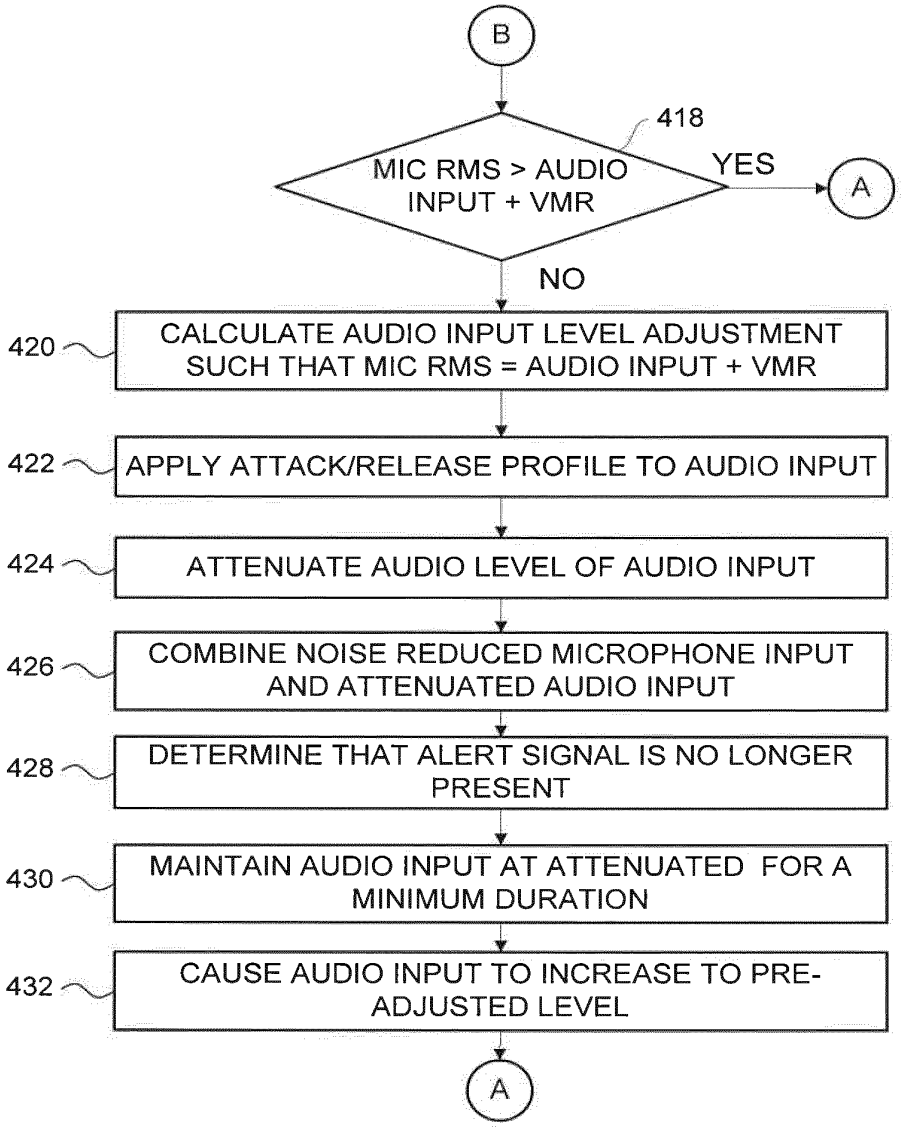

B
418
MIC RMS > AUDIO INPUT + VMR
YES
A
NO
420
CALCULATE AUDIO INPUT LEVEL ADJUSTMENT SUCH THAT MIC RMS = AUDIO INPUT + VMR
422
APPLY ATTACK/RELEASE PROFILE TO AUDIO INPUT
424
ATTENUATE AUDIO LEVEL OF AUDIO INPUT
426
COMBINE NOISE REDUCED MICROPHONE INPUT AND ATTENUATED AUDIO INPUT
428
DETERMINE THAT ALERT SIGNAL IS NO LONGER PRESENT
430
MAINTAIN AUDIO INPUT AT ATTENUATED FOR A MINIMUM DURATION
432
CAUSE AUDIO INPUT TO INCREASE TO PRE-ADJUSTED LEVEL
A

Figure 4B

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

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- US 20070189544 A1 **[0004]**
- WO 2011112382 A1 **[0005]**
- WO 2014022359 A2 **[0006]**

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Espacenet

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Applicants:

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null

Abstract

No abstract found. Please consult other publications of this patent family in "Available in", if displayed above.

US2017345406A1 METHOD, SYSTEM AND ITEM

Data originating from sources other than the EPO may not be accurate, complete, or up to date.

[0001] The invention relates to a method of, and a system for, enabling a hearer to hear desired sound while also being able to be aware of ambient sound. The invention is particularly, but not exclusively, applicable to cancelling sound, a system for, cancelling sound and an item comprising sound-cancelling apparatus.

[0002] It has become increasingly commonplace in recent years for people to listen to music and other audio supplied from mobile telephones and music players through in-ear phones or headphones which are so-called "enclosed", i.e. are significantly isolated acoustically from ambient sound. This has the disadvantage that the listener is unaware of ambient sound which he would definitely wish to hear, such as a warning of danger.

[0003] Many sound cancelling methods/applications are known in the art but generally are based on the same principle, namely, sound cancelling is achieved by creating a copy of a digital or analogue audio signal, reversing the polarity of the signal to create a corresponding 'negative' signal and then combining the two signals to achieve a total sound cancelling effect. The inherent problem with the known art is that this sound cancelling process cannot distinguish between wanted and unwanted sounds. Existing methods automatically assume that all background sound is unwanted and, therefore, are unsuitable for use as a sound cancelling system on mobile phones or on hearing aids where the user may need to be able to have a conversation or to be able to hear what is going on around him. To date, sound cancelling use has been limited because of the requirement to isolate acoustically a user who wears, for example, headphones (or earphones or headsets of any description). At present, expensive headphones/earphones are required for sound cancelling devices. Furthermore, users of hearing aids and music players, such as mp3 players, instinctively turn up the volume in their earphones to mask any unwanted ambient sound. This is disadvantageous as it can inadvertently cause damage to the user's hearing and, over a sustained period of use, may lead to permanent hearing loss.

[0004] The invention, which can be realised in mainly software or hardware form, is aimed at eliminating or significantly reducing the disadvantages of the known prior art.

[0005] Herein, the expression "non-audio" refers to transmission other than by sound waves through air, usually by electrical conduction.

[0006] According to a first aspect of the present invention, there is provided a method of enabling a hearer to hear desired sound while also being able to be aware of ambient sound, comprising providing a first non-audio signal representative of said desired sound, deriving a second non-audio signal from said ambient sound, combining the first and second non-audio signals in providing a third non-audio signal, and converting said third non-audio signal into sound.

[0007] According to a second aspect of the present invention, there is provided a system for enabling a hearer to hear desired sound while also being able to be aware of ambient sound, comprising a first outputting device for outputting a first non-audio signal representative of said desired sound, a second outputting device for outputting a second non-audio signal derived from said ambient sound, a combining arrangement for combining the first and second non-audio signal, and a converting device for emitting sound converted from said third non-audio signal.

[0008] Owing to those aspects of the invention, it is possible to enable a hearer to hear desired sound while also being able to be aware of ambient sound, especially in circumstances where, in order to listen to the desired sound, the hearer has deliberately excluded hearing most of ambient sound transmitted towards his ears through the air.

[0009] An "enclosed" headphone user equipped with a preferred embodiment of the present invention may listen to audio from any audio player or 'phone while also listening to and controlling/enhancing the level of surrounding ambient sound he hears after it has been picked up by an open microphone. The embodiment consists of a combination of audio processes, which may be employed in a number of configurations using some or all of the treatments of mixing, gating, equalizing, compressing, expanding and spectral enhancing. The user does not hear anything below a set threshold (gate), although he hears everything above it at a set maximum level (limiter/compressor), and such settings can be adjustable by the user, or pre-set by the supplier of the APP,

or automatically set to a level representing the average or other ambient sound amplitude measured over a pre-determined period. The embodiment preferably allows a user to:

- Hear ambient sound while using the headphones
- Set maximum audio level for ambient sound
- Hear only the ambient sound that exceeds a user-, pre-, or automatically, set threshold
- Choose either receiving a vibrate-only alarm or a visual message on the host device (e.g. MP3 player), or hearing the audio (ambience) picked up by the microphone when the threshold is exceeded
- Hear the microphone signal at any time by pressing a button
- Adjust the sensitivity of the microphone.

- Hear ambient sound while using the headphonesSet maximum audio level for ambient soundHear only the ambient sound that exceeds a user-, pre-, or automatically, set thresholdChoose either receiving a vibrate-only alarm or a visual message on the host device (e.g. MP3 player), or hearing the audio (ambience) picked up by the microphone when the threshold is exceededHear the microphone signal at any time by pressing a buttonAdjust the sensitivity of the microphone.

[0016] The embodiment may be constructed and configured in any number of alternate ways to produce versions that serve many purposes, e.g. alarm only, enhanced audio (e.g. louder), or processed to emphasise, exclude or include particular sounds or frequencies.

[0017] For example the embodiment can be configured to lower the heard music of an MP3 player and allow the user to hear the signal from the microphone when a set threshold is exceeded, or it can stop or pause the music or simply allow the microphone signal to be heard simultaneously with the music.

[0018] According to a third aspect of the present invention, there is provided a method of partially cancelling sound, comprising:

[0019] providing a first non-audio signal representative of a desired audio signal to be listened to;

[0020] providing a second non-audio signal representative of a partially-cancelling, ambient audio signal; and

[0000] combining the first and second non-audio signals in providing a heard audio signal.

[0021] According to a fourth aspect of the present invention, there is provided a method of partially cancelling sound, comprising:

[0022] providing a first non-audio signal representative of a desired audio signal to be listened to;

[0023] providing a second non-audio signal derived from an ambient audio signal;

[0024] altering the phase of the second non-audio signal to provide a substantially fully cancelling, phase-altered, non-audio signal and a partially-cancelling, phase-altered non-audio signal; and

[0025] combining the first non-audio signal, the substantially fully cancelling, phase-altered non-audio signal and the partially-cancelling, phase-altered non-audio signal to provide a heard audio signal.

[0026] According to a fifth aspect of the present invention, there is provided a system for partially cancelling sound, comprising:

[0027] a first outputting device for outputting a first non-audio signal representative of a desired audio signal to be listened to;

[0028] a providing device for providing a second non-audio signal representative of a partially-cancelling, phase-altered ambient audio signal;

[0029] a combining arrangement for combining the first and second non-audio signals in providing a third non-audio signal; and

[0030] a second outputting device for converting the third non-audio signal into a heard audio signal.

[0031] According to a sixth aspect of the present invention, there is provided a system for partially cancelling sound, comprising:

[0032] a first outputting device for outputting a first non-audio signal representative of a desired audio signal to be listened to;

[0033] a providing device for providing a second non-audio signal derived from an ambient audio signal;

[0034] a phase-altering arrangement for altering the phase of the second non-audio signal to provide a substantially fully cancelling, phase-altered, non-audio signal and a partially-cancelling, phase-altered non-audio signal;

[0035] a combining arrangement for combining the first non-audio signal, the substantially fully cancelling non-audio signal, and the partially-cancelling non-audio signal in providing a third non-audio signal; and

[0036] a second outputting device for converting the third non-audio signal into a heard audio signal.

[0037] According to a seventh aspect of the present invention, there is provided apparatus comprising an item consisting of any one of the group consisting of a telephone, a mobile phone, a television, a karaoke machine, a computer, a Hi-Fi, a surround-sound system, a games console, and an in-vehicle audio device, the apparatus comprising only one microphone, which is included in said item, and having a capability of playing audio files for a user to listen to,

[0038] wherein the microphone is capable of detecting ambient sound and the item further comprises:

[0039] a phase-altering arrangement for altering the phase of sound detected by the microphone to provide a phase-altered non-audio signal; and

[0040] a combining arrangement for combining the phase-altered non-audio signal and a representative non-audio signal representative of a desired audio signal from a stored audio file prior to a user's listening to a heard audio signal consequent upon combining of said phase-altered non-audio signal and said representative non-audio signal.

[0041] According to an eighth aspect of the present invention, there is provided a method of cancelling sound, comprising:

[0042] transmitting a desired audio signal towards a listener in a space and through air in said space containing said listener;

[0043] providing a non-audio signal representing unwanted ambient sound;

[0044] altering the phase of the non-audio signal to provide an at least partially cancelling, phase-altered non-audio signal; and

[0045] combining a non-audio signal representing the desired audio signal and the at least partially cancelling, phase-altered non-audio signal to provide an audio signal received by the listener.

[0046] Owing to those aspects of the invention, it is possible for the hearer to eliminate unwanted ambient sound, typically allowing him to hear more music at lower volume, so as not to damage his hearing. Further, it is possible for the hearer to enjoy the benefits of sound cancelling without the traditional limitations and also possible to include a number of groundbreaking features that improve a host device in terms of functionality and practical use, whilst improving user health and safety for headphone/earphone/headset dependent products and utilities.

[0047] Advantageously, embodiments of the invention allow headphone/earphone/headset users to choose the ambient sounds they want to hear, if any, and at a comfortable volume. Thus, a user is able to listen to music or make a call whilst the selective sound cancelling (SSC) eliminates, or significantly reduces, unwanted background sound, whilst allowing him to hear 'naturally required ambience' such as the sound of a doorbell or

an alarm. Furthermore, the user may be able to cancel, or significantly reduce, unwanted sound in any room, hall, or any venue. A typical application would be where a user wishes to cancel out any unwanted ambient sounds in a room while watching TV, listening to the radio or, even, just having a conversation. Embodiments of the invention can be incorporated into hardware such as 'phones and televisions (TVs) or used in web streaming websites (computers), or music players and systems.

[0048] Advantageously, embodiments of this invention allow a user finely to adjust the phase shift amount delay time, allowing him to make up for the natural delay time in processing the ambient audio signal, and allow compensation for the difference in the time it takes for sound to travel to the user's ear and the microphone. Preferably, some of the compressor settings may be available to a user and/or variable gate settings may be made available to the user to allow the user to determine when and how the gate acts, and for how long.

[0049] A preferred embodiment of the present invention uses the disclosed system/method and hardware and, in particular, an innovative algorithm that allows headphone users to benefit from advanced selective sound cancelling technology using standard generic headphones, whilst retaining audio awareness of immediate surroundings when so desired. Further, it allows sound cancelling to be enabled on a telephone (mobile or not) or anywhere, where two-way conversations are required—this being impossible whilst using current sound cancelling solutions.

[0050] It is possible to introduce a number of previously unavailable features to all audio devices, including mobile 'phones, mobile 'phones with music playing capacity, MP3 players, and websites (streaming), which increases the functionality and usefulness of the host device upon which embodiments of the invention are employed. Such embodiments enable the introduction of many groundbreaking benefits that are not otherwise available through the known art and which will be clearly discernable by those skilled in the art, as exemplified herein.

[0051] Embodiments of the present invention can be tailored to work on systems that depend on loud-speakers (or any other sound emitting devices). Examples of these are loud-speakers in cinemas, halls, and rooms—such as living rooms—meeting rooms and open-plan offices, in addition to open spaces. For these utilities, a number of high-quality microphones and sound reproduction loud-speakers may be used in this configuration depending on the type of sound that needs cancelling and the acoustic characteristics of the venue. The microphones are, typically, strategically placed near the sources of unwanted sound. The loud-speakers are placed around the hearer(s). Acoustic specialists can then calibrate the relevant embodiment of the invention so that it only cancels out the undesired sound and transmits both the desired sound and any desired ambient sound.

[0052] The following terms are well-known to those skilled in the art but a brief definition has been included for completeness.

[0053] 'Phase shift' is where the waveform of an analogue or digital signal is altered. A 180 degree phase shift produces an inverted waveform (typically) and a 90 degree phase shift causes a partial inversion.

[0054] A 'Compressor' is a processing algorithm that limits or expands the dynamics in signals. For example, a compressor can make loud sounds quieter and quiet sounds louder. There may be variable settings that affect the threshold before the compressor acts, such as, the speed and way it acts.

[0055] A 'Gate' is an algorithm that turns a signal on and/or off. Gates may also be triggered by external sources or devices, and feature a 'side chain' facility which means other processes (such as equaliser—EQ) can add to or control the behaviour of the gate.

[0056] 'EQ' is an abbreviation for equaliser, and is a process that allows the user to cut or boost particular frequencies in the frequency spectrum of a signal.

[0057] A 'Spectral Enhancer' is a process that improves the sonic quality of an audio file by adding to and manipulating harmonics, to produce a desired effect.

[0058] Other standard processes such as automatic gain control (to balance audio levels) may be used to condition the sounds to be heard—these are common procedures that have been omitted for the sake of simplicity from the embodiments shown in the accompanying drawings.

[0059] In order that the invention may be clearly and completely disclosed, embodiments thereof will now be described, by way of example only, with reference to the accompanying drawings, in which:

[0060] FIG. 1 is schematic drawing of a selective sound cancelling system;

[0061] FIG. 2 is a representation of an application interface through which the sound cancelling system can be operated;

[0062] FIG. 3A is schematic drawing of a simplified, selective, sound cancelling system;

[0063] FIG. 3B is a schematic drawing of a less simplified, selective, sound cancelling system;

[0064] FIG. 4 is a schematic drawing of a sound cancelling system and shows Sound Cancelling Mode A;

[0065] FIG. 5 is a schematic drawing of a sound cancelling system and shows Sound Cancelling Mode B;

[0066] FIG. 6 is a schematic drawing of a sound cancelling system and shows Sound Cancelling Mode C;

[0067] FIG. 7 is a schematic drawing of a sound cancelling system and shows Selective Sound Cancelling—Hearing Aids;

[0068] FIG. 8 is a schematic drawing of a sound cancelling system and shows Non-Microphone Sound Cancelling;

[0069] FIG. 9 is a schematic drawing of a sound cancelling system and shows Speaker-Based Selective Sound Cancelling;

[0070] FIGS. 10 and 11 are representations of respective simple application interfaces for operating the system;

[0071] FIGS. 12 to 15 are representations of advanced application interfaces for operating the system, the interfaces being where respectively ambience, compression, EQ and limiter settings can be fine-tuned;

[0072] FIGS. 16 and 17 are representations of application interfaces for operating the system and through which settings may be stored and recalled/imported, respectively;

[0073] FIG. 18 is a representation of a karaoke interface for operating the system;

[0074] FIG. 19 is a schematic drawing of a basic system applicable particularly to circumstances in which a listener to desired audio is using "enclosed" ear-or headphones;

[0075] FIG. 20 is a schematic drawing of a relatively simple example of a system within that basic concept;

[0076] FIG. 21 is a schematic drawing of a less simple example of a system within that basic concept;

[0077] FIG. 22 is a diagram of an "iPhone"® 'phone to which an app. (application) corresponding to the example of FIG. 21 has been uploaded; and

[0078] FIG. 23 is a schematic drawing of a simple system applicable particularly to circumstances in which a listener to desired audio using "enclosed" ear-or headphones wishes to exclude as much as feasible of ambient sound.

[0079] FIG. 1 shows an overall selective sound cancelling system to exemplify application of an embodiment of the invention to a modern music-capable mobile 'phone. This example is intended to explain the features employed to create a selective sound cancelling system and method and it should not be taken as a limitation of how and where the invention may be used. The specific example used is an Apple® iPhone® 'phone, as it is the simplest to describe. The built-in microphone of the mobile 'phone and/or an additional, external microphone can be used. The system may be configured to work with two or more microphones. The system can work with all headsets, including generic ear-bud headphones. However, it works best when used with 'enclosed' ear-or headphones. It should be mentioned that "FX" means "effects".

[0080] In the iPhone® example, a user may upload a software 'app.' (application) to his iPhone® mobile 'phone to enable the system and provide an application interface which operates the system. The user turns on the 'app.' and is presented with a screen, exemplified in FIG. 2. This Figure represents one simple interface, others being exemplified in FIGS. 10 and 11, which provides limited adjustment of sound cancelling parameters.

[0081] In a first mode of operation—in which no audio awareness is required—with the user wearing his ordinary headphones or earphones, the fader labelled 'calibrate' may be adjusted up or down until ambient sound is totally eliminated or vastly reduced. The user need not do anything else if he desires not to hear any ambient sound and he may simply press "play" on his usual music player. Following this type of adjustment only, the user may enjoy the music while hearing minimal or no background sound—which provides the advantage of the user's being able to hear more of the music at lower, safer levels of volume. If the 'phone is used to make a call in this first mode, then the 'app.' pauses until after the user has finished.

[0082] As a further possibility, the user may use the system to reduce or eliminate ambient sound without listening to music.

[0083] In a second mode, where audio awareness is required, in addition to the steps mentioned above with respect to the first mode, the user moves the fader labelled 'ambience' until he can hear his immediate surroundings at a desired level. The ambience fader allows the user to choose how much of the background sound he hears and allows him to hold a conversation, whilst eliminating or significantly reducing unwanted sound. The effects of both the ambience fader and the calibrate faders allow the user to listen to music, make 'phone calls, and hold live conversations, whilst retaining audio awareness of his immediate surroundings with active selective sound cancelling. In addition to the above, the user may use the 'app.' to 'noise cancel' the receiver sound during a phone call before that sound is transmitted to a recipient caller, which makes conversation for both parties easier and clearer at lower levels of volume—and allows easier and clearer conversations in areas of high background sound—for example when making a call in a windy environment.

[0084] In particular, the system allows a user to eliminate, practically all, if not all, background sound, even while listening to music, and still be able to hear things like a police siren, fire alarm, doorbell, or someone talking to him.

[0085] As exemplified in FIG. 3A, the system can allow a selected portion of the ambient sound received by the microphone 2 to be emitted by the headphones 11 in addition to the desired audio corresponding to the electrical output signal from the desired audio source 9 . FIG. 3B shows a less simplified schematic of selective sound cancelling; the system produces the effects described above by taking a signal from the microphone (Mic) of the mobile 'phone and phase-shifting it to produce a cancelling phase-altered signal. Typically this is done by altering the phase of the signal by 180 degrees (further allowing the user to fine-tune the phase shift + or −1 to 5 degrees in each direction). Those skilled in the art will know that, to produce a fully-cancelling phase-altered signal, some leeway is required in the number of degrees of phase shift, as the amount of shift required for correction is affected by numerous external factors, for example wind direction and speed. The phase-altered signal is processed and fed into the headphones, causing the sounds to 'cancel' each other out. However, as mentioned herein, this application of the system and method is only suitable when the user desires to cancel out all background ambience—as described in relation to the first mode of operation. For the second mode of operation, in addition to the above, the system takes a second identical signal from the Mic and produces a partially-cancelling phase-altered signal. Typically this is done by altering the phase of the signal by 45 degrees (although the amount of phase shift can be any partially-cancelling phase shift). This phase-altered signal is further processed to affect individual frequencies, volume, and other audio characteristics using EQ, compression, gates, and/or spectral enhancers, to achieve a desired effect. When this signal is fed into the headphones, no further noise cancelling is provided so that it remains audible. A user may now listen to music on his telephone while retaining a safe level of audio awareness.

[0086] Advantageously, this system allows a user to listen to his headphones in a number of useful modes such as: 'total sound cancelling' of all ambient sounds; 'alarm mode', where the user wishes to be alerted by, e.g. a fire alarm, a police siren or even when someone in the user's immediate vicinity talks to him; or 'audio awareness mode', where a user can control and turn down the background ambience he hears—allowing him to retain audio awareness of his immediate surroundings while listening to music or other audio, or when making or receiving a call on his music playing telephone.

[0087] An advanced interface is shown in FIGS. 12 to 15. Through the advanced interface the user has the option to edit key sound-cancelling parameters, allowing him effectively to choose the ambient background sound he wants to hear. For example, a user may want to 'noise cancel' the sound of train tracks whilst being able to hear a message over a train's public address system. A number of other innovative features are accessed via the advanced mode. FIG. 12 shows the range of features that can be accessed through the 'ambience' tab, for example, threshold, range and gain. FIG. 13 shows the range of features accessible through the 'compression' tab, for example, threshold, ratio, attack, release and gain. As to FIG. 14, this shows the features of EQ which may be adjusted through the 'EQ' tab, for example, hi, hi-mid, mid, lo-mid, and lo. A more sophisticated EQ may be used, for example, band filtering with an adjustable Q factor. FIG. 15 shows the range of features that are adjustable through the 'limiter' tab, for example min., max., sensitivity and gain.

[0088] FIGS. 16 and 17 show the interface through which advanced setting may be stored and recalled. A number of blank preset settings are provided for a user to save his own preferred settings. In addition, factory-set presets can be accessed from this interface.

[0089] FIGS. 4, 5 and 6 show Modes A to C for sound cancelling. Mode A relates to selective sound cancelling and shows one or more optional microphones. Mode B relates to maximum sound cancelling. In Mode C, the features enclosed in dotted lines can either be user-controlled via a software interface (see FIG. 1) or be included in an external controller unit.

[0090] An alternative embodiment is shown in FIG. 7, which exemplifies selective sound cancelling on hearing aids. Hearing aid users often find themselves caught in a vicious circle that results in damaging their hearing—it is instinctive for a user to turn up his hearing aid as his hearing deteriorates or when there is a lot of background sound. This can accelerates hearing loss. This embodiment solves this and other associated problems by cancelling out unwanted background ambient sound. This allows the user to hear more easily and at much safer levels of volume—preserving his hearing. In particular, hearing aids can be modified to include the present system. An optional external controller may be added to the hearing aid allowing a user to fine-tune what sound to cancel—to suit personal preference and requirements; in particular, components enclosed by the dotted lines could be incorporated in an external hardware module allowing users to fine tune the system to work according to their needs or preferences. Alternatively, a number of presets can be programmed into the hearing aid to work in various modes, e.g. selectively to reduce unwanted ambient sound such as machinery, motor cars, trains and/or aeroplanes.

[0091] FIG. 8 shows a further alternative embodiment which is a non-microphone version. The system may employ representations of common sound factors as presets—in effect allowing non-Mic devices to work with the system and without the use of a microphone, e.g. standard mp3 players. The 'app' for non-Mic platforms may contain a number of presets such as 'airplane mode', where the sound to be cancelled is engine or cabin sounds. A user also has the option of using and creating his own previously recorded sound samples to serve as counter signals for sound cancellation. For example, a user can record ambient sound using a mobile phone, save the file as an MP3 and import it to his MP3 player to be used as the sound source to cancel. In this mode the system allows a user to synthesise the sound he wants to cancel (i.e. not hear). Further to this, the system may employ an external microphone, allowing it to be used on any music player. There are a wide range of headphones that feature a built-in microphone, allowing mp3 hardware and all other audio delivery systems to work with the system.

[0092] A further alternative embodiment is shown in FIG. 9 and is directed towards speaker-based selective sound cancelling. Variations of the system can be used to suit specific applications. For example, a living room may require a 4-microphone configuration, whilst a large hall may require 30 microphones. An audio specialist would need to advise on placement of microphones and loud-speakers, and calibrate the system. This is useful for eliminating or significantly reducing unwanted background sound in rooms, in open-plan offices, in cinemas, concert halls, meeting rooms, and even at open-air events. The actual positioning of the microphones and loud-speakers would be determined after a full acoustic survey by an acoustic specialist who would also advise on the selective sound cancellation configuration and on the equipment required, e.g. the number of microphones and loud-speakers. A living room example, with a television set, three noise-receiving microphones and six noise-cancelling loud-speakers, is shown in FIG. 9. The system illustrated in FIG. 9 allows a user to tweak the remote control until optimal settings are achieved. In more complex situations, such as open plan offices or theatres, calibration would be completed by a trained specialist. In particular, it is noteworthy that a microphone is positioned in front of each source of unwanted sound in FIG. 9. Further, loud-speakers are positioned near the unwanted sound. Therefore, with the loud-speaker emitting a cancelling, phase-altered signal of that unwanted

sound, sound-cancelling is achieved. Although not shown, if a door bell were to chime, that sound would not be cancelled—unless desired by the user—and thereby would alert the user.

[0093] FIG. 18 shows a karaoke interface, particularly a multitrack recorder, in which mode a user can record vocals or any instrument such as guitar or drums. The user may adjust features, such as vocal, reverberation, delay and compressor, along with the master volume. Through this interface the user may adjust the sound-cancelling effects produced and save, import, export files/settings together with the normal play, etc. functions.

[0094] The following possible additional features of the system are now described in brief detail.

[0095] The 'app.' which implements the system can be designed to accept software plug-ins. The definition of a plug-in, as used on this system, is a separate piece of compatible software that can be used in conjunction with the system to add new features. Once a user has installed the 'app.' on his device he can download or otherwise acquire an 'add on' piece of software (plug-in) that is used by the main 'app.' to add or increase functionality. For example, a user can download a voice-changing or reverberation effect plug-in.

[0096] The system may feature a recording facility allowing a user to record meetings, messages, memos, or karaoke, with active selective sound cancelling. The recorded file can be played back on any music player and can be exported to any external device, e.g. PC or MP3 player and sent via mms or e-mail [and can be listened to by any caller].

[0097] The system may be incorporated into an external hardware unit that can feature a microphone to work on standard mp3 players, hearing aids or other devices that do not have microphones. Other hardware, such as external controllers (wired or wireless) in the form of faders, knobs, or touch screen, may be used to control the sound-cancelling parameters [and other features such as EQ, delay, reverberation etc.].

[0098] The system can be configured to make any mobile 'phone or (microphone equipped device) function as a powerful fully-featured hearing aid, with many advantages over traditional hearing aids stemming from selective sound-cancelling functionality.

[0099] The system may be configured to make any mobile 'phone or (microphone equipped device) into a powerful fully-featured listening device that can also transmit over the Internet (if the device is Internet-enabled) or as a normal telephone call.

[0100] Referring to FIG. 19, in this basic concept ambient sound is received by a microphone which transmits an electrical output signal representing the ambient sound to a treatment stage which receives another electrical output signal from a source of desired audio, such as music from a digital audio player (MP3). The two signals are combined, (with or without other treatment) and the resulting electrical signal is supplied to "enclosed" ear-or headphones and converted into audio heard by the wearer of the ear-or headphones.

[0101] In the example illustrated in FIG. 20, the electrical output signal from the microphone is gated such that, upon the ambient sound represented by that electrical output signal exceeding a predetermined volume, the gate begins to output an electrical signal to a mixer which also receives the electrical output signal from the source of desired audio (e.g. an MP3 player). The gate threshold is adjustable by the wearer using, for example, a slider. The wearer can also adjust the microphone volume by means of, for example, another slider. The electrical output signal from the mixer is supplied to the ear-or headphones the audio output volume from which is controllable by the user by means of, for example, a slider providing, in effect, a master volume control.

[0102] In an alternative example, the gate may have, in addition to the one threshold, an upper threshold, preferably settable independently of the lower threshold—or maybe even set by the manufacturer—to safeguard the wearer against extremely loud sound.

[0103] In the example shown in FIG. 21, a visual representation of the electrical output signal from the microphone and thus of the ambient sound is displayed. The electrical signal is again gated and the wearer can set the threshold. If the gate threshold is exceeded, the gate output may result in a vibratory, visual or audio alert if the wearer has chosen to switch an alert mode control "ON". The gated ambient sound volume represented by the gated electrical output signal can be adjusted by the wearer and the adjusted signal input to a mixer which receives the electrical output signal from the desired audio source (e.g. an MP3 player). The electrical output signal from the mixer is again fed to the ear-or headphones and a master volume control is again provided.

[0104] FIG. 22 shows an "iPhone®" touch-screen 'phone into which has been uploaded what we call our "Awareness!"™ app. The various features thereof will now be described progressing in a continuous loop around the 'phone from the top left-hand corner thereof. Tapping the left-hand end of the track scroll switches to the next previous track whilst tapping to the right-hand end thereof switches to the next following track. Alternatively, swiping left to right or right to left between those ends switches to the next previous or next following track. Tapping at the "essency"™ supplies help and other information. Tapping at "Awareness!"™ overrides the desired audio source and leaves simply the output derived from the ambient sound. If the LED (light emitting diode) representation is green, this means that the gate threshold has not been exceeded. If it turns red, this means that the threshold has been exceeded. In the event that, as is the case with this app. example being described, there are upper and lower gating thresholds, these may be set, as indicated by the lines "T-hi" and "T-lo", by carrying out a vertical pinching action on the screen. Shown on the screen is a 3D background with vertical lines which indicate presence and amplitude of ambient sound. The 'phone can be shaken to cause shuffling of the tracks in a random manner. The screen can be locked. Various settings, such as those for the gating threshold and the volumes can be set through sliders which are displayable upon the screen. The vibratory alert can be turned on or off. The microphone can be switched on or off by tapping or the finger can be held down to set the microphone output level and sensitivity. The iTunes library can be accessed. The screen can be swiped horizontally to raise or lower the player volume. Shown is screen 1 of the app. If it is inactivated for more than three seconds, the various controls disappear from the screen leaving only the 3D background, which is screen 2. With the player controls shown, pressing and holding the left or right button scrolls the music. Pressing the central button can produce "play", pressing it again produces "pause" and pressing and holding it returns the track to the beginning. If all three buttons are swiped, their positioning on the screen and their sizes become selectable, so that the user can resize them or drag them individually to anywhere on the screen.

[0105] Referring to FIG. 23, the inversion 5 , as regards both amplitude and frequency of the electrical output signal 3 from the microphone 2 , means that no ambient sound is mixed in at 8 , although of course that does not prevent a minor proportion of the ambient sound reaching the ears of the listener in other ways.

[0106] While the various embodiments of the invention have generally been described as treatments relating to the volumes of sound heard by the hearer, the invention is applicable also to treatment of frequencies of such sounds, or to both frequencies and volumes of such sounds.

US2017345406A1 METHOD, SYSTEM AND ITEM

Data originating from sources other than the EPO may not be accurate, complete, or up to date.

1 - 66 . (canceled)

67 . A method of enabling a hearer to hear desired sound while also being able to be aware of ambient sound, comprising providing a first non-sound signal representative of said desired sound, deriving a second non-sound signal representative of said ambient sound, combining the first and second non-sound signals in providing a combined non-sound signal, and converting said combined non-sound signal into sound.

68 . A method according to claim 67 and further comprising providing, for said second non-sound signal, a single threshold representative of a single threshold for the volume and/or frequency of said ambient sound, or providing for said second non-sound signal, both of a lower threshold and an upper threshold representative of a lower threshold and an upper threshold for the volume and/or frequency of said ambient sound.

claim 67

69 . A method according to claim 68, and further comprising variably adjusting said single threshold, or said lower threshold and/or said upper threshold for said second non-sound signal.

claim 68

70 . A method according to claim 69, wherein said adjusting is automatic.

claim 69

71 . A method according to claim 68, and further comprising emitting an alert upon exceeding of said single threshold, or of said lower threshold and/or said upper threshold for said second non-sound signal.

claim 68

72 . A method according to claim 71, wherein said alert comprises said ambient sound becoming available to said hearer.

claim 71

73 . A method according to claim 67, and further comprising variably adjusting said second non-sound signal to represent adjustment of the volume and/or frequency of said ambient sound.

claim 67

74 . A method according to claim 67, wherein the sound into which said combined non-sound signal is converted is transmitted to a listening person in a space and through air in said space and containing said person.

claim 67

75 . A method according to claim 67, wherein said second non-sound signal is representative of said ambient sound without phase shift.

claim 67

76 . A method according to claim 68, wherein a delay period is introduced between said second non-sound signal attaining a threshold level of said volume and/or frequency of said single threshold, or said lower threshold, or said upper threshold, and the converting of said combined single into sound.

claim 68

77 . A system for enabling a hearer to hear desired sound while also being able to be aware of ambient sound, comprising a first outputting device adapted to output a first non-sound signal representative of said desired

sound, a second outputting device adapted to output a second non-sound signal representative of said ambient sound, a combining arrangement adapted to combine the first and second non-sound signals to provide a combined non-sound signal, and a converting device adapted to emit sound converted from said combined non-sound signal.

78 . A system according to claim 77, wherein said second outputting device is adapted to provide a single threshold or both of a lower threshold and an upper threshold for said second non-sound signal.

claim 77

79 . A system according to claim 78, wherein said second outputting device is variably adjustable to adjust variably said single threshold, or said lower threshold and/or said upper threshold.

claim 78

80 . A system according to claim 78, and further comprising an alerting device able to communicate with said second outputting device to alert to exceeding of said single threshold, or of said lower threshold and/or said upper threshold.

claim 78

81 . A system according to claim 80, wherein said alerting device is adapted to cause said hearer to become aware of said ambient sound.

claim 80

82 . A system according to claim 77, wherein said second outputting device is variably adjustable to adjust variably said second non-sound signal'.

claim 77

83 . A system according to claim 78, and further comprising an arrangement adapted to introduce a delay period between said second non-sound signal attaining a threshold level of said volume and/or frequency of said single threshold, or said lower threshold, or said upper threshold, and the converting of said combined signal into sound.

claim 78

84 . A method comprising combining a non-sound signal with a derived signal derived from said non-sound signal by at least partial phase shift.

85 . A method according to claim 84, wherein said non-sound signal and said derived signal are combined with a second derived signal derived from said non-sound signal by substantially 180° phase shift.

claim 84

86 . A system comprising an outputting device adapted to output a non-sound signal, a deriving device adapted to derive a derived signal from said non-sound signal by at least partial phase shift, and a combining arrangement adapted to combine those two signals.

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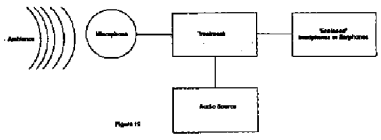
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最終頁に続く

(54)【発明の名称】方法、システム及び機器

(57)【要約】
周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにする方法であって、前記所望の音を表す第１の非音声信号を提供すること、前記周囲音から第２の非音声信号を得ること、第３の非音声信号を提供するため前記第１及び第２の非音声信号を合成すること、及び、前記第３の非音声信号を音に変換することを含む方法。



【特許請求の範囲】
【請求項１】
周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにする方法であって、前記所望の音を表す第１の非音声信号を提供すること、前記周囲音から第２の非音声信号を得ること、前記第１及び第２の非音声信号を、第３の非音声信号を提供するために合成すること、及び、前記第３の非音声信号を音に変換することを含む方法。
【請求項２】
前記周囲音の音量及び／又は周波数の調節を表すために前記第２の非音声信号を調節することをさらに含む、請求項１に記載の方法。
【請求項３】
前記周囲音の音量及び／又は周波数に対する低閾値を表す低閾値を前記第２の非音声信号に対して設けることをさらに含む、請求項１又は２に記載の方法。
【請求項４】
前記周囲音の音量及び／又は周波数に対する高閾値を表す高閾値を前記第２の非音声信号に対して設けることをさらに含む、請求項１ないし３のいずれか１項に記載の方法。
【請求項５】
前記第２の非音声信号に対して、前記低閾値及び／又は前記高閾値を調節することをさらに含む、請求項３又は４に記載の方法。
【請求項６】
前記第２の非音声信号に対して、前記低閾値及び／又は前記高閾値を超える際に警告を発することをさらに含む、請求項３ないし５のいずれか１項に記載の方法。
【請求項７】
前記第３の非音声信号から変換された音以外の前記周囲音が前記聞き手に聞こえるのを大幅に抑えることをさらに含む、請求項１ないし７のいずれか１項に記載の方法。
【請求項８】
周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにするシステムであって、
前記所望の音を表す第１の非音声信号を出力する第１の出力装置と、
前記周囲の音から得られた第２の非音声信号を出力する第２の出力装置と、
前記第１及び第２の非音声信号を合成する合成装置と、
前記第３の非音声信号を変換して得られた音を発する変換装置と、
を備えるシステム。
【請求項９】
前記第２の出力装置は、前記第２の非音声信号を調節するために調節可能である、請求項８に記載のシステム。
【請求項１０】
前記第２の出力装置は、前記第２の非音声信号に対して低閾値を設ける、請求項８又は９に記載のシステム。
【請求項１１】
前記第２の出力装置は、前記第２の非音声信号に対して高閾値を設ける、請求項８ないし１０のいずれか１項に記載のシステム。
【請求項１２】
前記第２の出力装置は、前記低閾値及び／又は前記高閾値を調節するために調節可能である、請求項１０又は１１に記載のシステム。
【請求項１３】
前記第１の閾値及び／又は前記第２の閾値を超えたことを警告するように前記第２の出力装置と通信することができる警報装置をさらに備える、請求項１０ないし１２のいずれか１項に記載のシステム。
【請求項１４】
前記変換装置によって発せられた音以外の前記聞き手に聞こえる前記周囲音を大幅に抑

える遮音装置をさらに備える、請求項８ないし１３のいずれか１項に記載のシステム。
【請求項１５】
部分的に音をキャンセルする方法であって、
聞きたい所望の音声信号を表す第１の非音声信号を提供すること、
部分的にキャンセルされる周囲音声信号を表す第２の非音声信号を提供すること、及び、
聞こえる音声信号を提供するため、前記第１及び第２の非音声信号を合成すること、
を含む方法。
【請求項１６】
選択的サウンドキャンセリングの形式によって必要な音と不必要な音とを区別する、請求項１５に記載の方法。
【請求項１７】
前記第２の非音声信号は、提供される前に予め設定されている、請求項１５又は１６に記載の方法。
【請求項１８】
ユーザが、前記予め設定された非音声信号を異なる複数の予め設定された非音声信号から選択することによって、キャンセルする音を選択する、請求項１７に記載の方法。
【請求項１９】
前記予め設定された非音声信号は、１つ以上のサウンドキャンセリングパラメータを含む、請求項１８に記載の方法。
【請求項２０】
ユーザが、１つ以上のサウンドキャンセリングパラメータを変えることによって、キャンセルする音を選択する、請求項１６に記載の方法。
【請求項２１】
前記１つ以上のサウンドキャンセリングパラメータは、
位相シフトの量、
時間遅延、
それぞれの周波数、
音量、
イコライザ（高、高－中、中、低－中、及び／又は低）、
コンプレッション、
ゲーティング、
スペクトルエンハンスメント、
閾値、
範囲、
ゲイン、
比率、
アタック、
リリース、
最小、
最大、
感度、
ボーカル、又は、
リバーブ、
若しくは、それらの任意の組み合わせである、請求項１９又は２０に記載の方法。
【請求項２２】
部分的にキャンセルされ、位相が変更された周囲音を生成するための位相シフトは、約１度ないし約１７９度の間であり、好ましくは４５度である、請求項１５ないし２１のいずれか１項に記載の方法。
【請求項２３】

前記聞こえる音声信号をある空間のリスナに、前記リスナを含む前記空間における空気を介して伝達することをさらに含む、請求項１５ないし２２のいずれか１項に記載の方法。

【請求項２４】
部分的に音をキャンセルする方法であって、
聞きたい所望の音声信号を表す第１の非音声信号を提供すること、
周囲音声信号から得られた第２の非音声信号を提供すること、
実質的完全キャンセル用位相変更非音声信号及び部分的キャンセル用位相変更非音声信号を提供するために、前記第２の非音声信号の位相を変えること、及び、
聞こえる音声信号を提供するために、前記第１の非音声信号と、前記実質的完全キャンセル用位相変更非音声信号と、前記部分的キャンセル用位相変更非音声信号とを合成すること、
を含む方法。

【請求項２５】
選択的サウンドキャンセリングの形式によって必要な音と不必要な音とを区別する、請求項２４に記載の方法。

【請求項２６】
前記実質的完全キャンセル用位相変更非音声信号及び前記部分的キャンセル用位相変更非音声信号の少なくとも一方は、提供する前に予め設定されている、請求項２４又は２５に記載の方法。

【請求項２７】
ユーザが、前記実質的完全キャンセル用位相変更非音声信号及び前記部分的キャンセル用位相変更非音声信号の前記少なくとも一方を、異なる複数の予め設定された非音声信号の範囲から選択することにより、キャンセルする音を選択する、請求項２６に記載の方法。

【請求項２８】
前記複数の予め設定された非音声信号のそれぞれは、１つ以上のサウンドキャンセリングパラメータを含む、請求項２６又は２７に記載の方法。

【請求項２９】
ユーザが、１つ以上のサウンドキャンセリングパラメータを変えることにより、キャンセルする音を選択する、請求項２５に記載の方法。

【請求項３０】
前記１つ以上のサウンドキャンセリングパラメータは、
位相シフトの量、
時間遅延、
それぞれの周波数、
音量、
イコライザ（高、高－中、中、低－中、及び／又は低）、
コンプレッション、
ゲーティング、
スペクトルエンハンスメント、
閾値、
範囲、
ゲイン、
比率、
アタック、
リリース、
最小、
最大、
感度、

ボーカル、又は
リバーブ、
若しくは、それらの任意の組み合わせである、請求項２８又は２９に記載の方法。
【請求項３１】
前記聞こえる音声信号をある空間のリスナに、前記リスナを含む前記空間における空気を介して伝達することをさらに含む、請求項２４ないし３０のいずれか１項に記載の方法。
【請求項３２】
前記部分的キャンセル用非音声信号を生成するための位相シフトは、およそ１度とおよそ１７９度の間であり、好ましくは４５度である、請求項２４ないし３１のいずれか１項に記載の方法。
【請求項３３】
添付の図面を参照して上述した任意の例に実質的に応じたサウンドキャンセリング方法。
【請求項３４】
部分的に音をキャンセルするシステムであって、
聞きたい所望の音声信号を表す第１の非音声信号を出力する第１の出力装置と、
部分的にキャンセルされる、位相が変更された周囲音声信号を表す第２の非音声信号を提供する提供装置と、
第３の非音声信号を提供するため、前記第１及び第２の非音声信号を合成する合成装置と、
前記第３の非音声信号を聞こえる音声信号へと変換する第２の出力装置と、
を備えるシステム。
【請求項３５】
前記提供装置は、予め設定された前記第２の非音声信号を提供する、請求項３４に記載のシステム。
【請求項３６】
前記提供装置は、選択的に、他の異なる予め設定された非音声信号を提供する、請求項３５に記載のシステム。
【請求項３７】
前記提供装置は、１つ以上のマイクロホンを備える、請求項３４に記載のシステム。
【請求項３８】
前記第１の出力装置は、ＭＰ３プレーヤ、ＣＤプレーヤ又はマイクロホンを含む、請求項３４ないし３７のいずれか１項に記載のシステム。
【請求項３９】
前記提供装置は、関連するプログラムを有するプロセッサを含む、請求項３４ないし３８のいずれか１項に記載のシステム。
【請求項４０】
前記合成装置は、関連するプログラムを有するプロセッサを含む、請求項３４ないし３８のいずれか１項に記載のシステム。
【請求項４１】
前記提供装置及び前記合成装置は、関連するプログラムを有するプロセッサを含む、請求項３４ないし３８のいずれか１項に記載のシステム。
【請求項４２】
前記第２の出力装置は、１つ以上のスピーカ、ヘッドホン又はオーディオファイルを記録する装置を含む、請求項３４ないし４１のいずれか１項に記載のシステム。
【請求項４３】
選択的サウンドキャンセリングを提供して必要な音と不要な音とを区別することができるように構成された、請求項３４ないし４２のいずれか１項に記載のシステム。
【請求項４４】

部分的に音をキャンセルするシステムであって、
聞きたい所望の音声信号を表す第１の非音声信号を出力する第１の出力装置と、
周囲音声信号から得られた第２の非音声信号を提供する提供装置と、
実質的完全キャンセル用位相変更非音声信号及び部分的キャンセル用位相変更非音声信号を提供するために、前記第２の非音声信号の位相を変えるフェーズ変更装置と、
第３の非音声信号を提供するため、前記第１の非音声信号と、前記実質的完全キャンセル用非音声信号と、前記部分的キャンセル用非音声信号とを合成する合成装置と、
前記第３の非音声信号を聞こえる音声信号へと変換する第２の出力装置と、
を備えるシステム。
【請求項４５】
前記提供装置は、前記実質的完全キャンセル用非音声信号及び前記部分的キャンセル用非音声信号の一方又は両方を、予め設定された非音声信号として提供する、請求項４４に記載のシステム。
【請求項４６】
前記提供装置が、複数の予め設定された非音声信号からの選択を可能とする、請求項４５に記載のシステム。
【請求項４７】
前記提供装置は、１つ以上のマイクロホンを含む、請求項４４に記載のシステム。
【請求項４８】
前記第１の出力装置は、デジタルオーディオプレーヤ、ＣＤプレーヤ又はマイクロホンを含む、請求項４４ないし４７のいずれか１項に記載のシステム。
【請求項４９】
前記フェーズ変更装置は、関連するプログラムを有するプロセッサを含む、請求項４４ないし４８のいずれか１項に記載のシステム。
【請求項５０】
前記合成装置は、関連するプログラムを有するプロセッサを含む、請求項４４ないし４８のいずれか１項に記載のシステム。
【請求項５１】
前記フェーズ変更装置及び前記合成装置は、関連するプログラムを有するプロセッサを含む、請求項４４ないし４８のいずれか１項に記載のシステム。
【請求項５２】
前記第２の出力装置は、１つ以上のスピーカ、ヘッドホン又はオーディオファイルを記録する装置を含む、請求項４４ないし５１のいずれか１項に記載のシステム。
【請求項５３】
選択的サウンドキャンセリングを提供して必要な音と不要な音とを区別することができるように構成された、請求項４４ないし５２のいずれか１項に記載のシステム。
【請求項５４】
添付の図面を参照して上述した任意の例に実質的に応じたサウンドキャンセリングのシステム。
【請求項５５】
電話、携帯電話、テレビ、カラオケマシン、コンピュータ、Ｈｉ－Ｆｉ、サラウンドサウンドシステム、ゲーム機及び車載オーディオ装置からなる群のいずれか１つからなる機器を備える装置であって、前記機器に含まれる１つのマイクロホンのみを備え、ユーザが聞くオーディオファイルを再生する機能を有し、
前記マイクロホンは、周囲音を検出することができ、前記機器は、
位相変更非音声信号を提供するために、前記マイクロホンによって検出された音の位相を変える位相変更装置と、
前記位相変更非音声信号と、記録されたオーディオファイルから所望の音声信号を表す代表的な非音声信号とを、前記位相変更非音声信号及び前記代表的な非音声信号を合成した結果として聞こえる音声信号をユーザが聞く前に合成する合成装置と、

を備える装置。
【請求項５６】
前記位相変更装置は、前記マイクロホンによって検出された音の位相を変更して、前記位相変更非音声信号を、部分的キャンセル用位相変更非音声信号の形式で提供する、請求項５５に記載の装置。
【請求項５７】
前記フェーズ変更装置は、前記マイクロホンによって検出された音の位相を変えて、実質的完全キャンセル用位相変更非音声信号も提供する、請求項５６に記載の装置。
【請求項５８】
添付の図面を参照して上述した任意の例に実質的に応じた、携帯電話、テレビ又は補聴器を含む機器を含む装置。
【請求項５９】
音をキャンセルする方法であって、
所望の音声信号を、ある空間のリスナに向かって、当該リスナを含む前記空間における空気を介して伝達すること、
不要な周囲音を表す非音声信号を提供すること、
少なくとも部分的なキャンセル用位相変更非音声信号を提供するために前記非音声信号の位相を変えること、及び、
前記リスナによって受信される音声信号を提供するために、前記所望の音声信号及び前記少なくとも部分的なキャンセル用位相変更非音声信号を合成すること、
を含む方法。
【請求項６０】
選択的サウンドキャンセリングの形式によって必要な音と不必要な音とを区別する、請求項５９に記載の方法。
【請求項６１】
不要な周囲音を表す前記第２の非音声信号は、提供される前に予め設定されている、請求項５９又は６０に記載の方法。
【請求項６２】
ユーザが、前記予め設定された非音声信号を、異なる複数の予め設定された非音声信号の範囲から選択することによりキャンセルする音を選択する、請求項６１に記載の方法
【請求項６３】
前記予め設定された非音声信号は、１つ以上のサウンドキャンセリングパラメータを含む、請求項６２に記載の方法。
【請求項６４】
ユーザが、１つ以上のサウンドキャンセリングパラメータを変えることにより、キャンセルする音を選択する、請求項６０に記載の方法。
【請求項６５】
前記１つ以上のサウンドキャンセリングパラメータは、
位相シフトの量、
時間遅延、
それぞれの周波数、
音量、
イコライザ（高、高－中、中、低－中、及び／又は低）、
コンプレッション、
ゲーティング、
スペクトルエンハンスメント、
閾値、
範囲、
ゲイン、
比率、

アタック、
リリース、
最小、
最大、
感度
ボーカル、又は、
リバーブ、
若しくは、それらの任意の組み合わせである、請求項６３又は６４に記載の方法。
【請求項６６】
前記空間とは、部屋、講堂及び野外からなる群のうちの何れかである、請求項５９ないし６５のいずれか１項に記載の方法。
【発明の詳細な説明】
【技術分野】
【０００１】
本発明は、周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにする方法及びシステムに関する。特に、本発明は、サウンドキャンセリング、サウンドキャンセリングのためのシステム及び、サウンドキャンセリング装置を備えた機器に適用可能であるが、それらに限定されるものではない。
【背景技術】
【０００２】
近年、いわゆる「クローズド型」すなわち、音響的に周囲の音から著しく隔離されたインナーイヤホンやヘッドホンを通して携帯電話や音楽プレーヤからの音楽やその他音声を聞くことが常識となりつつある。しかしこれには危険警報等、聞き手が確実に聞くことを望むであろう周囲音を認識できないという欠点がある。
【０００３】
当該技術分野において多くのサウンドキャンセリング方法／アプリケーションが知られているが、概して同じ原理に基づいている。つまり、サウンドキャンセリングは、デジタル又はアナログの音声信号のコピーを生成し、当該信号の極性を反転させて「負」の信号を生成し、その後、この２つの信号を合成して完全なサウンドキャンセリング効果を実現することにより得られるものである。上記既知の技術には、このサウンドキャンセリング処理では必要な音と不必要な音とを区別することができないという固有の問題がある。既存の方法は、全ての周囲音が不要であると自動的に想定するためユーザが会話をしたり周囲で何が起こっているかを聞いたりする必要がある場合のある携帯電話や補聴器におけるサウンドキャンセリングシステムには適さない。今まで、例えば、ヘッドホン（又は、イヤホンやヘッドセット）を装着したユーザを音響的に隔離するための要件によって、サウンドキャンセリングの利用は限定的であった。現在、サウンドキャンセリング装置には、高価なヘッドホン／イヤホンが必要である。更に、補聴器やｍｐ３プレーヤ等の音楽プレーヤのユーザは、不要な周囲音を遮断するために、無認識にイヤホンの音量を上げている。これは、ユーザの聴覚を損傷しかねず、長期にわたる使用は、永続的な難聴につながりかねない。
【発明の概要】
【発明が解決しようとする課題】
【０００４】
多くのソフトウェア又はハードウェアの形で実現可能な本発明は、従来技術の欠点をなくする又は、著しく低減させることを意図するものである。
【０００５】
本明細書において、「非音声」という表現は通常、電気伝導等の空気を介する音波以外を用いた伝達に関する。
【課題を解決するための手段】
【０００６】

本発明の第１の態様によると、周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにする方法であって、前記所望の音を表す第１の非音声信号を提供すること、前記周囲の音から第２の非音声信号を得ること、第３の非音声信号の提供するために前記第１及び第２の非音声信号を合成すること、及び、前記第３の非音声信号を音に変換することを含む方法が提供される。

【０００７】

本発明の第２の態様によると、周囲音も認識可能でありながら所望の音を聞くことを聞き手ができるようにするシステムであって、前記所望の音を表す第１の非音声信号を出力する第１の出力装置と、前記周囲の音から得た第２の非音声信号を出力する第２の出力装置と、前記第１及び第２の非音声信号を合成する合成装置と、第３の非音声信号を変換して得られた音を発する変換装置と、を備えるシステムが提供される。

【０００８】

本発明のこれらの態様により、特に、聞き手が所望の音を聞くために空気を介して当該聞き手の耳に伝達される周囲音の多くを意図的に遮断した状況において、当該聞き手が所望の音を聞けるようにするとともに、周囲音も認識することができる。

【０００９】

本発明の好適な実施形態を備えた「クローズド型」ヘッドホンのユーザは、オープンなマイクロホンによって拾われてから聞こえる周囲音を聞き、そのレベルを制御し／高めながら、任意のオーディオプレーヤ又は携帯電話からの音声を聞くことができる。本実施形態は、ミキシング、ゲーティング、イコライジング、コンプレッション、拡張及びスペクトルエンハンシングのいくつか又は全てを用いる多くの構成において採用され得るオーディオ処理の組み合わせからなる。ユーザは、設定された閾値（ゲート）よりも高いものは設定された最大レベル（リミッタ／コンプレッサ）で全て聞こえるが、それよりも低いものは何も聞こえない。それら設定はユーザにより調節、ＡＰＰの提供者による事前設定、所定の期間にわたって測定された平均的又は他の周囲の音の大きさを表すレベルへの自動設定が可能である。本実施形態は、好ましくはユーザに対して以下のことを可能とする。

・ヘッドホンを使用中に周囲音が聞こえること

・周囲音に対して最大のオーディオレベルを設定すること

・ユーザにより、事前に、又は自動的に設定した閾値を超える周囲音だけが聞こえること

・閾値を超えた場合、振動だけのアラーム、ホスト装置（例えば、ＭＰ３プレーヤ）上の視覚的メッセージ又は、マイクロホンによって拾われる音声（周囲）を聞くかを選択すること

・ボタンを押すことでいつでもマイクロホンの信号を聞けること

・マイクロホンの感度を調節すること

【００１０】

本実施形態は、例えば、アラームだけ、音量を上げた音声（例えば、大きい音）等の多くの目的に供するバージョンを生成する任意の数の代替的方法で組み立てられ構成される又は、特定の音や周波数を強調する又は排除する、あるいは含めるように処理される。

【００１１】

例えば、本実施形態は、設定された閾値を超える場合に、ＭＰ３プレーヤから聞こえる音楽を小さくし、ユーザがマイクロホンからの信号を聞けるように構成することができる。又は、音楽を停止又は一旦停止したり、単に音楽と同時にマイクロホンの信号を聞いたりするようにすることができる。

【００１２】

本発明の第３の態様によると、部分的に音をキャンセルする方法であって、

聞きたい所望の音声信号を表す第１の非音声信号を提供すること、

部分的にキャンセルする周囲音声信号を表す第２の非音声信号を提供すること、及び、

聞こえる音声信号を提供するため、前記第１及び第２の非音声信号を合成することを含む方法が提供される。

【００１３】
本発明の第４の態様によると、部分的に音をキャンセルする方法であって、
聞きたい所望の音声信号を表す第１の非音声信号を提供すること、
周囲音声信号から得られた第２の非音声信号を提供すること、
実質的完全キャンセル用位相変更非音声信号及び部分的キャンセル用位相変更非音声信号を提供するために、前記第２の非音声信号の位相を変えること、及び、
聞こえる音声信号を提供するために、前記第１の非音声信号、前記実質的完全キャンセル用位相変更非音声信号と、前記部分的キャンセル用位相変更非音声信号とを合成することを含む方法が提供される。
【００１４】
本発明の第５の態様によると、部分的に音をキャンセルするシステムであって、
聞きたい所望の音声信号を表す第１の非音声信号を出力する第１の出力装置と、
部分的キャンセル用位相変更周囲音声信号を表す第２の非音声信号を提供する提供装置と、
第３の非音声信号を提供するため、前記第１及び第２の非音声信号を合成する合成装置と、
前記第３の非音声信号を聞こえる音声信号へと変換する第２の出力装置と、を備えるシステムが提供される。
【００１５】
本発明の第６の態様によると、部分的に音をキャンセルするシステムであって、
聞きたい所望の音声信号を表す第１の非音声信号を出力する第１の出力装置と、
周囲の音声信号から得られた第２の非音声信号を提供する提供装置と、
実質的完全キャンセル用位相変更非音声信号及び、部分的キャンセル用位相変更非音声信号を提供するために、前記第２の非音声信号の位相を変えるフェーズ変更装置と、
第３の非音声信号を提供するため前記第１の非音声信号、前記実質的完全キャンセル用非音声信号及び、前記部分的キャンセル用非音声信号を合成する合成装置と、
前記第３の非音声信号を聞こえる音声信号へと変換する第２の出力装置と、を備えるシステムが提供される。
【００１６】
本発明の第７の態様によると、電話、携帯電話、テレビ、カラオケマシン、コンピュータ、Ｈｉ－Ｆｉ、サラウンドサウンドシステム、ゲーム機及び、車載オーディオ装置からなる群のいずれか１つからなる機器を含む装置であって、当該装置は、前記機器に含まれる１つのマイクロホンだけを備え、ユーザが聞くオーディオファイルを再生する性能を有し、
前記マイクロホンは、周囲音を検出することができ、前記機器は、
位相変更非音声信号を提供するために、前記マイクロホンによって検出された音の位相を変える位相変更装置と、
前記位相変更非音声信号と、記憶されたオーディオファイルから所望の音声信号を表す代表的な非音声信号を、前記位相変更非音声信号及び前記代表的な非音声信号を合成する結果として得られる聞こえる音声信号をユーザが聞く前に合成する合成装置と、を備える装置が提供される。
【００１７】
本発明の第８の態様によると、音をキャンセルする方法であって、
所望の音声信号を、ある空間におけるリスナに向かって、当該リスナを含む当該空間の空気を介して伝達すること、
不要な周囲音を表す非音声信号を提供すること、
少なくとも部分的キャンセル用位相変更非音声信号を提供するために前記非音声信号の位相を変えること、及び、
前記リスナによって受信される音声信号を提供するために、前記所望の音声信号及び前記少なくとも部分的キャンセル用位相変更非音声信号を合成すること、を含む方法が提供

される。

【００１８】

本発明のこれらの態様によると、聞き手は、不要な周囲音を排除することが可能となり、典型的には、聞き手がより低い音量で音楽を聞くことができるようにして、聴力を損なうことがないようになる。更に、聞き手は、従来の制限なしにサウンドキャンセリングの効果を得ることができる。また、ヘッドホン／イヤホン／ヘッドセットに依存する製品及び設備に対するユーザの健康及び安全性を向上させながら機能性及び実用性においてホスト装置を向上させる多くの画期的な特徴を含むことも可能となる。

【００１９】

本発明の実施形態により、ヘッドホン／イヤホン／ヘッドセットのユーザは、もしあれば、快適な音量で聞きたい周囲音を選択できるようになるという利点がある。そのためユーザは、選択的サウンドキャンセリング（ＳＳＣ）で不要な背景音を排除又は著しく低減させて、呼び鈴やアラームの音等「自然に要求される周囲音」を聞きながら、音楽を聞いたり、電話をかけたりすることが可能である。更に、ユーザは、任意の部屋、ホール又は任意の場所における不要な音をキャンセルする又は著しく低減させることができてもよい。典型的な適用としては、ユーザが部屋の中でテレビを見たり、ラジオを聞いたり、ただ会話していたりする間の不要な周囲音をキャンセルしたい場合等がある。本発明の実施形態は、携帯電話及びテレビ（ＴＶ）等のハードウェアに組み込むことができ、ウェブストリーミングのウェブサイト（コンピュータ）又は音楽プレーヤ及びシステムにおいて用いることができる。

【００２０】

本発明の実施形態によりユーザは、位相シフト量や遅延時間を細かく調節することが可能となり、周囲音の信号を処理する際の必然的な遅延時間を補うこと可能となり、音がユーザの耳及びマイクロホンに移動する際にかかる時間の差を埋め合わせることができるという利点がある。いくつかのコンプレッサの設定がユーザによって利用可能で、及び／又は、可変のゲート設定がユーザによって利用可能となり、ユーザが、いつ、どのように、そして、どのくらいの期間ゲートが動作するのかを決定することができるようになることが好ましい。

【００２１】

本発明の好適な実施形態は、開示されたシステム／方法及びハードウェア、特に、ヘッドホンユーザが標準的な一般のヘッドホンを用いて必要な時に身近な周囲の音を認識しながら、高度な選択的サウンドキャンセリング技術による利点を得ることを可能とする進歩的なアルゴリズムを用いる。更に、現在のサウンドキャンセリングソリューションでは不可能な、電話（携帯電話であってもなくても）又は、二方向の会話が要求される任意の状況でのサウンドキャンセリングを可能とする。

【００２２】

携帯電話、音楽プレーヤ機能を有する携帯電話、ＭＰ３プレーヤ、ウェブサイト（ストリーミング）等を含むあらゆる音響機器に対し、従来なかった多くの特徴を導入することが可能であり、それが本発明の実施形態が採用されるホスト装置の機能性及び利便性を向上する。そのような実施形態は、本明細書に例示するように、公知の技術を通しては利用できないものであるが、当業者であれば明確に認められる多くの画期的な利点を導入することを可能とする。

【００２３】

本発明の実施形態は、スピーカ（又は、他の音響放射装置）を利用したシステム上で作動するように設計することができる。これらの例は、映画館、ホール及び居間等の部屋、会議室、間仕切りのないオフィス並びに屋外等におけるスピーカである。これらの施設については、キャンセルする必要のある音のタイプ及び会場の音響特性によっては、この構成で多くの高性能なマイクロホン及び音響再生スピーカを用いてもよい。典型的には、マイクロホンは、不要な音の音源の近くに意図的に配置される。スピーカは、聞き手の周りに配置される。そして、音響専門家は、本発明の実施形態に関してキャリブレーションを

行い、不要な音だけをキャンセルし、所望の音と所望の周囲音の両方が伝達されるようにする。

【００２４】

以下の用語は、当業者には公知であるが、完璧を期して大まかな定義を含める。

【００２５】

「位相シフト」は、アナログ又はデジタル信号を変えることである。１８０度の位相シフトにより、反転した波形（典型的に）が生成され、９０度の位相シフトにより、部分的な反転が起きる。

【００２６】

「コンプレッサ」は、信号のダイナミクスを制限したり拡張したりする処理アルゴリズムである。例えば、コンプレッサは、大きな音を静かな音にし、静かな音を大きな音にすることができる。速度や動作の形態等、コンプレッサが動作する前に閾値に影響を与える可変設定があってもよい。

【００２７】

「ゲート」は、信号をオン／又はオフにするアルゴリズムである。ゲートは、外部のソース又は装置によって引き起こされてもよく、ゲートの振る舞いを追加する又は制御することができる他のプロセス（例えば、イコライザーＥＱ等）を意味する「サイドチェーン」設備を特徴とすることもできる。

【００２８】

「ＥＱ」は、イコライザの略語であり、ユーザが信号の周波数スペクトルにおいて特定の周波数をカットする又はブーストすることができるようにする処理である。

【００２９】

「スペクトルエンハンサ」は、所望の効果を得るため、高周波を追加操作することによってオーディオファイルの音質を向上させる処理である。

【００３０】

自動利得制御（オーディオレベルのバランスを保つため）等の他の標準的な処理により聞こえる音を調整してもよい。これらは、簡潔性のために、添付の図面において示した実施形態からは省略されている一般的な手順である。

【図面の簡単な説明】

【００３１】

本発明を明瞭かつ完全に開示するために、例示に過ぎないがその実施形態を以下の添付図面を参照してここに説明する。

【図１】選択的サウンドキャンセリングシステムの概略図である。

【図２】サウンドキャンセリングシステムを操作することができるアプリケーションインターフェースの図である。

【図３Ａ】簡略型選択的サウンドキャンセリングシステムの概略図である。

【図３Ｂ】簡略度合いを抑えた選択的サウンドキャンセリングシステムの概略図である。

【図４】サウンドキャンセリングシステムの概略図であり、サウンドキャンセリングモードＡを示す。

【図５】サウンドキャンセリングシステムの概略図であり、サウンドキャンセリングモードＢを示す。

【図６】サウンドキャンセリングシステムの概略図であり、サウンドキャンセリングモードＣを示す。

【図７】サウンドキャンセリングシステムの概略図であり、選択的サウンドキャンセリング－補聴器を示す。

【図８】サウンドキャンセリングシステムの概略図であり、非マイクロホンサウンドキャンセリングを示す。

【図９】サウンドキャンセリングシステムの概略図であり、スピーカに基づいた選択的サウンドキャンセリングを示す。

【図１０－１１】それぞれ、システムを操作するための単純なアプリケーションインター

フェースの図である。
【図１２－１５】システムを操作するための高度なアプリケーションインターフェースの図であり、当該インターフェースはそれぞれ、アンビエンス、コンプレッション、ＥＱ及びリミッタ設定を微調整することができるものである。
【図１６－１７】システムを操作するためのアプリケーションインターフェースの図であり、設定を記憶し、それぞれ呼び出し／インポートすることができる。
【図１８】システムを操作するためのカラオケインターフェースの図である。
【図１９】所望の音声のリスナが「クローズド型」イヤホン又はヘッドホンを使用している状況に対して特に適用可能な基本システムの概略図である。
【図２０】その基本的コンセプト内のシステムの比較的単純な一例の概略図である。
【図２１】その基本的コンセプト内のシステムの若干複雑な一例の概略図である。
【図２２】図２１の例に対応するアプリ（アプリケーション）がアップロードされた「ｉＰｈｏｎｅ（登録商標）」の略図である。
【図２３】「クローズド型」イヤホン又はヘッドホンを使用している所望のオーディオのリスナが、できる限り周囲の音を排除したいと希望する状況に対して特に適用可能な単純なシステムの概略図である。
【発明を実施するための形態】
【００３２】
図１は、本発明の実施形態の現代の音楽再生可能な携帯電話への適用の一例である、選択的サウンドキャンセリングシステムの全体を示す。この例は、選択的サウンドキャンセリングシステム及び方法を作り出すために採用した特徴を説明することを意図するものであり、本発明の使用方法、使用状況を制限するものと理解されるべきではない。説明が最も簡単となるよう「アップル」（登録商標）社の携帯電話「ｉＰｈｏｎｅ」を具体的な例として用いた。携帯電話の内蔵マイクロホン及び／又は追加的な外付けマイクロホンを用いることができる。本システムは、２つ以上のマイクロホンにより動作するように構成されてもよい。本システムは、一般的な小型イヤホンを含む全てのヘッドセットにより動作することができるが、「クローズド型」イヤホン又はヘッドホンとともに用いた場合最も効果が高い。なお、「ＦＸ」は、「効果」を意味する。
【００３３】
「ｉＰｈｏｎｅ」の例では、ユーザは、システムを実現し、当該システムを操作するアプリケーションインターフェースを提供するソフトウェア「アプリ（アプリケーション）」を携帯電話「ｉＰｈｏｎｅ」にアップロードすることができる。ユーザが「アプリ」を立ち上げると、図２に例示した画面が表示される。この図は、サウンドキャンセリングパラメータの調節が限定的な１つの簡単なインターフェースを示しており、他の例が図１０及び図１１に示めされている。
【００３４】
ユーザが通常のヘッドホン又はイヤホンを装着し、音声の認識を必要としない第１の操作モードでは、「ｃａｌｉｂｒａｔｅ（キャリブレート）」と表示されたフェーダを、周囲音が完全に遮断される又は大幅に低減されるまで上下に調節できる。ユーザは、周囲の音を聞きたくない場合は、他に特に何もする必要はなく、ただ通常の音楽プレーヤの「ｐｌａｙ（再生）」を押せばよい。このような調節のみで、ユーザは、最小限の背景音を聞きながら又は背景音を聞かずに音楽を楽しむことができ、ユーザがより低く安全なレベルの音量で音楽を聞くことができるという利点がある。第１のモードで、携帯電話が通話に用いられる場合、「アプリ」は、ユーザが通話を終えるまで一時停止する。
【００３５】
又は、ユーザはシステムを使って、音楽を聞かずに周囲の音を低減又は遮断してもよい。
【００３６】
音声の認識が必要とされる第２のモードでは、第１のモードの上述したステップに加えて、ユーザは、ごく身近なものが所望のレベルで聞こえるまで「アンビエンス（ａｍｂｉ

ｅｎｃｅ）」と表示されたフェーダを動かす。ユーザは、アンビエンスフェーダにより、背景音をどの程度聞こえるようにするかを選択し、不要な音を遮断又は著しく低減させつつ会話することができるようになる。ユーザは、アンビエンスフェーダ及びキャリブレートフェーダの両方の効果によって、アクティブなサウンドキャンセリングでごく身近な音声を認識しながら、音楽を聴いたり、電話をかけたり、生の会話をできるようになる。更にユーザは、「アプリ」を用いて、通話中の受信器の音を、受け手側に伝送される前に「ノイズキャンセル」でき、それにより、両者が低レベルの音量でより簡単かつより明瞭な会話をすることが可能となるとともに、例えば、風の強い環境で電話をかける時のように、背景音が大きいエリアにおいても、より簡単かつ明瞭なに会話をすることが可能となる。

【００３７】

特に、ユーザは、このシステムによって、音楽を聞く際、完全に又は実質的に完全に背景音を遮断しながら、警察のサイレン、火事の警報、呼び鈴又は、話しかけられた声等を聞くことができる。

【００３８】

図３Ａに例示したように、ヘッドホン１１によって、マイクロホン２が受信した周囲の音の選択した部分を所望の音声源９からの電気的出力信号に対応する所望の音声とともに発することができるようになる。図３Ｂは、選択的サウンドキャンセリングの若干複雑な概略図を示し、このシステムでは、携帯電話のマイクロホン（Ｍｉｃ）から信号を取り出し、位相をシフトしてキャンセリング用位相変化信号を生成することにより、上述した効果が得られる。典型的には、これは、信号の位相を１８０度変える（更にユーザが、各方向において位相シフトを＋又は－１から５度の微調整ができるようにする）ことによって行われる。完全にキャンセル効果を得る位相変更信号を生成するためには、補正に必要なシフトの量が、例えば、風の方向や速度等多くの外的要因によって影響されるため、位相シフト度にいくらかの余裕が必要であることが当業者には理解される。位相変更信号が処理され、ヘッドホンに供給されると、音同士が「打ち消し合う」。しかしながら、本明細書で言及するように、このシステム及び方法の適用は、操作の第１モードに関連して説明したように、ユーザが周囲の音を完全に打ち消したい時にだけ適切である。操作の第２モードに関しては、上記に加えて、本システムは、Ｍｉｃから同一の第２の信号を取り出し、部分的なキャンセル効果を得る位相変更信号を生成する。典型的にこれは、信号のフェーズを４５度（この位相シフトの量は、任意の部分的なキャンセル効果を得る位相シフトとすることができる）変えることによってなされる。所望の結果を得るため、それぞれの周波数、音量及び、他のオーディオ特性に作用するようこの位相変更信号をＥＱ、コンプレッサ、ゲート及び／又はスペクトルエンハンサを用いて更に処理する。この信号をヘッドホンに供給すると、さらなるノイズキャンセリングは提供されないので、周囲音は聞こえたままである。ユーザは、安全なレベルで音声を認識しながら電話で音楽を聞くことができる。

【００３９】

本システムによってユーザは、「周囲音の完全なサウンドキャンセリング」や、ユーザが、例えば火災警報や警察のサイレンにより注意を促されることを希望する場合又は、ユーザがすぐ近くで話しかけられる場合等の「アラームモード」又は、ユーザが聴いている背景音を制御したり下げたりすることで音楽やその他音声を聴きながら又は音楽再生電話で通話や受話しながら、ユーザのごく身近な音声を認識することが可能な「音声認識モード」等の多くの利便性の高いモードにおいてヘッドホンで聞くことが可能となるという利点がある。

【００４０】

より高度なインターフェースを図１２から図１５に示す。ユーザは、この高度なインターフェースにより主要なサウンドキャンセリングのパラメータを編集することができ、聴きたい周囲の背景音を効果的に選択することが可能となる。例えばユーザは、車内放送システムのメッセージは聴ける状態で、線路の音を「ノイズキャンセル」することを希望す

る場合がある。高度なモードによりその他多くの新規の特性がアクセスされる。図１２は、閾値、範囲、ゲイン等の「アンビエンス」タブでアクセス可能な特性の範囲を示す。図１３は、例えば、閾値、比率、立ち上がり、解除、ゲイン等の「コンプレッション」タブでアクセス可能な特性の範囲を示す。図１４は、例えば、高、高－中、中、低－中、低等の「ＥＱ」タブを介して調節可能なＥＱの特性を示す。例えば、調節可能なＱ係数で帯域フィルタリング等、より高度なＥＱを用いてもよい。図１５は、例えば、最小、最大、感度、ゲイン等の「リミッタ（ｌｉｍｉｔｅｒ）」タブを介して調節可能な特性の範囲を示す。

【００４１】

図１６及び図１７は、高度な設定を記憶したり呼び出したりするインターフェースを示す。好みの設定を保存するための多くの空白の事前設定がユーザに提供される。また、出荷時の設定もこのインターフェースからアクセス可能である。

【００４２】

図４、図５及び図６は、サウンドキャンセリング用モードＡ～Ｃを示す。モードＡは、選択的サウンドキャンセリングに関し、オプションの１つ以上のマイクロホンを示す。モードＢは、最大サウンドキャンセリングに関する。モードＣでは、点線で囲んだ要素は、ソフトウェアインターフェース（図１参照）を介したユーザ制御が可能又は外付け制御ユニットに含むことができる。

【００４３】

別実施形態を、補聴器における選択的サウンドキャンセリングを例示する図７に示す。補聴器のユーザは、聴覚を損傷する悪循環に陥ったことに気が付くことが多い。聴覚が落ちた又は背景音が多い場合にユーザは、本能的に補聴器のボリュームを上げる。これにより、難聴が益々加速されてしまう。本実施形態は、この問題及び他の関連する問題を、不要な背景の周囲音を打ち消すことにより解決するものである。これによりユーザは、より安全なレベルの音量でより容易に聞けるようになり、聴力の低下を防ぐことができる。特に、補聴器を、本システムを含むよう変形可能である。オプションの外付けコントローラを補聴器に追加することで、どの音をキャンセルするか微調整することができるようになり、個人の好みや要求に適合させることができる。特に、点線で囲んだ構成要素を外付けハードウェアモジュールに組み込んで、ユーザが必要性や好みに従ってシステムが動作するように微調整可能としても良い。あるいは、多くの事前設定を、補聴器にプログラムして、例えば、機械、自動車、電車及び／又は飛行機等の不要な周囲音を選択して低減させる等、様々なモードで動作するようにしてもよい。

【００４４】

図８は、非マイクロホンバージョンである更なる別実施形態を示す。本システムは、事前設定として共通の音係数の表示を採用して、例えば標準的ｍｐ３プレーヤ等のマイクロホンを用いない非Ｍｉｃ装置で当該システムと動作することを可能とする。非Ｍｉｃプラットフォーム用の「アプリ」は、キャンセルされる音がエンジン又は客室の音である場合の「飛行機モード」等、多くの事前設定を含むことができる。また、ユーザは、サウンドキャンセリングに対するカウンタ信号として機能する予め録音した音のサンプルを使用及び生成することもできる。例えば、ユーザは、携帯電話を用いて周囲音を録音し、ＭＰ３ファイルとして保存し、それをユーザのＭＰ３プレーヤにインポートしてキャンセル用音源として使用できる。このモードにおいてユーザは、システムを用いてキャンセルしたい（すなわち、聴きたくない）音を合成可能である。それだけではなく、本システムは、外付けマイクロホンを使用して、任意の音楽プレーヤで用いることを可能とする。ｍｐ３ハードウェア及び他の全ての音声配信システムが本システムと動作することを可能とする内蔵型マイクロホンを特徴とする数多くのヘッドホンがある。

【００４５】

スピーカに基づく選択的サウンドキャンセリングに関する更なる別実施形態を図９に示す。特定の用途に見合うよう本システムの変形例を利用できる。例えば、居間では、４マイクロホン構成が必要であり、大広間では３０のマイクロホンが必要となる。音響専門家

による、マイクロホンやスピーカの配置に対するアドバイスや、システムのキャリブレーションが必要な場合がある。これは、室内、間仕切りのないオフィス、映画館、コンサートホール、会議室、更には屋外のイベント等における不要な背景音を排除する又は著しく低減させるために有益である。マイクロホン及びスピーカの実際の配置は、選択的サウンドキャンセリング構成及びマイクロホンやスピーカの数等必要な設備についてアドバイスを行う音響専門家による徹底した音響調査の後に決定される。テレビと、３つのノイズを受けるマイクロホンと６つのノイズキャンセリングスピーカとがある居間の例を図９に示す。図９に例示したシステムによってユーザは、最適な設定となるまでリモコンを微調整することができる。間仕切りのないオフィスや劇場等のより複雑な状況では、熟練した専門家がキャリブレーションを完了させてもよい。特に注目すべきは、マイクロホンは図９における不要な音の音源の前に置かれていることである。更に、スピーカは、不要な音の近くに置かれている。そのため、不要な音のキャンセリング信号であるキャンセル用フェーズ変更信号を発するスピーカによって、サウンドキャンセリングが実現する。図示していないが、呼び鈴が鳴る場合、ユーザが望まない限りその音はキャンセルされないので、ユーザの注意を喚起できる。

【００４６】

図１８は、カラオケインターフェース、特に、マルチトラックレコーダを示し、このモードでは、ユーザはボーカル、並びにギターやドラムなどの楽器の録音が可能である。またユーザは主音量とともにボーカルやリバーブ、ディレイ、コンプレッサ等の特性を調節することができる。このインターフェースを介してユーザは、生成されたサウンドキャンセリング効果を調節し、通常の再生等の機能とともにファイル／設定を保存、インポート、エクスポートすることができる。

【００４７】

以下に本システムの追加可能な特徴を簡単に説明する。

【００４８】

本システムを実行する「アプリ」は、ソフトウェアのプラグインを受け付けるように設計することができる。このシステム上で用いられるプラグインの定義は、新しい特徴を追加するためにシステムとともに用いることができる互換性のある個別のソフトウェアである。「アプリ」を一旦装置にインストールすると、ユーザは機能を追加する又は増やすためのメインの「アプリ」が用いる「アドオン」ソフトウェア（プラグイン）をダウンロード又は取得することができる。例えば、ユーザは、ボイスチェンジング又はリバーブ効果プラグインをダウンロードすることができる。

【００４９】

本システムの一特徴には、ユーザが会議、メッセージ、メモ又はカラオケを、アクティブ選択的サウンドキャンセリングを施して録音することを可能とする録音設備がある。録音されたファイルは、任意の音楽プレーヤ上で再生することができるとともに、ＰＣやＭＰ３プレーヤ等の外付け装置にエクスポートすることができ、ｍｍｓや電子メールを介して送信することができる［とともに、任意の通話者が聞くこともできる］。

【００５０】

本システムは、標準的ｍｐ３プレーヤ、補聴器又はマイクロホンを持たない他の装置上で動作するマイクロホンを特徴とすることができる外付けハードウェアユニットに組み込んでもよい。フェーダ、ノブ又はタッチスクリーンの形をした（有線又は無線の）外付けコントローラ等の他のハードウェアを用いて、サウンドキャンセリングパラメータ［及び、ＥＱ、ディレイ、リバーブ等の他の特徴］を制御してもよい。

【００５１】

本システムは、任意の携帯電話（又はマイクロホンを装備した装置）を従来の補聴器に対して選択的サウンドキャンセリング機能から生じる多くの利点を有する、強力な、特性を完全装備した補聴器として機能するように構成することができる。

【００５２】

本システムは、任意の携帯電話（又はマイクロホンを装備した装置）を、（装置がイン

ターネットを使用可能な場合）インターネットを通じて、又は、通常の電話発信として通話可能な強力な、特性を完全装備した音楽プレーヤとして構成することができる。

【００５３】

図１９を参照すると、この基本的概念において、周囲音は、当該周囲音を表す電気出力信号はマイクロホンで受信され、デジタルオーディオプレーヤ（ＭＰ３）からの音楽等所望の音声源から他の電気出力信号を受け取る処理ステージに送られる。２つの信号は、合成されて、（他の処理経て又は他の処理なしで）結果として得られた電気信号は「クローズド型」イヤホン又はヘッドホンに供給され、イヤホン又はヘッドホンの装着者に聴こえる音声に変換される。

【００５４】

図２０に示した例では、マイクロホンからの電気出力信号は以下の通りゲートされる。即ち、所定の音量を超える電気出力信号によって表される周囲音があると、ゲートが、所望の音声の音源（例えばＭＰ３プレーヤ）からの電気出力信号も受信するミキサに電気信号を出力し始める。装着者は、例えばスライダを用いてゲート閾値を調節可能である。また、装着者は、例えば他のスライダを用いてマイクロホンの音量を調節することもできる。ミキサからの電子出力信号を、音声出力音量がユーザにより、例えばスライダを用いて制御可能なイヤホン又はヘッドホンに供給される。

【００５５】

別の例において、非常に大きな音に対して装着者を保護するために、ゲートは１つの閾値に加えて上限を有してもよい。上限は、下限と別に設定される、又は製造業者により設定される。

【００５６】

図２１に示した例では、マイクロホンからの電気出力信号、すなわち、周囲音の視覚的表現が表示される。電気信号は再びゲートされ、装着者は、閾値を設定できる。装着者がアラートモード制御を「ＯＮ」に切り換えていた場合、ゲート閾値を超えるとゲート出力により振動や、視覚又は音声により警告が出される。装着者は、ゲートされた電気出力信号によって表されたゲートされた周囲音の音量を調節することができ、調節された信号は、所望の音声源（例えば、ＭＰ３プレーヤ）からの電気出力信号を受けるミキサに入力される。ミキサからの電気出力信号は再びイヤホン又はヘッドホンに供給され、主音量制御が再び行われる。

【００５７】

図２２は、「アウェアネス（Ａｗａｒｅｎｅｓｓ）」（登録商標）と呼ばれるアプリがアップロードされた携帯電話「ｉＰｈｏｎｅ」のタッチスクリーンを示す。携帯電話の周りを左上隅から時計回りに様々な特徴を説明する。トラックスクロールの左端をタップすると、前のトラックに切り換え、右端をタップすることで、次のトラックに切り換える。あるいは、これらの端部の間で左から右又は右から左へとスワイプすることで、前のトラック又は次のトラックに切り換える。「ｅｓｓｅｎｃｙ」（登録商標）をタップすることで、ヘルプ及びその他の情報が提供される。「Ａｗａｒｅｎｅｓｓ」をタップすることで、所望の音源がオーバーライドされ、周囲音からの出力のみが残る。ＬＥＤ（発光ダイオード）の表示が緑の場合、ゲート閾値を超えていないことを意味する。赤に変わると、閾値を超えたことを意味する。上述したアプリのように、上限及び下限のゲート閾値がある場合、「Ｔ－高」及び「Ｔ－低」の線で示した通り、スクリーン上で垂直方向のピンチ動作により上限下限を設定する。周囲音の存在と大きさを示す垂直の線を有する３次元背景がスクリーンに表示されている。携帯電話を振って、ランダムにトラックをシャッフルさせることができる。スクリーンは、ロックすることができる。ゲート閾値や音量に関する様々な設定は、スクリーン上に表示可能なスライダにより設定することができる。振動によるアラートは、オンとオフに切り替えることができる。マイクロホンは、タップによりオン、オフを切り替えることができ、指を押しつけてマイクロホンの出力レベルや感度を設定することができる。ｉＴｕｎｅｓライブラリにアクセスすることができる。スクリーンを水平にスワイプすることで、プレーヤの音量を上げ下げすることができる。アプリの

スクリーン１が示される。４秒以上起動されなければ、各種制御表示が消え、スクリーン２の３次元背景のみが残る。図示のプレーヤ制御表示によって、左又は右のボタンを押すことでトラックがスクロールされる。中央のボタンを押すと「再生」され、再びそのボタンを押すと「一時停止」し、長押しするとトラックの頭に戻る。これら全てのボタンをスワイプすることで、ボタンのスクリーン上の位置及び大きさを選択することができるようになるので、ユーザは、ボタンの大きさを変えたり、それぞれをスクリーン上の好きな場所にドラッグしたりできる。

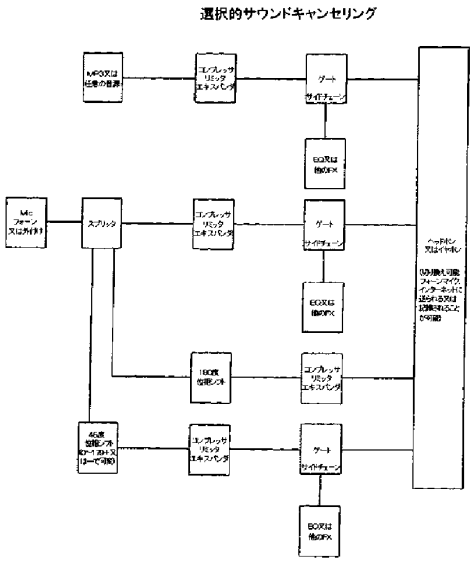
【００５８】

図２３を参照すると、マイクロホン２からの電気出力信号３の大きさ及び周波数の両方に関して、インバージョン５があることで、８において周囲音が混ざっていないことを示すが、もちろん、他の方法で聞き手の耳に届く周囲音の僅かな部分を防ぐものではない。

【００５９】

本発明の様々な実施形態を、聞き手に聴こえる音の音量に関する処理として全般的に説明してきたが、本発明は、そのような音の周波数の処理又は周波数及び音量の処理にも適用可能である。

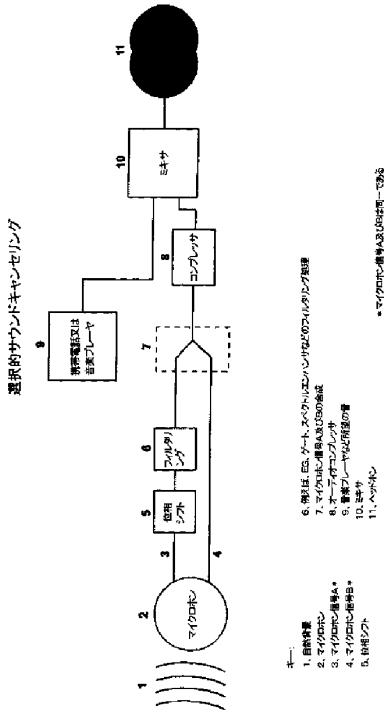
【図１】



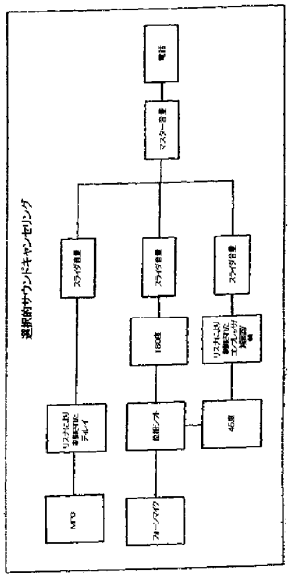
【図２】



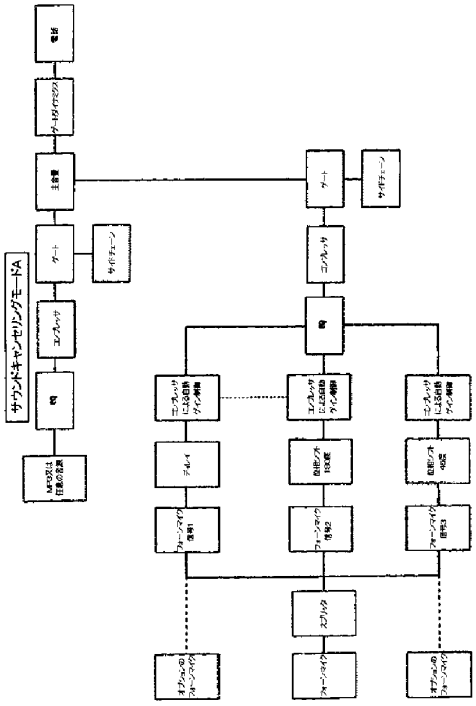
【図３Ａ】



【図３Ｂ】



【図４】



【図５】

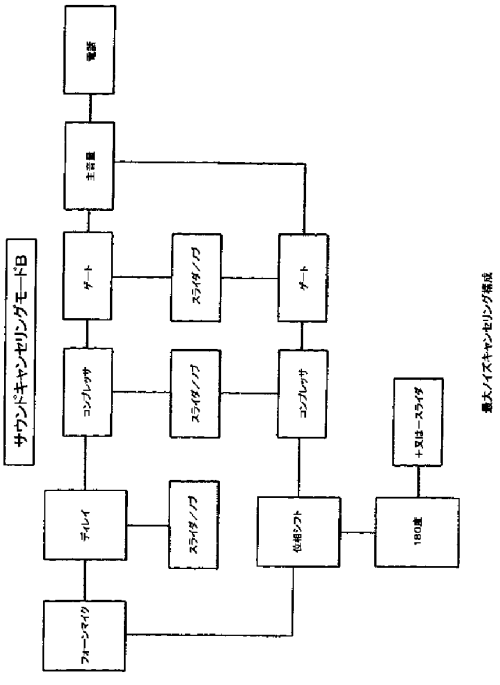
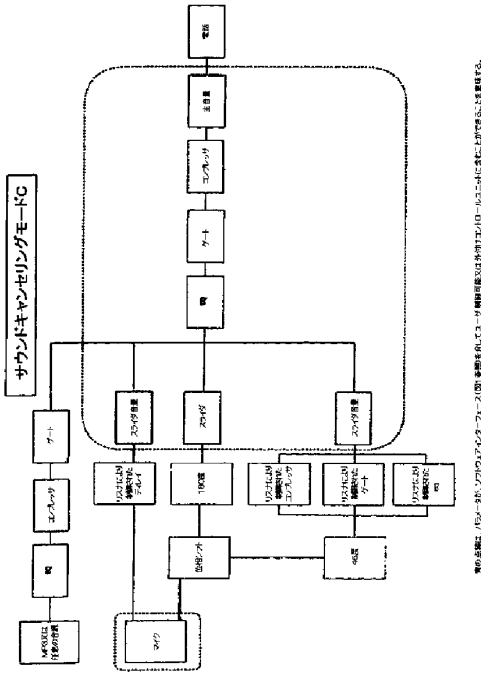
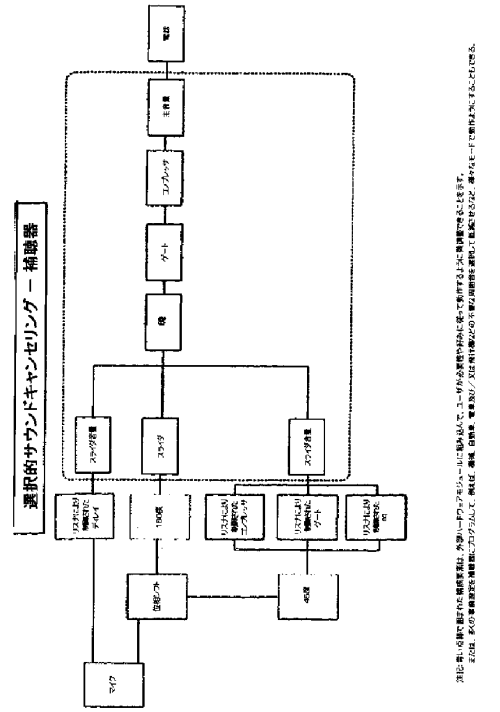


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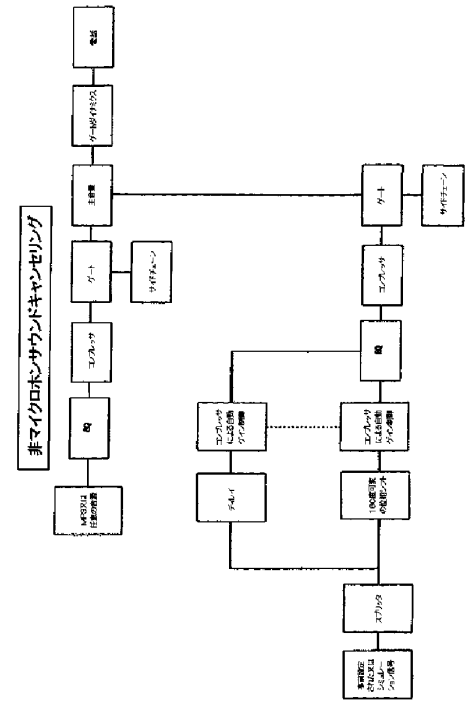
【図６】


サウンドキャンセリングモードC

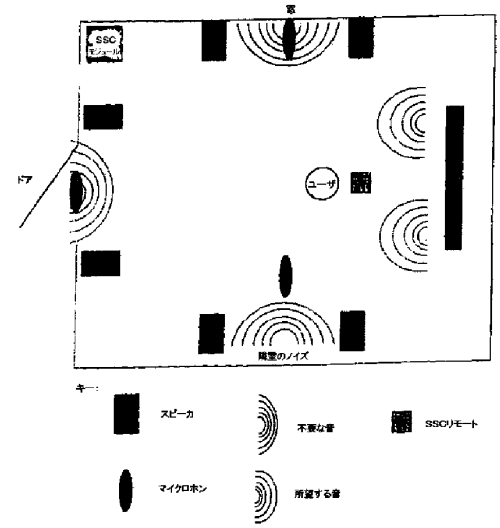
【図７】


選択的サウンドキャンセリング－補聴器

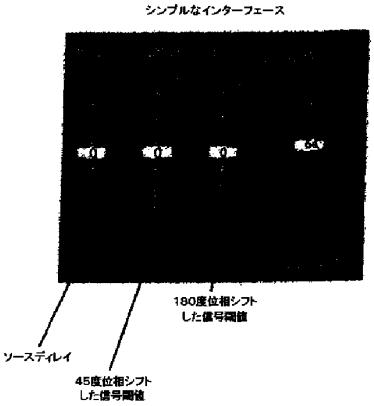
【図８】


非マイクロホンサウンドキャンセリング

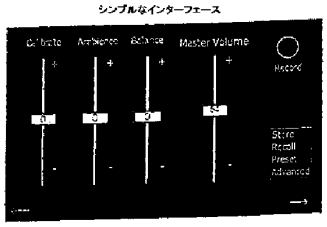
【図９】


SSC
周囲のノイズ
スピーカ
不要な音
SSCリモート
マイクロホン
所望する音

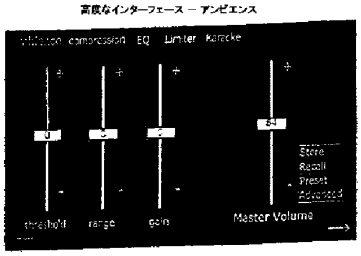
【図１０】



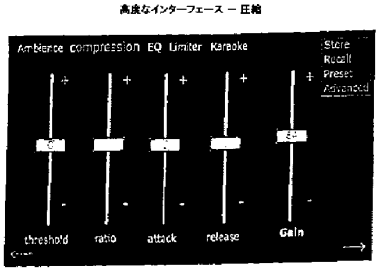
【図１１】



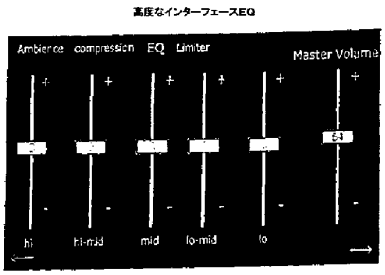
【図１２】



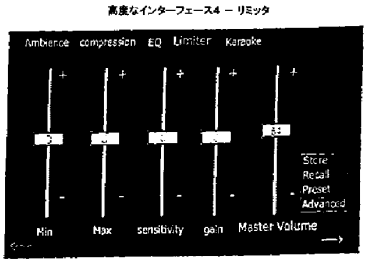
【図１３】



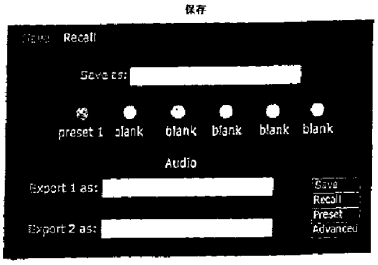
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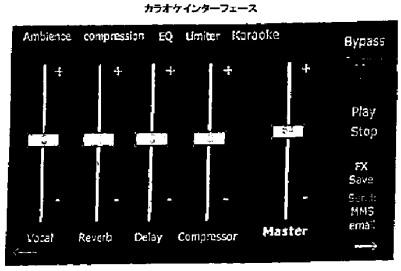
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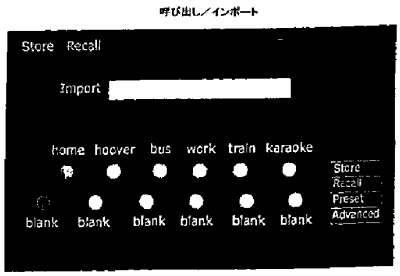
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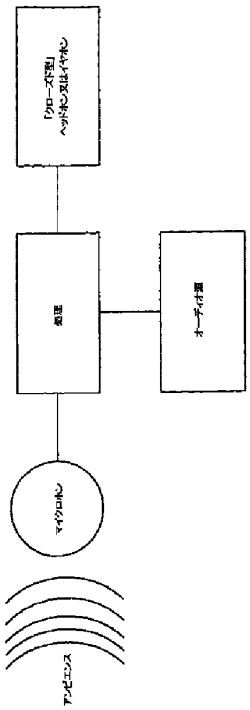
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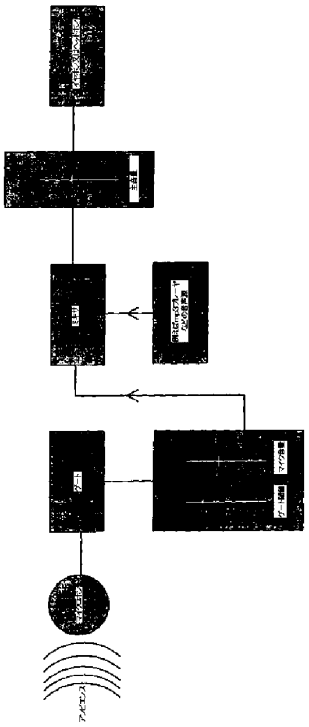
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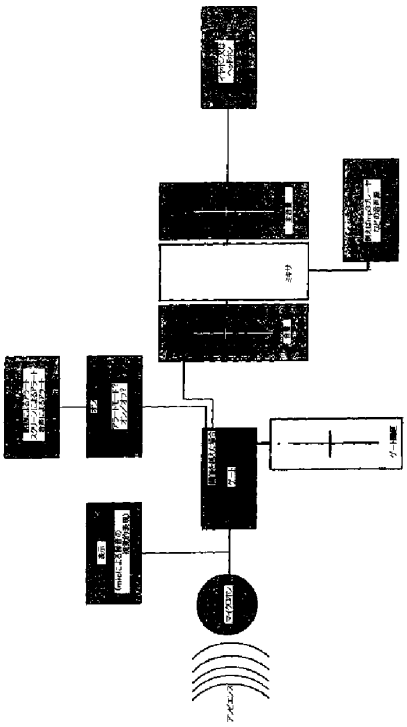
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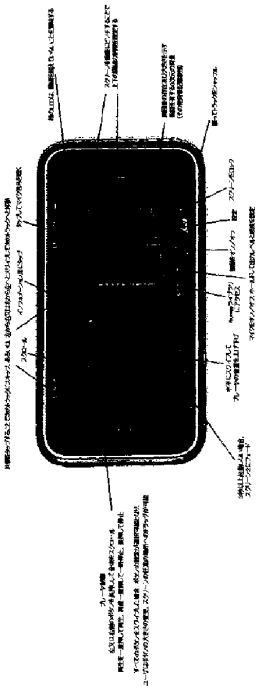
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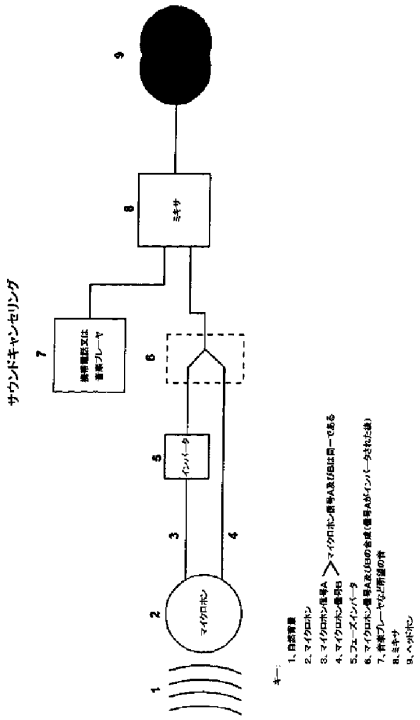
【図２１】



【図２２】



【図２３】



【国際調査報告】

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2010/001549

A. CLASSIFICATION OF SUBJECT MATTER
INV. G10K11/178 H04R1/10
ADD.

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
G10K H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)
EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 2003/003944 A1 (ROSENZWEIG MICHAEL D [US]) 2 January 2003 (2003-01-02) paragraphs; figures 1-3	1-66
X	US 2002/141599 A1 (TRAJKOVIC MIROSLAV [US] ET AL) 3 October 2002 (2002-10-03) paragraphs [0005], [0007] - [0026]; figures 1-5	1-56, 58-66
X	GB 2 436 657 A (SONAPTIC LTD [GB]) 3 October 2007 (2007-10-03) page 15, line 8 - page 19, line 4; figures 4-8	1-56, 58-66

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[X] Further documents are listed in the continuation of Box C. [X] See patent family annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
11 February 2011	21/02/2011

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Fax: (+31-70) 340-3016	Trique, Michael

2

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INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2010/001549

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 2006/076369 A1 (TARGUS GROUP INTERNATIONAL INC [US]; BEASLEY NIGEL [GB]) 20 July 2006 (2006-07-20) paragraphs [0026] - [0028]; figure 1 paragraph [0051]; figure 9 -----	1-56, 58-66
X	US 2007/223717 A1 (BOERSMA JOHAN [NL]) 27 September 2007 (2007-09-27) paragraphs [0044] - [0064]; figures 1-4 -----	1-14

2

Form PCT/ISA/210 (continuation of second sheet) (April 2005)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/GB2010/001549

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
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US 2002141599 A1	03-10-2002	EP 1377961 A2	07-01-2004
		WO 02082422 A2	17-10-2002
		JP 2004526375 T	26-08-2004
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WO 2006076369 A1	20-07-2006	NONE	
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		EP 1994728 A1	26-11-2008
		WO 2007102047 A1	13-09-2007
		JP 2009529275 T	13-08-2009
		KR 20080111004 A	22-12-2008

Form PCT/ISA/210 (patent family annex) (April 2005)

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Ｆターム(参考) 5D005 BB11
5D061 FF02
5D220 AA02 BA04

Espacenet

CN203761556U Double-microphone noise reduction earphone

Applicants: FUNG SHING COMPANY LTD

Inventors: HUI KWAI MING CLIVE

Classifications:
IPC ***H04R1/10; H04R3/00;***

Priorities: CN201320754777U 2013-11-25

Application: CN201320754777U·2013-11-25

Publication: CN203761556U·2014-08-06

Published as: CN203761556U

Double-microphone noise reduction earphone

Abstract

The utility model relates to the earphone and particularly to the microphone noise reduction earphone. In order to solve the problem in the prior art that the noise reduction effect of the earphone is not ideal. The utility model discloses a double-microphone noise reduction earphone comprising a head band, a voice generation unit, an earshield and a lead. The double-microphone noise reduction earphone comprises a control circuit, an external noise reduction microphone and an internal noise reduction microphone. The external microphone is externally positioned on the earshield and the internal noise reduction microphone is positioned inside of the earshield. The input terminal of the external noise reduction microphone is connected to the control circuit; the output terminal of the internal noise reduction microphone is connected to the control circuit; and the output terminal of the control circuit is connected to the voice generation unit. The double-microphone noise reduction earphone achieves a better noise reduction effect by reducing the noise both inside the earshield and outside the earshield.

CN203761556U Double-microphone noise reduction earphone

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

Dual microphone noise canceling headphones

Technical field

The utility model relates to an earphone, in particular to a microphone noise canceling earphone.

Background technique

The commonly used earphones are mainly composed of four parts: a headband, a sounding unit, an earmuff and a lead; the headband is used to fix the earmuff to the ear, the lead is used to convey the audio signal from the sound to the sounding unit, and the sounding unit is arranged at Inside the earmuffs. The existing earphones usually have a noise reduction function, and a microphone is arranged outside each of the left and right earmuffs to collect sound samples. When the external noise is more diverse and variable, each earmuff has only one sampling microphone that collects all noises, which is bound to be The direction of the noise sampling is narrow and not precise and precise, so that the high-quality noise reduction effect cannot be achieved.

Summary of the invention

In order to solve the problem that the noise reduction effect of the existing earphone is not ideal, the utility model provides a double microphone noise canceling earphone, which can achieve better noise reduction effect by reducing the noise of the inner environment of the earmuff and the external environment.

In order to achieve the above object, the utility model adopts the following technical solutions:

A dual microphone noise canceling earphone comprising a headband, a sounding unit, an earmuff and a lead, wherein the earphone further comprises a control circuit, an external noise reduction microphone and an internal noise reduction microphone; and the external part is disposed outside the earmuff a noise reduction microphone, the internal noise reduction microphone is disposed inside the ear cover; an output end of the external noise reduction microphone is connected to the control circuit; an output end of the internal noise reduction microphone is connected to the control circuit; The output of the control circuit is connected to the sounding unit; the external noise canceling microphone is used to collect the noise audio signal outside the ear cup. An internal noise reduction microphone is used to capture the noise audio signal inside the ear cup.

The noise audio signal collected by the external noise reduction microphone is called audio 1, and the noise audio signal collected by the internal noise reduction microphone is called audio 2. The audio 1 and the audio 2 are respectively input to a control circuit which receives and calculates waveform data of the noise sound waves of the sound waves of the audio 1 and the audio 2, and outputs the data to the sounding unit. The inventive unit, based on the data provided by the control circuit, produces sound waves of the same amplitude and opposite phase as the noise sound waves to automatically cancel the processing noise. In this way, the sound emitted by the sounding unit can avoid the interference of environmental noise to the greatest extent, so that people can listen in a relatively quiet environment, and the sound heard is clear and the sound quality is beautiful.

Further, the external noise reduction microphone and the internal noise reduction microphone respectively collect noise audio signals in opposite directions.

Further, the sounding unit is a fidelity treble.

Further, the control circuit is connected to an external power source. The power source may be a battery or a plug that can be connected to the socket.

Further, the external noise reduction microphone and the internal noise reduction microphone are used to collect ambient low frequency noise at a frequency of 100-500 Hz.

Further, the external noise reduction microphone is mounted on the earmuff, and the device for collecting noise faces the earmuff; the internal noise reduction microphone is installed inside the earmuff, and the device for collecting noise faces the external noise reduction microphone. The device for collecting noise faces in the opposite direction; and the external noise reduction microphone and the internal noise reduction microphone are both mounted below the sounding unit; the sounding unit is horizontally opposed to the ear.

Further, the angle between the direction in which the noise collecting device of the external noise reduction microphone faces and the horizontal direction of the ear is 30-60.

Further, the angle between the direction in which the noise collecting device of the external noise reduction microphone faces and the horizontal direction of the ear is 45°.

Compared with the prior art, the invention adopts double microphone sampling, adopts a dual method to reduce noise, effectively improves the sensitivity and accuracy of noise sampling, thereby effectively reducing environmental noise inside and outside the earmuff; inside the earmuff The sound wave is opposite to the external microphone, so as to achieve a better sound sample collection effect, and the sound wave with the same amplitude and opposite amplitude is prepared by the control circuit and the sounding unit, and the automatic cancellation is performed, thereby achieving better. Ground noise reduction effect. To enable users to enjoy better music entertainment and to solve various small noise problems in daily life (such as airport and industrial noise), just like suddenly moving from a noisy environment In a very quiet environment.

The noise reduction principle of the dual microphone noise canceling earphone provided by the utility model is as follows: the two low-frequency noises (100-500HZ) heard by the ear are detected by two microphones respectively installed inside and outside the earmuff, and the noise signal is transmitted to the noise signal. The control circuit, the control circuit, through the instant operation, transmits the sound wave with the opposite phase of the noise and the same amplitude to cancel the noise through the high-fidelity speaker, so that the low-frequency noise in the incoming human ear is greatly reduced, and the auditory environment is quieter.

Traditional headphones, no external, built-in microphone system, only a single internal or external microphone, up to 20db noise reduction. The dual microphone noise canceling earphone provided by the utility model has the following beneficial effects:

1、In the case that the environment 250Hz noise reaches 60 decibels (db), people will feel uncomfortable. When the earphone using this technology starts the noise reduction function, it can effectively reduce the noise of the external 25db-30db and create a quieter atmosphere.

2、It has strong practicability and can effectively and effectively reduce and offset the regular and irregular noise of 100Hz-500Hz.

DRAWINGS

1 is a schematic structural view of a microphone noise canceling earphone provided by the present invention;

2 is a schematic diagram of sound waves in a noise processing process;

Figure 3 is a schematic diagram of sound waves after noise processing.

Detailed ways

The technical solution provided by the present invention will be further described below by way of embodiments with reference to the accompanying drawings.

Example 1

As shown in FIG. 1 , the dual microphone noise canceling earphone provided by the present invention comprises a headband, a sounding unit 5, an earmuff 10 and a lead wire, and the earphone further comprises a control circuit 3, an external noise reduction microphone 2 and an internal noise reduction microphone. 1; the

external noise reduction microphone 2 is disposed outside the earmuff 10, and the internal noise reduction microphone 1 is disposed inside the earmuff 10; the output end of the external noise reduction microphone 1 is connected to the control circuit 3; The output of the internal noise reduction microphone 2 is connected to the control circuit 3; the output of the control circuit 3 is connected to the sounding unit 5.

The external noise reduction microphone 2 and the internal noise reduction microphone 1 respectively collect noise audio signals in opposite directions.

The sounding unit 5 is a fidelity treble horn.

The control circuit 3 is connected to an external power source 4.

As shown in FIG. 2 and FIG. 3, A in FIG. 2 represents sound waves of low frequency noise in the environment, and B represents sound waves of reverse noise calculated by the control circuit. The dual microphone noise canceling earphone provided in this embodiment passes through the control circuit and sounds. The unit produces sound waves of opposite amplitude and opposite amplitude to the real noise (referred to as reverse noise), which offsets the ambient noise and achieves better noise reduction.

Example 2

As shown in FIG. 1 , the dual microphone noise canceling earphone provided by the utility model comprises a headband, a fidelity treble horn 5, an earmuff 10 and a lead wire, and the earphone further comprises a control circuit 3, an external noise reduction microphone 2 and an internal part. a noise reduction microphone 1; the external noise reduction microphone 2 is disposed outside the earmuff 10, and the internal noise reduction microphone 1 is disposed inside the earmuff 10; the output end of the external noise reduction microphone 1 is connected to the control circuit 3 The output end of the internal noise reduction microphone 2 is connected to the control circuit 3; the output end of the control circuit 3 is connected to the fidelity treble horn 5.

The external noise reduction microphone 2 and the internal noise reduction microphone 1 respectively collect noise audio signals in opposite directions.

The control circuit 3 is connected to an external power source 4.

Further, 1) the core material of the product: a horn composed of a composite film high-preserving magnetic copper coil, a low-noise low-voltage high-power power-amplified imported chip. Among them, the external noise reduction microphone and the internal noise reduction microphone (the division of sound to collect the sound waves in the forward and reverse directions), the imported chip, with anti-interference components, consists of high-frequency shielding structure.

2） The earmuffs are made of protein holster and inert sound-absorbing cotton as filling and auxiliary materials, which can more effectively assist the noise reduction effect and improve the comfort of use.

3） The earphone product shell is made of imported APS material, PVC material and environmentally friendly electronic materials, and the product is more durable.

The above is only the preferred embodiment of the present invention and is not intended to limit the scope of the present invention. All changes and modifications made in accordance with the teachings of the present invention are encompassed within the scope of the invention.

CN203761556U Double-microphone noise reduction earphone

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

1.A dual microphone noise canceling earphone comprising a headband, a sounding unit, an earmuff and a lead, wherein the earphone further comprises a control circuit, an external noise reduction microphone and an internal noise reduction microphone; and the external part is disposed outside the earmuff a noise reduction microphone, the internal noise reduction microphone is disposed inside the ear cover; an output end of the external noise reduction microphone is connected to the control circuit; an output end of the internal noise reduction microphone is connected to the control circuit; The output of the control circuit is connected to the sounding unit; the external noise reduction microphone is used to collect a noise audio signal outside the earmuff, and the internal noise reduction microphone is used to collect the noise audio signal inside the earmuff.

2.The dual microphone noise canceling earphone according to claim 1, wherein said external noise reduction microphone and said internal noise reduction microphone respectively collect noise audio signals in opposite directions.

3.The dual microphone noise canceling earphone according to claim 1, wherein said sounding unit is a fidelity treble horn.

4.The dual microphone noise canceling earphone according to claim 1, wherein said control circuit is connected to an external power source.

5.The dual microphone noise canceling earphone according to claim 1, wherein said external noise reduction microphone and internal noise reduction microphone are used to collect ambient low frequency noise at a frequency of 100-500 Hz.

6.The dual microphone noise canceling earphone according to claim 1, wherein said external noise reduction microphone is mounted on the earmuff, and the means for collecting noise faces the outside of the earcup; said internal noise reduction microphone is mounted inside the earmuff; The direction in which the device collecting noise faces is opposite to the direction in which the noise collecting microphone of the external noise reduction microphone faces; and the external noise reduction microphone and the internal noise reduction microphone are both mounted below the sounding unit; The unit is at the level of the ear.

7.The dual microphone noise canceling earphone according to claim 6, wherein the angle of the direction in which the noise collecting device of the external noise reduction microphone faces is 30 to 60 degrees from the horizontal direction in which the ear is located.

8.The dual microphone noise canceling earphone according to claim 6, wherein the direction in which the noise collecting device of the external noise reduction microphone faces is at an angle of 45 with the horizontal direction in which the ear is located.

(19)中华人民共和国国家知识产权局

(12)实用新型专利

(10)授权公告号 CN 203761556 U
(45)授权公告日 2014.08.06

(21)申请号 201320754777.8

(22)申请日 2013.11.25

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代理人 陈浩

(51)Int.Cl.
H04R 1/10(2006.01)
H04R 3/00(2006.01)

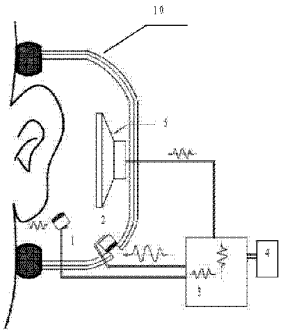
权利要求书1页 说明书3页 附图1页

(54)实用新型名称

双重麦克风降噪耳机

(57)摘要

本实用新型涉及耳机，尤其涉及麦克风降噪耳机。为了解决现有耳机降噪效果不理想的问题，本实用新型提供一种双重麦克风降噪耳机。该双重麦克风降噪耳机，包括头带、发声单元、耳罩和引线，所述耳机还包括控制电路，外部降噪麦克风和内部降噪麦克风；在耳罩外部设置所述外部降噪麦克风，在耳罩内部设置所述内部降噪麦克风；所述外部降噪麦克风的输出端接入所述控制电路；所述内部降噪麦克风的输出端接入所述控制电路；所述控制电路的输出端接入发声单元。该双重麦克风降噪耳机通过降低耳罩内部环境和外部环境的噪声，实现更好地降噪效果。



CN 203761556 U

1. 一种双重麦克风降噪耳机，包括头带、发声单元、耳罩和引线，其特征在于，所述耳机还包括控制电路，外部降噪麦克风和内部降噪麦克风；在耳罩外部设置所述外部降噪麦克风，在耳罩内部设置所述内部降噪麦克风；所述外部降噪麦克风的输出端接入所述控制电路；所述内部降噪麦克风的输出端接入所述控制电路；所述控制电路的输出端接入发声单元；所述外部降噪麦克风用于采集耳罩外部的噪音音频信号，所述内部降噪麦克风用于采集耳罩内部的噪音音频信号。

2. 根据权利要求 1 所述的双重麦克风降噪耳机，其特征在于，所述外部降噪麦克风与所述内部降噪麦克风分别采集相反方向的噪音音频信号。

3. 根据权利要求 1 所述的双重麦克风降噪耳机，其特征在于，所述发声单元为保真高音喇叭。

4. 根据权利要求 1 所述的双重麦克风降噪耳机，其特征在于，所述控制电路连接外部电源。

5. 根据权利要求 1 所述的双重麦克风降噪耳机，其特征在于，所述外部降噪麦克风和内部降噪麦克风用于收集频率为 100-500HZ 的环境低频噪声。

6. 根据权利要求 1 所述的双重麦克风降噪耳机，其特征在于，所述外部降噪麦克风安装在耳罩上，收集噪声的装置面向耳罩外；所述内部降噪麦克风安装在耳罩内部，收集噪声的装置面向的方向与所述外部降噪麦克风的收集噪声的装置所面向的方向相反；并且，所述外部降噪麦克风和内部降噪麦克风均安装在发声单元的下方；所述发声音单元与耳朵水平相对。

7. 根据权利要求 6 所述的双重麦克风降噪耳机，其特征在于，所述外部降噪麦克风的收集噪声的装置面向的方向与耳朵所在的水平方向的夹角是 30-60° 。

8. 根据权利要求 6 所述的双重麦克风降噪耳机，其特征在于，所述外部降噪麦克风的收集噪声的装置面向的方向与耳朵所在的水平方向的夹角是 45° 。

双重麦克风降噪耳机

技术领域

[0001] 本实用新型涉及耳机，尤其涉及麦克风降噪耳机。

背景技术

[0002] 常用的耳机主要由四个部分组成：头带、发声单元、耳罩和引线；头带用于将耳罩固定在耳朵上，引线用于将音频信号从音响输送到发声单元，发声单元设置在耳罩内。现有的耳机通常具有降噪功能，在左右两个耳罩外部各设置一个麦克风采集声样，当外部噪音较为多样且多变的时候，每个耳罩只有一个收集全部噪音的采样麦克风，势必造成噪音采样的方向狭窄和不精细精准，从而无法达到优质的降噪效果。

发明内容

[0003] 为了解决现有耳机降噪效果不理想的问题，本实用新型提供一种双重麦克风降噪耳机，通过降低耳罩内部环境和外部环境的噪声，实现更好地降噪效果。

[0004] 为了达到上述目的，本实用新型采用下述技术方案：

[0005] 一种双重麦克风降噪耳机，包括头带、发声单元、耳罩和引线，其特征在于，所述耳机还包括控制电路，外部降噪麦克风和内部降噪麦克风；在耳罩外部设置所述外部降噪麦克风，在耳罩内部设置所述内部降噪麦克风；所述外部降噪麦克风的输出端接入所述控制电路；所述内部降噪麦克风的输出端接入所述控制电路；所述控制电路的输出端接入发声单元；外部降噪麦克风用于采集耳罩外部的噪音音频信号。内部降噪麦克风用于采集耳罩内部的噪音音频信号。

[0006] 外部降噪麦克风采集的噪音音频信号称为音频1，内部降噪麦克风采集的噪音音频信号称为音频2。音频1和音频2分别输入控制电路，控制电路接收并计算音频1和音频2的声波混合后的噪声声波的波形数据，并将该数据输送到发声单元。发明单元根据控制电路提供的数据，制造出与噪声声波的振幅相同、相位相反的声波，来自动抵消处理噪声。这样，发声单元发出的声音可以最大程度的避免环境噪声的干扰，使得人们能在相对安静的环境中倾听，且听到的声音清晰，音质优美。

[0007] 进一步的，所述外部降噪麦克风与所述内部降噪麦克风分别采集相反方向的噪音音频信号。

[0008] 进一步的，所述发声单元为保真高音喇叭。

[0009] 进一步的，所述控制电路连接外部电源。所述电源可以是电池，也可以是能够与插座相连接的插头。

[0010] 进一步的，所述外部降噪麦克风和内部降噪麦克风用于收集频率为100-500HZ的环境低频噪声。

[0011] 进一步的，所述外部降噪麦克风安装在耳罩上，收集噪声的装置面向耳罩外；所述内部降噪麦克风安装在耳罩内部，收集噪声的装置面向的方向与所述外部降噪麦克风的收集噪声的装置所面向的方向相反；并且，所述外部降噪麦克风和内部降噪麦克风均安装在

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发声单元的下方；所述发声音单元与耳朵水平相对。

[0012] 进一步的，所述外部降噪麦克风的收集噪声的装置面向的方向与耳朵所在的水平方向的夹角是 30-60° 。

[0013] 进一步的，所述外部降噪麦克风的收集噪声的装置面向的方向与耳朵所在的水平方向的夹角是 45° 。

[0014] 与现有技术相比，本发明采用双重麦克风取样，采用双重的方式去降噪，有效地提高了噪声采样的灵敏度及精确性，从而能够有效降低耳罩内部和外部的环境噪声；耳罩内部和外部的麦克风分工采集方向相反的声波，从而达到一个更优质的声样采集效果，并通过控制电路和发声单元制备出同真实噪声相反的、幅度相等的声波，进行自动抵消，从而达到更好地降噪效果。让使用者能够得到更好的音乐娱乐享受，以及为使用者解决日常生活（如机场、工业上的噪声）中的各种有关噪音的小困扰，就像是突然从一个嘈杂的环境中，转移到了一个非常安静的环境。

[0015] 本实用新型提供的双重麦克风降噪耳机的降噪原理简述如下：由两个分别安装于耳罩内外的麦克风先侦测耳朵听到的环境低频噪声（100-500HZ），将噪声信号传至控制电路，控制电路经过即时的运算，通过高保真喇叭来发射与噪声相位相反，振幅相同的声波来抵消噪声，使传入人耳中的低频噪声大幅降低，使听觉环境更加安静。

[0016] 传统的耳机，没有外、内置麦克风系统，只有单个内或外的麦克风，最多达到 20db 的降噪声功能。而本实用新型提供的双重麦克风降噪耳机具有下述有益效果：

[0017] 1、在环境 250Hz 噪声达到 60 分贝（db）的情况下，人们会感到难受，当使用该技术的耳机启动降噪功能时，可以有效降低外来 25db-30db 的噪声，塑造出一个较为安静的氛围。

[0018] 2、实用性强，可以卓越有效地降低、抵消 100Hz-500Hz 有规律及无规律的噪声。

附图说明

[0019] 图 1 为本实用新型提供的麦克风降噪耳机的结构示意图；

[0020] 图 2 为噪音处理过程的声波示意图；

[0021] 图 3 为噪音处理后的声波示意图。

具体实施方式

[0022] 下面通过实施例结合说明书附图进一步说明本实用新型提供的技术方案。

[0023] 实施例 1

[0024] 如图 1 所示，本实用新型提供的双重麦克风降噪耳机，包括头带、发声单元 5、耳罩 10 和引线，所述耳机还包括控制电路 3，外部降噪麦克风 2 和内部降噪麦克风 1；在耳罩 10 外部设置所述外部降噪麦克风 2，在耳罩 10 内部设置所述内部降噪麦克风 1；所述外部降噪麦克风 1 的输出端接入所述控制电路 3；所述内部降噪麦克风 2 的输出端接入所述控制电路 3；所述控制电路 3 的输出端接入发声单元 5。

[0025] 所述外部降噪麦克风 2 与所述内部降噪麦克风 1 分别采集相反方向的噪音音频信号。

[0026] 所述发声单元 5 为保真高音喇叭。

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[0027] 所述控制电路 3 连接外部电源 4。

[0028] 如图 2 和图 3 所示，图 2 中 A 表示环境中低频噪声的声波，B 表示控制电路计算出的反向噪声的声波，本实施例提供的双重麦克风降噪耳机，通过控制电路和发声单元制备出与真实噪声相位相反，振幅相同的声波（简称反向噪声），抵消了环境噪音，从而达到更好地降噪效果。

[0029] 实施例 2

[0030] 如图 1 所示，本实用新型提供的双重麦克风降噪耳机，包括头带、保真高音喇叭 5、耳罩 10 和引线，所述耳机还包括控制电路 3，外部降噪麦克风 2 和内部降噪麦克风 1；在耳罩 10 外部设置所述外部降噪麦克风 2，在耳罩 10 内部设置所述内部降噪麦克风 1；所述外部降噪麦克风 1 的输出端接入所述控制电路 3；所述内部降噪麦克风 2 的输出端接入所述控制电路 3；所述控制电路 3 的输出端接入保真高音喇叭 5。

[0031] 所述外部降噪麦克风 2 与所述内部降噪麦克风 1 分别采集相反方向的噪音音频信号。

[0032] 所述控制电路 3 连接外部电源 4。

[0033] 进一步的，1）产品的核心材料：选用复合膜高保磁纯铜线圈组成的喇叭、低噪声低电压高功率的功率放大的进口芯片。其中，外部降噪麦克风与内部降噪麦克风（分工采集正反方向的声波），选用进口芯片，带抗干扰元件，由高频屏蔽结构组成。

[0034] 2）耳罩产品选用蛋白皮套，惰性吸音棉作为填充及辅助材料，更加有效地辅助降噪效果，提高使用的舒适度。

[0035] 3）耳机产品外壳采用进口 APS 材料、PVC 材料、环保电子材料组成，产品更坚固耐用。

[0036] 以上所述，仅为本发明的较佳实施例而已，并非用于限定本发明的保护范围。凡是根据本发明内容所做的均等变化与修饰，均涵盖在本发明的专利范围内。

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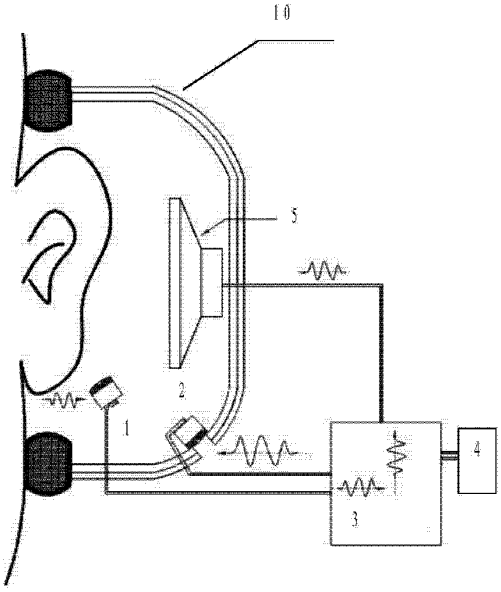


图 1

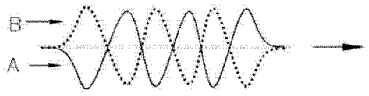


图 2

图 3

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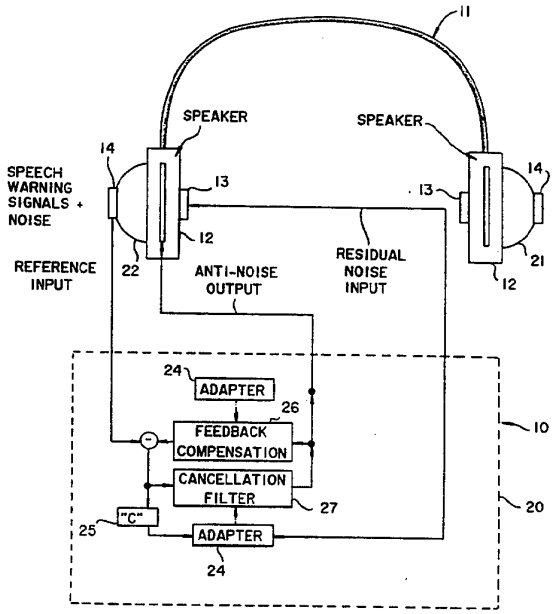
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(54) Title: ACTIVE/PASSIVE HEADSET WITH SPEECH FILTER

(57) Abstract

An active/passive headset with a speech filter characterized by a controller (20) processing speech, noise and warning signals input from microphones (14) on closed back muffs (20) and providing an anti-noise output by a digital virtual earth or adaptive feedforward algorithm to speakers (12) adjacent a user's ear.



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ACTIVE/PASSIVE HEADSET WITH SPEECH FILTER

This invention relates to the area of headsets worn by a user in specific environments and, in particular, those areas where the background noise must be abated while simultaneously allowing speech through.

Generally, environmental noise can be broken into several areas such as noises of short duration with varying spectral characteristics such as conversations, other long duration noise with fixed or relatively fixed spectral characteristics such as operating pumps, fans, engines etc., and noises of very brief duration with "spike" characteristics such as something rattling, etc.

The instant invention is intended for use in environments found in industry, such as those requiring attenuation of low frequency noise as well as noise covering the speech band (300 - 3300 Hz).

Passive headsets have been employed and work extremely well at higher frequencies above 1000 Hz but don't perform at all in the lower frequency ranges. In these lower ranges active cancellation has been shown to work extremely well and the provision of high levels of protection are afforded users in the 50 to 1000 Hz range.

The instant headset provides protection at both the high frequency range (above 1000 Hz) as well as the low range (50 to 1000 Hz).

In the past, attempts to combine the two protections, i.e., high and low frequency attenuation, has resulted in not only the noise being attenuated but also the speech that the wearer needs to hear. Some systems met only limited success with fixed or "near-stationary" noise but not with the other noise of either (a) varying spectral characteristics or (b) brief duration noises with "spikes". Examples of such a system is found in U.S. Patent No. 4,025,721, to Graupe et al and U.S. Patent 4,185,168 to Graupe et al.

The instant invention solves the problem now existant, that of <u>total</u> attenuation of the noise and speech, by providing a solution of an active/passive headset that can employ any of several selective algorithms such as that disclosed in U.S. Patent No.

5,091,953 to Tretter, hereby incorporated by reference herein. Alternatively, it can employ the algorithm disclosed in U.S. Patent No. 5,105,377 to Ziegler which is also incorporated herein by reference. In addition, it can employ other algorithms such as that disclosed in the application of Ziegler in U.S. Patent Application No. 07/421 759 which is hereby incorporated by reference herein.

The technique of actively controlling noise is disclosed in various patents to Barry Chaplin especially U.S. Patent 4,654,871 which is hereby incorporated by reference herein.

Accordingly it is an object of this invention to provide an active/passive headset that will attenuate a wide range of environmental noise while allowing speech to be heard.

It is another object of this invention to provide a headset that blocks unwanted noise but allows speech to pass through a speech filter or "in-wire" canceller and be combined with the output of an active noise controller.

Still another object of this invention is to combine passive quieting with active noise cancellation in a headset and allow for passage of speech with simultaneous quieting of environmental noise.

These and other objects of the invention will become readily apparent when reference is had to the accompanying drawings in which

Fig. 1 shows a typical active/passive headset system incorporating the instant invention.

Fig. 2 shows a typical active/passive headset system using a digital virtual earth controller.

Fig. 3 is a more detailed block diagram of the digital virtual earth and adaptive feed forward systems shown in Figures 1 and 2.

Fig. 4 is a detailed description of an adaptive speech filter system.

Fig. 5 shows an active/passive closed back headset that combines the cancellation system with the speech filter system.

In Fig. 1 there is shown an active/passive closed back headset system 10. It consists of a typical passive headset 11, loudspeakers 12 that drive the anti-noise and residual microphones 13 to sense any remaining noise near the ear and reference microphones 14 to send advanced information for feed forward approaches and a system controller 20 which synthesizes the anti-noise signal.

The headset shown has closed backs 21, 22 for passive attenuation. Without the speakers, microphones and system controller, this headset would be a typical passive hearing protector.

The system is designed to use various algorithms such as that of Ziegler in U.S. Patent 5,105,377 or an adaptive feed forward approach by using the reference sensor as an input to the cancellation algorithm. Both these algorithms use a reference signal as inputs. The digital virtual earth (DVE) algorithm develops a reference signal by subtracting an equalized version of its own anti-noise signal from the residual signal. The adaptive feed forward uses the reference microphone as its input and is very effective on complicated noise environments that are broadband and random in character. The Least Means Square (LMS) adapter 24 shown in Fig 1 are Filtered - X versions which have inherent compensation for the effects of the feedback delays around the loop. Box "C" at 25 is the impulse response of the entire cancellation system.

Feedback compensator 26 and cancellation filter 27 complete the component portions of the controller.

DVE is highly effective to use in simple noise environments having only a few harmonics even where the noise varies tremendously.

Actuators 12 of the headset are large enough to be capable of producing anti-noise at the same level as the noise to be canceled. They have little or no distortion and have a minimum of input-to-output delay as any delay in the feedback loop slows down the system adaptation rate.

Residual sensors 13 are typically small electret microphones mounted on the speaker frame near the ear. It must faithfully reproduce the sound that remains at the

ear after cancellation so that the controller can make further adjustments to the anti-noise signal.

Reference microphones 14 are small electret microphones attached to the outside of the headset at a distance from the ear canal. This reference microphone is used to provide advanced information about the noise. The higher the frequency of the noise the more advanced information is needed to effectively cancel the noise, however, the passive muff limits the need for cancellation above 1000 Hz.

Figure 2 shows a basic block diagram of a digital virtual earth system, phenomena (such as noise) is detected by a residual sensor which sends out a sensor signal. The sensor signal is affected by filters and other factors shown in Figure 3 before reaching the processor. The DVE processor produces an output cancellation signal y. This output signal is affected by filters, and other factors. The output from the actuator combines with the original phenomena and the residual is detected at a location by the residual sensor of Figure 2.

The processor output for a specific sample k are given by

$$\underline{x}_k = \underline{r}_k - \underline{y}_{k-1} . \underline{C}_k$$

$$Y_k = -\underline{A}_k r_k$$

where

y_k is cancellation output value

$\underline{y}_{k-1}$ is a vector of previous cancellation output values

$\underline{C}_k$ is a vector of filter coefficients approximating S x E where S is the impulse response of the system after the processor and E is the impulse response of the system prior to the processor where a x b is the convolution of a and b.

$\underline{x}_k$ is a vector of estimates of the noise

$\underline{r}_k$ is a vector of most recent values of the residual signal

$\underline{A}_k$ is a vector of cancellation filter coefficients and a.b is the product of the two vectors.

When an LMS algorithm is used the cancellation filter coefficients are adapted as

$$\underline{g}_k = \underline{C}_k \underline{x}_k$$

$$A_{k+1} = A_k + \alpha_k g_k$$

where

$\underline{g}_k$ is the result of convolving $\underline{C}_k$ and $\underline{x}_k$

A_{k+1} is a vector of the most recent values of g_k and α is set for convergence.

Figure 3 shows in more detail the active cancellation system in either a DVE or AFF configuration. The active cancellation system has an input that receives residual noise signals (and reference signals in an AFF configuration) from a residual sensor (and reference sensor in the AFF configuration) and provides an output signal to an actuator. The actuator produces canceling phenomena from an electronic waveform (the output signal) into the area to be controlled (the ear canal). The canceling phenomena combines with the original phenomena so that the two sum algebraically.

Figure 4 shows in detail the adaptive speech filtering system. The system has as an input a reference signal that is composed of noise and speech from a reference sensor. From this input the adaptive speech filter separates the speech from noise and provides an output signal composed of speech only to an actuator.

Fig. 5 shows the instant invention with an adaptive speech filter 30 and the impulse "C" using a DVE cancellation approach. Reference microphones 14 not only pick up damaging noise but also pick up speech and warning signals. The embodiment of Fig. 2 is designed to provide protection against damaging noise which typically interferes with the speech band. It also attenuates noise in the audible range of the human ear effectively attenuating any warning siren. Using reference microphones 14 as inputs to a speech filter 30 that is then mixed with the anti-noise signal to be sent to the speaker is a means of allowing speech to pass through to the worker. If warning signals are outside the speech band and outside the frequency range of the active controller 20, these can be passed to the worker as well. Fig. 5 is a diagrammatic view

of the integration of active/passive hearing protection with speech (and warning signal) filtering so that local communication can be maintained for the safety of the worker.

The speech filter 30 separates speech from noise and output speech to the speaker with a minimal effect on intelligibility. The specific in-wire approach to use is dependent upon the noise but the DVE algorithm could be used. Various approaches have specific advantages in selectively filtering speech. Even the approach used in the Graupe patents can be used and since the output of filter 30 cannot interfere with the working of controller 20, it needs to be filtered from the control system as shown in Fig. 5 and, using the equations developed for the DVE algorithm, the output signal u_k is a specific sample k given by

$$w_k = \underline{T}_k \underline{v}_k$$

$$u_k = y_k + w_k$$

where

u_k is the cancellation and speech output value.

w_k is the filtered speech value.

$\underline{T}_k$ is a vector of filter coefficients determined by the adaptive speech filtering algorithm.

y_k is the cancellation output value as before.

$\underline{v}_k$ is the reference signal (speech and noise) input vector.

The residual sensor used to control the cancellation system also picks up the speech that is passed via the actuator. Therefore, the residual signal r_k must remove the speech signal w_k before input to the cancellation system so as not to interfere with the operation of the controller. It does so by modifying the residual signal r_k by

$$z_k = r_k - \underline{C}_k w_k$$

where

z_k is the filtered residual input to the controller.

$\underline{r}_k$ is the original residual input.

$\underline{C}_k$ is a vector of filter coefficients that is the overall impulse response of the cancellation system, as before.

w_k is the filtered output speech signal that is output to the actuator with the cancellation signal y_k.

The need to remove $\underline{C}_k w_k$ from the residual signal may not be required for all applications, for example, a siren canceling headset system will not try and cancel speech while the siren is in operation.

While the preferred embodiment of the invention is described it will be obvious to those of ordinary skill in the art that many changes and modifications can be made without departing from the scope of the appended claims.

CLAIMS

What is claimed is:

1. An active/passive noise canceling headset for attenuating high and low frequency noise, said headset comprising

a headset means having closed back earpiece means adapted to prevent external noise from entering the ear through passive attenuation,

speaker means associated with said earpiece means and adapted to produce anti-noise of the same level as the noise to be canceled,

reference sensing means also associated with said earpiece means and adapted to produce advanced information about the noise to be canceled, and

controller means adapted to secure information from said reference sensing means and cause emission of anti-noise by said speaker means.

2. A headset as in claim 1 and including residual sensing means also associated with said earpiece means and adapted to produce a signal characterizing the sound remaining at the ear.

3. A headset as in claim 2 and including a feedback loop means connecting said residual sensing means to said controller means and whereby said controller means makes adjustments to the anti-noise in response to the sound sensed by said residual sensing means.

4. A headset as in claim 3 wherein said reference sensing means are electret microphones or other type of microphones.

5. A headset as in claim 3 wherein said residual sensing means are electret microphones.

6. A headset as in claim 1 and wherein said controller means contains an adaptive speech filter means adapted to attenuate disturbing noise in the audible range of the human ear yet leave the speech to the headset wearer intelligible.

7. The method of controlling the amount of noise interfering with speech, said method comprising sensing signal containing speech and noise and converting it to electronic signals, using "in-wire" control approach producing "anti-noise" to cancel said noise component of the signal.

8. The method of claim 7 and including the step of passively quieting the higher frequency portions of the noise to be canceled simultaneous with the production of anti-noise.

9. The method of claim 8 and including the further step of sensing the noise remaining adjacent the ear after cancellation by said anti-noise and changing the anti-noise production in response to the amount of remaining noise.

10. The method of claim 9 and including the step of mixing the signals from sensing the noise containing noise and speech with the signal producing the anti-noise thus allowing speech to pass to the ear.

11. The method of filtering speech from noise said method comprising sensing signal containing speech and noise and converting it to electronic signals, adaptively filtering the signal to separate the noise component from the speech component and producing a signal that contains only the speech component.

12. The method of manual selection of either the method of claim 7 or the method of claim 8 or both based upon characteristics of the noise environment wherein

said manual selection is a switch that allows the user to select a preferred approach for allowing speech to be most intelligible.

13. The method of automatic selection of either the method of claim 7 or the method of claim 8 or both based upon the spectral and temporal characteristics of the noise environment wherein said automatic selection is achieved using a detection algorithm that continuously monitors the spectral and temporal characteristics of the noisy signal.

14. The method as in claim 10 wherein said selection of adaptive speech filtering occurs when the noise is near-stationary and of limited bandwidth.

15. The method as in claim 10 wherein said selection of in-wire control occurs when the noise is varying and tonal or broadband.

16. The method of manual selection of claim 12 wherein a setting to automatic selection can be made.

17. The method of manual selection of claim 12 wherein a setting to an off position can be made.

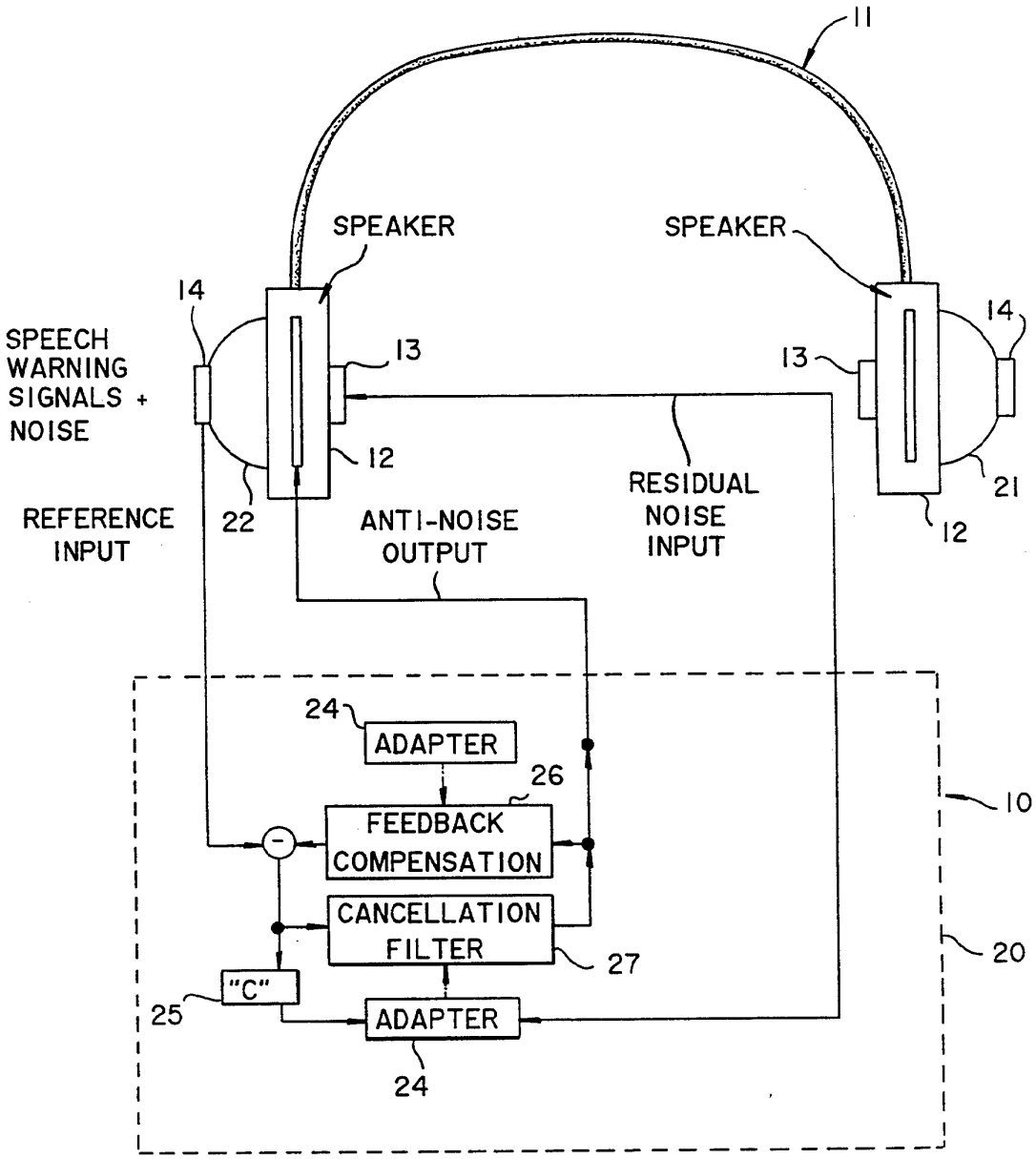

11
SPEAKER
SPEAKER
14
14
SPEECH
WARNING
SIGNALS +
NOISE
13
13
12
21
REFERENCE
INPUT
22
ANTI-NOISE
OUTPUT
RESIDUAL
NOISE
INPUT
12
24
ADAPTER
26
10
FEEDBACK
COMPENSATION
CANCELLATION
FILTER
27
20
"C"
25
ADAPTER
24

FIG.1

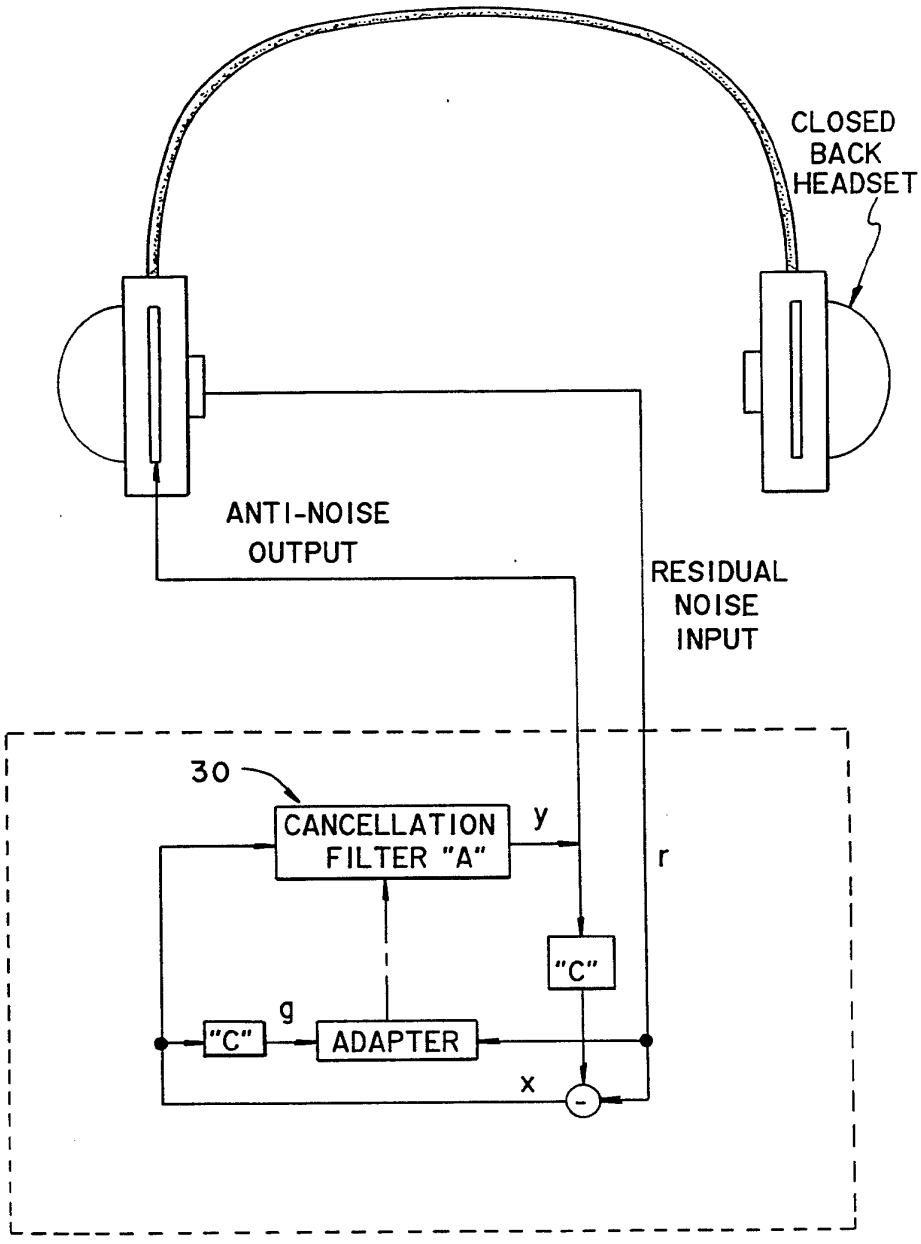

CLOSED
BACK
HEADSET
ANTI-NOISE
OUTPUT
RESIDUAL
NOISE
INPUT
30
CANCELLATION
FILTER "A"
y
r
"C"
g
"C"
ADAPTER
x
-

FIG.2

SUBSTITUTE SHEET

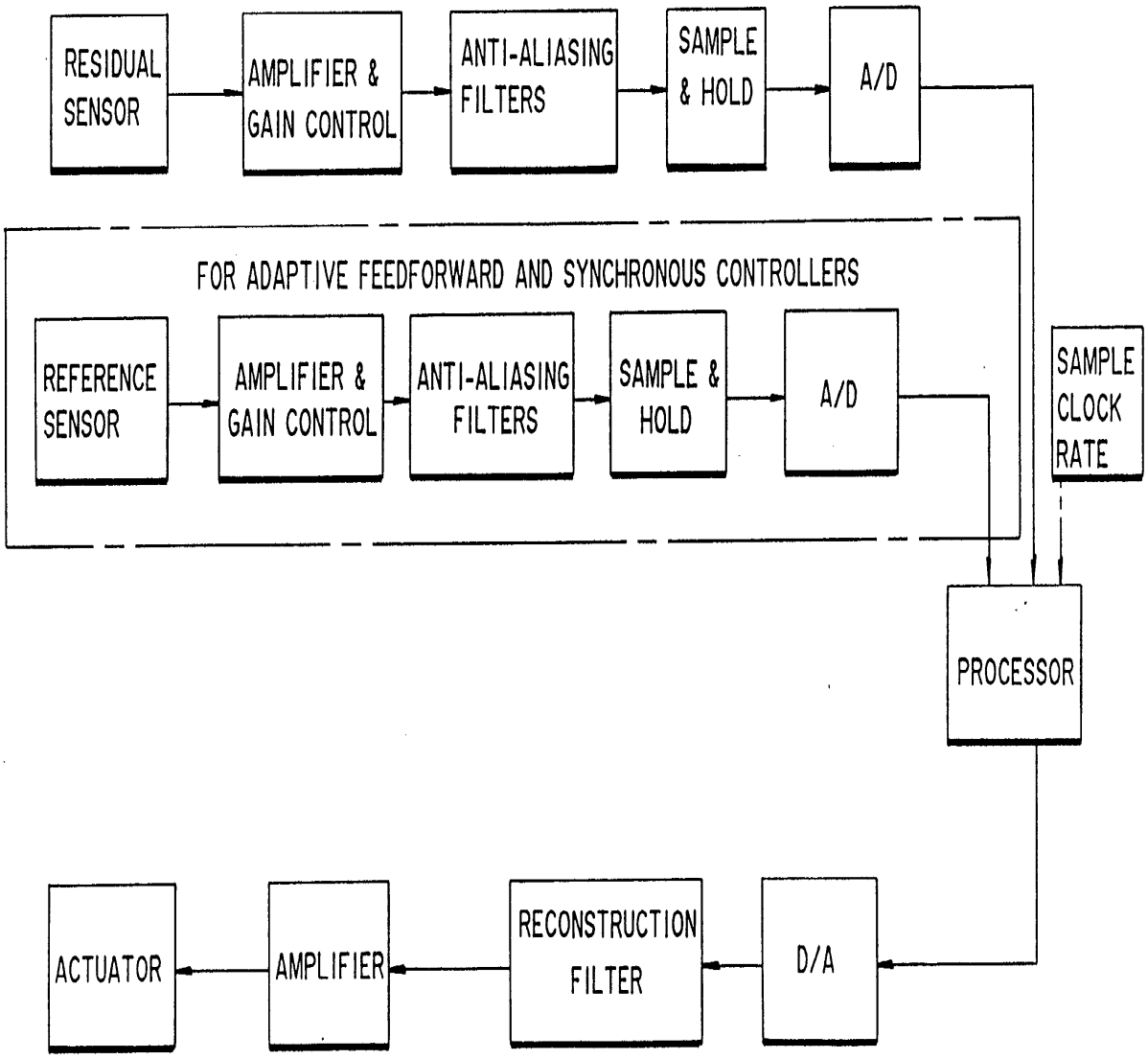


FIG.3

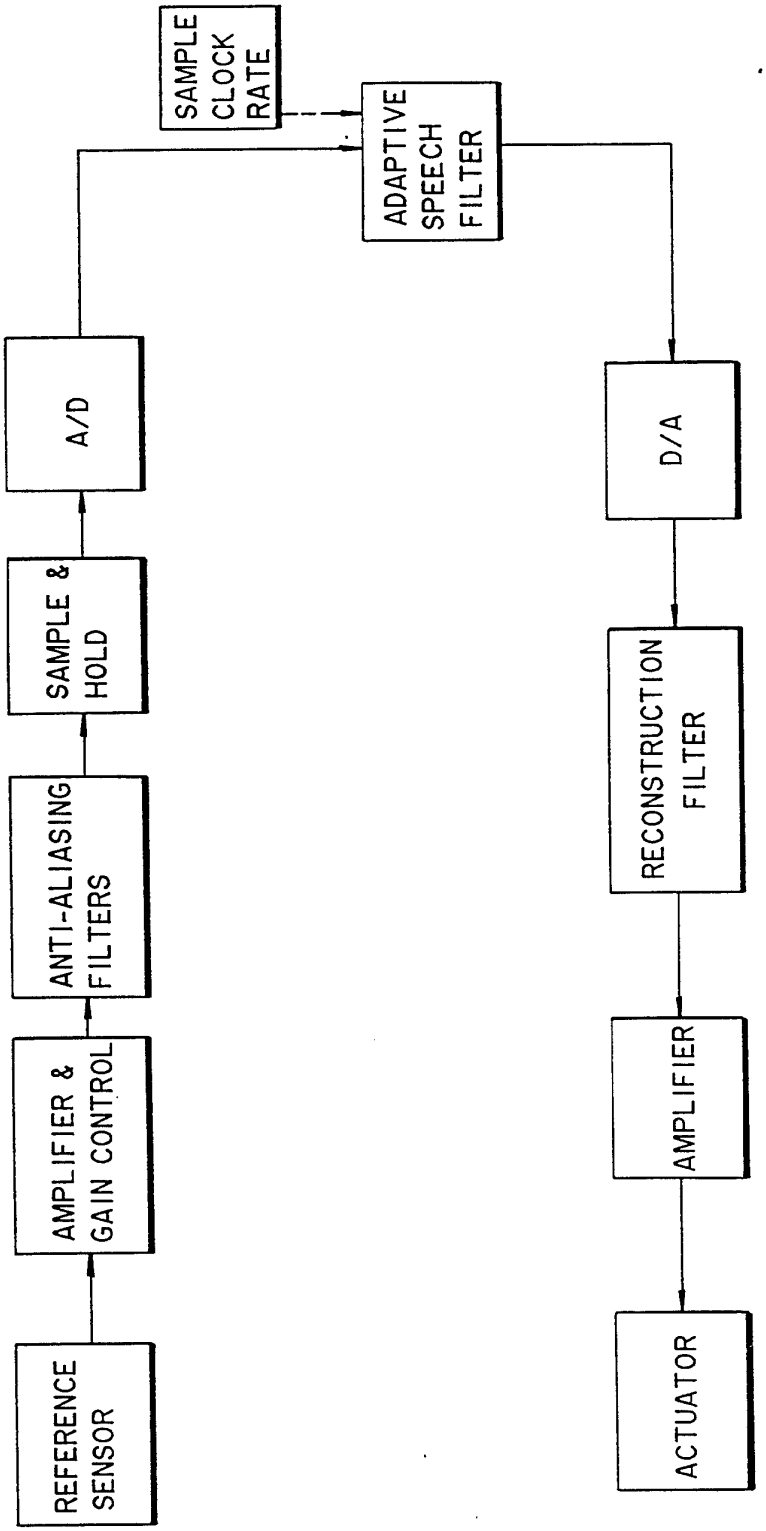

SAMPLE CLOCK RATE
ADAPTIVE SPEECH FILTER
A/D
SAMPLE & HOLD
ANTI-ALIASING FILTERS
AMPLIFIER & GAIN CONTROL
REFERENCE SENSOR
D/A
RECONSTRUCTION FILTER
AMPLIFIER
ACTUATOR

FIG.4

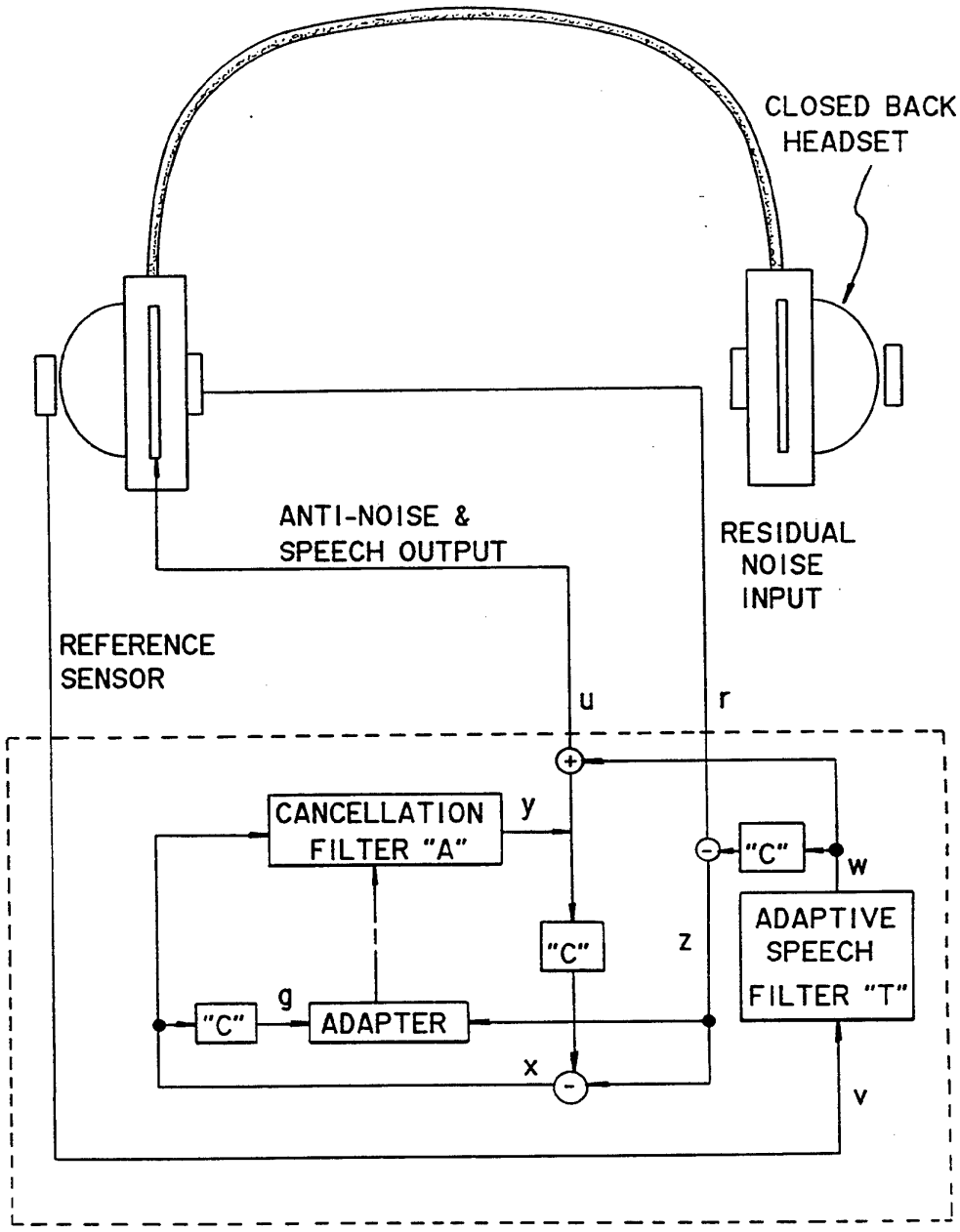

CLOSED BACK HEADSET
ANTI-NOISE & SPEECH OUTPUT
RESIDUAL NOISE INPUT
REFERENCE SENSOR
u
r
CANCELLATION FILTER "A"
y
"C"
w
ADAPTIVE SPEECH FILTER "T"
z
"C"
g
ADAPTER
x
v

FIG.5

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/04568

A. CLASSIFICATION OF SUBJECT MATTER

IPC(5) :H03B 29/00
US CL :381/71

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/72, 74

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X Y	US, A, 3,952,158 (Kyle et al.) 20 April 1976 See Figs. 1-5	1 1-6
X Y	US, A, 4,061,079 (Freifeld) 06 December 1977 See Figs. 2-5	1 2-6
Y,E	US, A, 5,126,681 (Ziegler, Jr. et al.) 30 June 1992 See Abstract and Figs. 1-3	7-17
Y	US, A, 5,105,377 (Ziegler, Jr.) 14 April 1992 See Abstract and Figs. 1-5	1-17
Y	US, A, 4,953,217 (Twiney et al.) 28 August 1990 See Abstract Fig. 1	2-6
Y	US, A, 5,091,953 (Tretter) 25 February 1992 See Abstract and Figs. 1 and 5	7-17
Y	US, A, 4,064,362 (Williams) 20 December 1977 See Abstract and Fig. 1	1
A	US, A, 4,677,678 (McCutchen) 30 June 1987 See Abstract and Figs. 1-7	1-6
A	US, A, 4,455,675 (Bose et al.) 19 June 1984 See Abstract and Figs. 1 and 5-6	1-6

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

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Date of the actual completion of the international search 06 AUGUST 1992	Date of mailing of the international search report 09 NOV 1992
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INTERNATIONAL SEARCH REPORT

International application No.
PCT/US92/04568

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US, A, 5,046,103 (Warnaka et al.) 03 September 1991 See Abstract and Figs. 1-2	1-6

Form PCT/ISA/210 (continuation of second sheet)(July 1992)*

PTO/SB/08a (01-22)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	4,533,795	A	1985-08-06	Baumhauer, Jr. et al.	
	2	5,251,263		1993-10-05	Andrea et al.	
	3	6,647,368	B2	2003-11-11	Nemirovski	
	4	6,671,379	B2	2003-12-30	Nemirovski	
	5	6,728,385	B2	2004-04-27	Kvaloy et al.	
	6	6,754,359	B1	2004-06-22	Svean et al.	
	7	7,502,484	B2	2009-03-10	Ngia et al.	
	8	7,903,826	B2	2011-03-08	Boersma	

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	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

	9	7,953,241	B2	2011-05-31	Jorgensen et al.	
	10	7,983,433	B2	2011-07-19	Nemirovski	
	11	8,014,553	B2	2011-09-06	Radivojevic et al.	
	12	8,150,084	B2	2012-04-03	Jessen et al.	
	13	8,351,634	B2	2013-01-08	Khenkin	
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	1	20010046304	A1	2001-11-29	Rast	
	2	20030035551	A1	2003-02-20	Light et al.	
	3	20040047486	A1	2004-03-11	Van Doom et al.	

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	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

	4	20060083388	A1	2006-04-20	Rothschild	
	5	20060153394	A1	2006-07-13	Beasley	

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	1	1519625	EP	A2	2005-03-30	Starkey Laboratories Inc		
	2	1640972	EP	A1	2006-03-29	Sonova Holding AG		

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	1	90/019,169, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., REQUEST FOR EX PARTE REEXAMINATION OF U.S. PATENT NO. 11,244,666	
	2	Oshana, DSP Software Development Techniques for Embedded and Real-Time Systems, Introduction, pp. xi-xvii (2006)	
	3	Mulgrew et al., Digital Signal Processing: Concepts and Applications, Introduction, pp. xxiii-xxvi (2nd ed. 2002)	

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	Art Unit	2651
	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

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	Examiner Name	Lunsee Lao
	Attorney Docket Number	11472-39-5_CONT2-16247186

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OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-04-07
Name/Print	Peter A. Chiabotti	Registration Number	54603

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7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

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9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Espacenet

JP6389232B2 パーソナルオーディオデバイスのための短待ち時間マルチドライバ適応雑音消去（ＡＮＣ）システム

Applicants: シラス ロジック、インコーポレイテッド

Inventors: ヘンドリックス， ジョン ディー．,アルダーソン， ジェフリー,アブドラーザデー， ミラーニ アリ,ジョー， ダヨン,ル， ヤン

Classifications:
IPC ***G10K11/178; H04R1/10; H04R3/14;***

CPC **G10K11/17854 (EP,US); G10K11/17855 (EP,US); G10K11/17881 (EP,US); G10K11/17885 (EP,US); H04R1/1083 (EP,US); H04R1/24 (EP,US); H04R3/005 (US); H04R3/12 (EP,US);** G10K2210/1081 (EP,US); G10K2210/3019 (EP,US); G10K2210/3028 (US); H04R2410/05 (US); H04R2460/01 (EP,US); H04R3/14 (EP);

Priorities: US201313968007A 2013-08-15; US201361783267P 2013-03-14; US2014016833W 2014-02-18

Application: JP2016500286A·2014-02-18

Publication: JP6389232B2·2018-09-12

Published as: **CN105074814A; CN105074814B; EP2973540A1; EP2973540B1; EP3410431A1; EP3410431B1; JP2016510915A;** JP6389232B2; **KR10215971B1; KR20150127268A; US2014270222A1; US2016316291A1; US9414150B2; US9955250B2; WO2014158449A1**

パーソナルオーディオデバイスのための短待ち時間マルチドライバ適応雑音消去（ＡＮＣ）システム

Abstract

Abstract not available for JP6389232B2 – abstract of corresponding document: JP2016510915A

ソースオーディオ信号の異なる周波数帯を再現するための複数の出力変換器を含むパーソナルオーディオデバイスが、反雑音信号を発生させるために周囲オーディオを測定する、少なくとも１つのマイクロホン信号から、変換器のそれぞれのための反雑音信号を適応的に発生させる、適応雑音消去（ＡＮＣ）回路を含む。反雑音信号は、その対応する変換器において周囲オーディオの実質的消去を生じさせるように、反雑音信号が、別個の適応フィルタによって発生される。別個の適応フィルタの使用は、クロスオーバが、反雑音を適切な周波数帯に分割することを必要とされないため、短待ち時間動作をもたらす。適応フィルタは、特定の適応フィルタに対応する周波数帯のみにおいて、反雑音を発生させるように実装またはバイアスされることができる。反雑音信号は、対応する変換器のための出力を提供するために、適切な周波数帯のソースオーディオと組み合わせられる。

JP6389232B2

パーソナルオーディオデバイスのための短待ち時間マルチドライバ適応雑音消去（ＡＮＣ）システム

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A short latency multi-driver adaptive noise cancellation (ANC) system for personal audio devices

[0001] FIELD OF THE INVENTION The present invention relates generally to personal audio devices, including adaptive noise cancellation (ANC) and multiple drivers for different frequency bands.

[0002] BACKGROUND OF THE INVENTION Mobile consumer / mobile phones, wireless telephones such as cordless phones, and other consumer audio devices such as MP3 players are widely used. The performance of such devices with respect to clarity can be determined by providing an ANC using a reference microphone to measure the ambient sound event and then using signal processing to insert the anti-noise signal into the output of the device, Can be improved by eliminating acoustic events.

[0003] Most audio systems implemented for personal audio devices rely on a single output transducer but when transducers or ear speakers mounted on the enclosure of a radiotelephone are used, In the case of a pair of transducers when a telephone or other device employs a stereo speaker, separate transducers for high frequencies and low frequencies are used for high quality audio reproduction, as in high quality ear speakers It may be desirable to provide it. However, when implementing ANC in such a system, the latency introduced by the crossover that divides the signal between the low frequency converter and the high frequency converter results in a delay, Reduce the effectiveness of the ANC system due to the waited time.

[0004] Accordingly, it would be desirable to provide a personal audio system that includes a radiotelephone and / or ear speaker that provides short latency ANC operation while using multiple output transducers that handle different frequency bands.

[0005] DISCLOSURE OF THE INVENTION The above stated objectives of providing a personal audio device with ANC and employing multiple output transducers for handling different frequency bands are achieved in a personal audio system, method of operation, and integrated circuit .

[0006] The personal audio device includes a low frequency output converter for reproducing a source audio signal for reproduction to the listener and an anti-noise signal for canceling the influence of ambient audio sound on the acoustic output of the transducer, Converters are included. Personal audio devices also include integrated circuits that provide adaptive noise cancellation (ANC) functionality. The method is a personal audio system and a method of operating an integrated circuit. A reference microphone is mounted on the device housing to provide a reference microphone signal indicative of ambient audio sound. The personal audio system further includes an ANC processing circuit for adaptively generating the anti-noise signal from the reference microphone signal such that the anti-noise signal causes substantial cancellation of the ambient audio sound in its corresponding transducer Including. An adaptive filter is used to generate the anti-noise signal by filtering the reference microphone signal.

[0007] The foregoing and other objects, features, and advantages of the present invention will be apparent from the following more particular description of preferred embodiments of the invention, as illustrated in the accompanying drawings. The specification of the present application also provides, for example, the following items. (Item 1) A personal audio system, wherein the personal audio system is a source of audio for reproduction, the source of the audio providing a source audio signal, a source of audio, a first transformation Wherein the first transducer comprises a first transducer for canceling the influence of the source audio signal for reproduction to the listener and the ambient audio tone in the acoustic output of the first transducer, A first transducer for reconstructing the high frequency content with the anti-noise signal and a second transducer for converting the source audio signal for reproduction to the listener, A second transducer for reconstructing low frequency content with a second anti-noise signal for canceling the influence of ambient audio sound in the acoustic output of the second transducer; At least one microphone for providing at least one microphone signal

indicative of a diopter sound and a second filter for using the first filter to match the at least one microphone signal with the first anti- Processing circuitry for generating a second anti-noise signal from the at least one microphone signal to reduce the presence of the ambient audio sound at the first converter and the second converter, the processing circuit Uses the second filter to generate the second anti-noise signal from the at least one microphone signal to coincide with the at least one microphone signal, and wherein the first transducer and the And processing circuitry for reducing the presence of said ambient audio sound in said second transducer Ioshisutemu. The first filter is a first adaptive filter having a first response adapted to reduce the presence of the ambient audio sound, and the second filter is a first adaptive filter having a first response adapted to reduce the presence of the ambient audio sound, Wherein the first adaptive filter is a second adaptive filter adaptive to reduce the presence. The processing circuit may limit the content of the first anti-noise signal to the first predefined frequency by limiting the first frequency response of the first adaptive filter to a first predetermined frequency range, The processing circuit limits the content of the second anti-noise signal to the second predefined frequency by limiting the second response of the second adaptive filter to a second predefined frequency range, Wherein the first predefined frequency range and the second predetermined frequency range are substantially different from each other, the personal audio device according to item 1. 4. The system of claim 1, further comprising an error microphone for providing an error microphone signal indicative of ambient audio sound and acoustic output of the first transducer and the second transducer, wherein the first adaptive filter comprises: Comprising a first coefficient generator adapted to minimize components of a reference microphone signal present in a microphone signal, said processing circuit comprising: a first coefficient generator adapted to receive a first signal Limiting the adaptation of the first frequency response by modifying the frequency content of the first microphone signal and the second adaptive filter to adapt to minimize the components of the reference microphone signal present in the error microphone signal Wherein the processing circuit includes a second coefficient generator that modifies the frequency content of the second signal input to the second coefficient generator Therefore, to limit the application of the first frequency response, the personal audio device of claim 3. (Item 5) The processing circuit may further include a first additional signal having first predetermined frequency content within the first predetermined frequency range, the first additional signal input to the first coefficient generator, , The processing circuit modifies the frequency content of the first signal input to the first coefficient generator and the processing circuit changes the second predetermined frequency content within the second predetermined frequency range to The frequency content of the second signal input to the second coefficient generator is set to be equal to or lower than the frequency content of the second signal input to the second coefficient generator by inputting a second additional signal having the second frequency component to the second signal input to the second coefficient generator The personal audio device according to item 4 which alters. 6. The personal audio device of claim 5, wherein the first additional signal and the second additional signal are noise signals. 7. The system of claim 1, wherein the processing circuitry receives the source audio signal to provide a crossover that generates a higher frequency content source audio signal and a lower frequency content source audio signal, Wherein said processing circuit further combines said higher frequency content source audio signal with said first anti-noise signal and combines said lower frequency content source audio signal with said second anti-noise signal, item 1 To the personal audio device. 8. The personal audio device of claim 1, wherein the first transducer is a high frequency transducer of an ear speaker and the second transducer is a low frequency transducer of the ear speaker. And a third transducer, wherein the third transducer comprises: a second source audio signal; and a third transducer, wherein the third transducer comprises a second source audio signal and a third transducer for canceling the influence of the ambient audio tone in the acoustic output of the third transducer A fourth transducer for reconstructing the high frequency content of the first source audio signal and the anti-noise signal of the third source audio signal, and a fourth transducer, And a fourth anti-noise signal for canceling the influence of the ambient audio sound in the acoustic output of the transducer of the third transducer, wherein the processing circuit further comprises: , Generating said third anti-noise signal and said fourth anti-noise signal from said at least one microphone signal to match said at least one microphone signal using said third transform vessel The processing circuit reduces the presence of the ambient audio sound in the at least one microphone by using the fourth filter to match the at least one microphone signal with the fourth anti- From the signal to reduce the presence of the ambient audio sound in the fourth transducer. 10. A method for canceling the influence of ambient audio sounds by a personal audio system, the method comprising: measuring ambient audio sound using at least one microphone to generate at least one microphone signal , Firstly generating a first anti-noise signal from said at least one microphone signal to coincide with said at least one microphone signal using a first filter, Reducing the presence of the ambient audio sound and secondly using a second filter to convert the second anti-noise signal to the at least one microphone signal To reduce the presence of the ambient audio sound in the second transducer Providing a source of audio for reproduction, the source of the audio providing a source audio signal; and using the first converter to convert the source audio signal and the first Reproducing the high frequency content with the anti-noise signal of the source audio signal and the second anti-noise signal; and reproducing the low frequency content of the source audio signal and the second anti-noise signal using the second converter, Method. 11. The method of claim 1, wherein

the first filter is a first adaptive filter having a first response adapted to reduce the presence of the ambient audio sound, and the second filter is a first adaptive filter Wherein the first adaptive filter is a second adaptive filter adapted to reduce the presence. 12. The method of claim 1, wherein the first generating comprises: limiting the first frequency response of the first adaptive filter to a first predefined frequency range such that the content of the first anti- 1, the second occurrence further comprising limiting the second response of the second adaptive filter to a second predefined frequency range, the second generating further comprising limiting the second response of the second adaptive filter to a second predefined frequency range, And limiting the content of the anti-noise signal to the second predefined frequency range, wherein the first predefined frequency range and the second predetermined frequency range are substantially different. 13. The method of claim 1, further comprising measuring an ambient audio and acoustic output of the first transducer and the second transducer using an error microphone to generate an error microphone signal, wherein the first Comprises adapting the coefficients of a first coefficient generator that controls said first frequency response to minimize the components of the reference microphone signal present in said error microphone signal , Said second generating adapting coefficients of a second coefficient generator controlling a second frequency response in order to minimize the components of the reference microphone signal present in the error microphone signal Wherein the first generating is performed by modifying the frequency content of the first signal input to the first coefficient generator Limiting the adaptation of the first frequency response and the second generating further comprises modifying the frequency content of the second signal input to the second coefficient generator so that the second frequency The method according to item 12, wherein the adaptation of the response is restricted. 14. The method of claim 1, wherein the first generating further comprises generating a first additional signal having a first predetermined frequency content within the first predefined frequency range to at least one 1 signal input to limit adaptation of the first frequency response and the second occurrence is a second second frequency content having a second predetermined frequency content within the second predetermined frequency range Limiting the adaptation of the second frequency response by introducing an additional signal of the second frequency response to at least one second signal input to the second coefficient generator. 15. The method of claim 14, wherein the first additional signal and the second additional signal are noise signals. 16. The method of claim 1, further comprising: receiving the source audio signal and filtering the source audio signal to implement a crossover that generates a higher frequency content source audio signal and a lower frequency content source audio signal , Further comprising combining said higher frequency content source audio signal with said first anti-noise signal and combining said lower frequency content source audio signal with said second anti-noise signal. the method of. 17. The method of claim 10, wherein the first transducer is a high frequency transducer of an ear speaker and the second transducer is a low frequency transducer of the ear speaker. In order to counteract the influence of the ambient audio sound in the acoustic output of the third transducer, the third transducer is used to increase the second source audio signal and the third anti-noise signal high Reproducing the frequency content and using the fourth converter to cancel out the influence of the ambient audio sound in the acoustic output of the fourth transducer by comparing the second source audio signal and the fourth counter Reproducing the low frequency content with the noise signal and using the third filter to convert the third anti-noise signal and the fourth anti-noise signal to coincide with the at least one microphone signal Generating from said at least one microphone signal to reduce the presence of said ambient audio sound in said third transducer and said fourth transducer; Use, to match the at least one microphone signal,Generating the fourth anti-noise signal from the at least one microphone signal and reducing the presence of the ambient audio sound at the third transducer and the fourth transducer. . 19. An integrated circuit for implementing at least a portion of a personal audio system, the integrated circuit being a source of audio for reproduction, the source of the audio providing a source audio signal , A source of audio, and a first output for providing a first output signal to a first transducer, said first transducer comprising: a source audio signal and said first transducer A first output for reconstructing high frequency content with a first anti-noise signal for canceling the influence of ambient audio sound in the acoustic output of the second transducer, and for providing a second output signal to the second transducer A second output of the second source audio for playback to the listener and an ambient audio sound in the acoustic output of the second ear speaker, A second output for reconstructing a second audio signal that includes both a first anti-noise signal for said ambient audio sound and at least one microphone input for providing at least one microphone signal indicative of said ambient audio sound; Generating the first anti-noise signal and the second anti-noise signal from the at least one microphone signal to coincide with the at least one microphone signal using a first filter, Processing circuitry for reducing the presence of the ambient audio sound in one transducer and the second transducer, wherein the processing circuitry uses a second filter to match the at least one microphone signal So as to generate said second anti-noise signal from said at least one microphone signal, said first transducer and Serial wherein reducing the presence of the ambient audio sound in the second transducer, and a processing circuit, an integrated circuit. 20. The method of claim 19, wherein the first filter is a first adaptive filter having a first response adapted to reduce the presence of the ambient audio sound, and the second filter is a first adaptive filter Wherein the first adaptive filter is a second adaptive filter adapted to reduce presence. 21. The apparatus of claim 19, wherein the processing circuit is configured to limit the content of the

first anti-noise signal to the first predetermined frequency The processing circuit limits the content of the second anti-noise signal to the second predefined frequency by limiting the second response of the second adaptive filter to a second predefined frequency range, Wherein the first predefined frequency range and the second predetermined frequency range are substantially different, the integrated circuit according to item 19. 22. The system of claim 1, further comprising an error microphone for providing an error microphone signal indicative of ambient audio sound and audio output of the first transducer and the second transducer, wherein the first adaptive filter comprises: Comprising a first coefficient generator adapted to minimize components of a reference microphone signal present in a microphone signal, said processing circuit comprising: a first coefficient generator adapted to receive a first signal Limiting the adaptation of the first frequency response by modifying the frequency content of the first microphone signal and the second adaptive filter to adapt to minimize the components of the reference microphone signal present in the error microphone signal , And the processing circuit modifies the frequency content of the second signal input to the second coefficient generator. By limiting the adaptation of the first frequency response, the integrated circuit of claim 21. 23. The apparatus of claim 1, wherein the processing circuit is configured to generate a first additional signal having a first predetermined frequency content within the first predetermined frequency range, the first additional signal input to the first coefficient generator, , The processing circuit modifies the frequency content of the first signal input to the first coefficient generator and the processing circuit changes the second predetermined frequency content within the second predetermined frequency range to Of the second signal input to the second coefficient generator by inputting a second additional signal having the second signal to the second signal input to the second first coefficient generator The integrated circuit according to item 22, wherein the frequency content is modified. 24. The integrated circuit of claim 23, wherein the first additional signal and the second additional signal are noise signals. 25. The apparatus of claim 25, wherein the processing circuitry receives the source audio signal to provide a crossover that generates a higher frequency content source audio signal and a lower frequency content source audio signal, Wherein said processing circuit further combines said higher frequency content source audio signal with said first anti-noise signal and combines said lower frequency content source audio signal with said second anti-noise signal, item 19 . 26. The integrated circuit of claim 19, wherein the first transducer is a high frequency transducer of an ear speaker and the second transducer is a low frequency transducer of the ear speaker. 27. A third output for providing a third output signal to a third transducer, the third transducer comprising: a second source audio signal; and a third transducer And a third anti-noise signal for canceling the influence of the ambient audio sound in the acoustic output of the third transducer, to provide a fourth output signal to the fourth transducer Wherein the fourth converter comprises a fourth output for canceling the influence of the second source audio signal and the ambient audio sound in the acoustic output of the fourth transducer, And a fourth output for reconstructing the low frequency content with the signal, the processing circuit further comprising: a third filter for using the third filter to match the at least one microphone signal with the third Anti-noise signal and said fourth Generating a noise signal from the at least one microphone signal and reducing the presence of the ambient audio sound in the third converter and the fourth converter, the processing circuitry using a fourth filter Generating the fourth anti-noise signal from the at least one microphone signal so as to coincide with the at least one microphone signal, and generating the fourth anti-noise signal from the at least one microphone signal, The integrated circuit according to item 26, which reduces the presence. 28. A personal audio system, comprising: a plurality of output transducers; at least one microphone for providing at least one microphone signal indicative of ambient audio sound; and an adaptive noise cancellation A plurality of adaptive filters for generating a plurality of anti-noise signals for a corresponding one of the plurality of output transducers and for generating a plurality of anti-noise signals corresponding to the corresponding frequency bands Wherein said plurality of frequency converters operate as crossovers for separating said at least one microphone signal into said plurality of frequency bands corresponding to said plurality of output converters by generating said plurality of anti-noise signals within said plurality of output converters. 29. A method of canceling the influence of ambient audio sound by a personal audio system, the method comprising: measuring ambient audio sound using at least one microphone to generate at least one microphone signal And generating a plurality of anti-noise signals for providing to a corresponding one of the plurality of output transducers using a corresponding one of the plurality of adaptive filters, A corresponding adaptive filter generates the plurality of anti-noise signals in corresponding frequency bands of the plurality of frequency bands to generate the at least one microphone signal as the plurality of output transducers corresponding to the plurality of output transducers Operate as a crossover to separate into frequency bands. 30. An integrated circuit for implementing at least a portion of a personal audio system, the integrated circuit comprising: a plurality of input transducers for providing a plurality of output signals to corresponding ones of the plurality of output transducers At least one microphone input for receiving at least one microphone signal indicative of ambient audio sound and processing circuitry for implementing adaptive noise cancellation, wherein the plurality of adaptive filters are adapted to receive the plurality of outputs Generating a plurality of anti-noise signals at a corresponding one of the plurality of frequency bands, and generating the plurality of anti-noise signals within a corresponding one of the plurality of frequency bands,

thereby generating the at least one microphone signal, And operates as a crossover to separate into the plurality of frequency bands corresponding to the output transducers of the plurality of frequency converters.

[0008] FIG. 1A is an illustration of an exemplary radiotelephone 10 and a pair of earphones EB 1 and EB 2. FIG. 1B is a schematic diagram of a circuit inside the radiotelephone 10. FIG. 2 is a block diagram of a circuit inside the radiotelephone 10. FIG. 3 is a block diagram depicting the signal processing circuitry and functional blocks of various exemplary ANC circuits that can be used to implement the ANC circuit 30 of the CODEC integrated circuit 20A of FIG. FIG. 4 is a block diagram depicting signal processing circuitry and functional blocks within the CODEC integrated circuit 20.

[0009] BEST MODE FOR CARRYING OUT THE INVENTION The present invention encompasses noise cancellation techniques and circuits that can be implemented in personal audio systems such as wireless telephones and connected earphones. The personal audio system measures and adjusts the ambient acoustic environment at an earphone or other output transducer location, such as on a personal audio device enclosure that receives or generates a source audio signal, adaptive noise It includes erase (ANC) circuit. To provide high quality audio output, multiple transducers are used, including low frequency and high frequency converters to reproduce the corresponding source audio frequency bands. The ANC circuit generates a separate anti-noise signal that is provided to each of the plurality of transducers to eliminate ambient sound events at the transducer. A reference microphone is provided for measuring the ambient acoustic environment and this means that the short wait time is maintained by eliminating the need for a crossover filtering the generated anti- Providing an input to a separate adaptive filter that generates a signal. The source audio crossover can then be placed prior to the adder of the component specific to the source audio frequency band and its corresponding anti-noise signal, the adaptive filter being suitable for its corresponding transducer And can be controlled so as to generate anti-noise only within the frequency range.

[0010] FIG. 1A shows a wireless telephone 10 and a pair of earphones EB 1 and EB 2, respectively, attached to the listener's corresponding ears 5 A, 5 B. Although the illustrated radiotelephone 10 is an example of a device in which the techniques disclosed herein may be employed, it is to be understood that all elements or configurations illustrated within the circuit depicted in the radiotelephone 10 or subsequent illustrations , It is understood that it is not required. Wireless telephone 10 is connected to earphones EB 1, EB 2 by a wired or wireless connection, eg BLUETOOTH® connection (BLUETOOTH® is a trademark of Bluetooth® SIG, Inc.) . Earphones EB 1 and EB 2 each receive source audio, including input of remote utterances, ringing tones, stored audio program material, and near-end utterances (ie user utterances of wireless telephone 10) received from the radiotelephone 10 Respectively, a corresponding pair of converters SPKLH / SPKLL and SPKRH / SPKRL. The transducers SPKLH and SPKRH are high frequency transducers or "tweeters" that reproduce the audible frequency and the higher range of the transducer SPKLL, SPKRL is a low frequency transducer or transducer that reproduces the lower range of the audio frequency "Woofer" is. The source audio is also required to recreate the audio indicator such as source audio from a web page or other network communication received by the wireless telephone 10 and low battery capacity and other system event notification , And any other audio. The reference microphones R1, R2 are provided on the surface of the housing of the individual earphones EB1, EB2 to measure the ambient sound environment. Another pair of microphones, the error microphones E1, E2, when the earphones EB1, EB2 are inserted into the outer part of the ears 5A, 5B, the respective transducer pairs close to the corresponding ears 5A, 5B Is provided to further improve ANC operation by providing measurements of ambient audio combined with audio reproduced by SPKLH / SPKLL and SPKRH / SPKRL.

[0011] Radio telephone 10 places anti-noise signals on SPKLH, SPKLL, SPKRH, and SPKRL and improves the clarity of distant utterances and other audio reproduced by converters SPKLH, SPKLL, SPKRH, and SPKRL. Adaptive Noise Erase (ANC) circuits and features. An exemplary circuit 14 within the radiotelephone 10 includes an audio integrated circuit 20 that receives signals from the reference microphones R 1, R 2, the near speech microphone NS, and the error microphones E 1, E 2 and includes an RF integrated Interface with other integrated circuits such as circuit 12. In other implementations, the circuits and techniques disclosed herein may be implemented in a single integrated circuit containing control circuitry and other functionality for implementing the entirety of a personal audio device, such as an MP3 player on chip integrated circuit For example. Alternatively, the ANC circuit may be included in a module located along the wired connection in the housing of the earpiece EB 1, EB 2 or between the radiotelephone 10 and the earphones EB 1, EB 2. For illustrative purposes, the ANC circuitry is described as being provided within the radiotelephone 10, but the aforementioned variations are understandable to those skilled in the art, and the earphone EB 1, EB 2, the radiotelephone 10, and The resulting signals required between the third modules can be easily determined for those variants on demand. The close speech microphone NS captures the near-end utterance provided to the housing of the radiotelephone 10 and transmitted from the radiotelephone 10 to other conversation participants.

Alternatively, the near-speech microphone NS may be mounted on a support member affixed to one of the earphones EB 1, EB 2 on the outer surface of one of the earphones EB 1, EB 2, or one or both of the radiotelephone 10 and the earphones EB 1, EB 2 Or on an appendage located between the two.

[0012] 1B shows a reference microphone R1, which provides measurements of the ambient audio perimeter 1, perimeter 2, filtered by the ANC processing circuitry in the audio integrated circuits 20A, 20B located in the corresponding earphones EB1, EB2, A simplified schematic of audio integrated circuits 20 A, 20 B, including ANC processing, so as to be coupled to R 2. The audio integrated circuits 20A, 20B may alternatively be combined into a single integrated circuit such as an integrated circuit 20 in the radiotelephone 10. The audio integrated circuits 20A, 20B are amplified by the associated one of the amplifiers A1-A4 and provide the outputs for their corresponding channels, which are provided to the corresponding converter pairs SPKLH / SPKLL and SPKRH / SPKRL generate. The audio integrated circuits 20 A, 20 B receive (from the reference microphones R 1, R 2, the near utterance microphone NS, and the error microphones E 1, E 2 (wired or wireless depending on a specific configuration)) signals. The audio integrated circuits 20A, 20B also interface with other integrated circuits, such as the RF integrated circuit 12, which contains the radio telephone transceiver shown in FIG. 1A. In other arrangements, the circuitry and techniques disclosed herein may be implemented in a single integrated circuit containing control circuitry and other functionality for implementing the entirety of a personal audio device, such as an MP3 player on chip integrated circuit For example. Alternatively, a plurality of integrated circuits may be used, for example, when a wireless connection is provided to the wireless telephone 10 from each of the earphones EB 1, EB 2, and / or some or all of the ANC processing is performed by the earphones EB 1, EB 2, 10 when it is performed in a module arranged along a cable connecting the earphones EB 1, EB 2.

[0013] In general, the ANC technique illustrated herein measures the ambient sound events (in contrast to the transducers SPKLH, SPKLL, SPKRH, and SPKRL and / or near-end utterances) that impinge on the reference microphones R1, R2 , And also measure the same ambient sound event that collides with the error microphones E1, E2. The ANC processing circuits of the integrated circuits 20A, 20B individually adapt the anti-noise signals generated from the outputs of the corresponding reference microphones R1, R2, and in the corresponding error microphones E1, E2, the amplitude of the ambient sound event is minimized As shown in FIG. Since the acoustic path PL (z) extends from the reference microphone R 1 to the error microphone E 1, the ANC circuit in the audio integrated circuit 20 A essentially consists of the response of the audio output circuit of the audio integrated circuit 20 A and the conversion And the acoustic / electric transfer function of the speakers SPKLH and SPKLL, the acoustic path PL (z) combined with the effects of the electroacoustic paths SLH (z) and SLL (z) eliminated. The presumed responses are the transducers SPKLH and SPKLL and the error microphones E 1 (E 1) within a specific acoustic environment, which are influenced by the proximity and structure of the ear 5 A and other physical objects and human head structures that may be in close proximity to the earphone EB 1 And the like. Likewise, the audio integrated circuit 20B includes the electroacoustic paths SRH (z) and SRL (z) representing the response of the audio output circuit of the audio integrated circuit 20B and the acoustic / electrical transfer functions of the transducers SPKRH and SPKRL, Estimate the combined acoustic path PR (z) with the influence removed.

[0014] Referring now to FIG. 2, the earphone EB 1, EB 2 and the circuitry within the radiotelephone 10 are shown in a block diagram. The circuit shown in FIG. 2 also applies to the other configurations described above, but the signal transmission between the CODEC integrated circuit 20 in the radiotelephone 10 and the other units is such that the audio integrated circuits 20 A, When located in the outside of the radiotelephone 10, for example in the corresponding earpiece EB 1, EB 2, it is provided by cable or wireless connection. In such a configuration, a single integrated circuit 20 implementing the integrated circuits 20 A - 20 B, signaling between the error microphones E 1, E 2, the reference microphones R 1, R 2, and the converters SPKLH, SPKLL, SPKRH, and SPKRL Is provided by a wired or wireless connection when the audio integrated circuit 20 is located within the radiotelephone 10. In the illustrated embodiment, the audio integrated circuits 20A, 20B are shown as separate and substantially the same circuit, so only the audio integrated circuit 20A is described in detail below.

[0015] The audio integrated circuit 20A includes an analog / digital converter (ADC) 21A which receives the reference microphone signal from the reference microphone R1 and generates a digital representation ref of the reference microphone signal. The audio integrated circuit 20 A also includes an ADC 21 B for receiving an error microphone signal from the error microphone E 1 and for generating a digital representation err of the error microphone signal and a near utterance microphone signal from a near speech microphone NS and a near utterance microphone signal And an ADC 21C for generating the digital representation ns of (The audio integrated circuit 20B receives from the audio integrated circuit 20A the digital representation ns of the near

speech microphone signal via a wireless or wired connection as described above.) The audio integrated circuit 20A converts the output of the combiner 26A to Generates an output for driving the converter SPKLH from the amplifier A1 which amplifies the output of the digital / analog converter (DAC) 23A to be received. The combiner 26 C combines the left channel internal audio signal ual with the source audio ds, which is received from the radio frequency (RF) integrated circuit 22. The combiner 26A combines the source audio dsh + ialh, which is the high frequency band component of the output of the combiner 26C with the high frequency band anti-noisel signal generated by the left channel ANC circuit 30, Has the same polarity as the noise in the microphone signal ref and is therefore subtracted by combiner 26A. The combiner 26A is also responsive to the attenuated high frequency part of the near-speech signal ns, ie, the side-tone information, so that the user of the radiotelephone 10 can appropriately associate with the downlink utterance ds and hear its own voice Combine sth. The near-speech signal ns is also provided to the RF integrated circuit 22 and transmitted to the service provider as an uplink speech via the antenna ANT. Likewise, the left channel audio integrated circuit 20 A generates an output from the amplifier A 2 for driving the converter SPKLL, which outputs the output of the digital / analog converter (DAC) 23 B which receives the output of the combiner 26 B To amplify. The combiner 26B combines the source audio dsl + iall, which is the low frequency band component of the output of the combiner 26C with the low frequency band anti-noisell signal generated by the ANC circuit 30 and is typically a reference microphone signal has the same polarity as the noise in ref and is therefore subtracted by combiner 26B. The combiner 26B also combines the attenuated part of the near-speech signal ns, ie, the side-tone low-frequency information stl.

[0016] Referring now to FIG. 3, a detailed embodiment within the ANC circuit 30, as may be used to implement the audio integrated circuit 20B of FIG. 2, is shown. The same circuit is used to implement the audio integrated circuit 20A and involves changes to the interior of the channel indicator diagram as noted below. High frequency channel 50 A and low frequency channel 50 B, respectively, are provided to generate anti-noise signals anti-noiserh and anti-noiserl, respectively. In the following description where the signal and response indicator contains the letter "r" indicating the right channel, the letters are shown in FIG. 3 as implemented within the audio integrated circuit 20 A of FIG. 2, It will be replaced with "l" indicating the left channel. If the signal and the response are labeled with the letter "h" for the low frequency in the high frequency channel 50A, the corresponding element in the low frequency channel 50B is the signal and response labeled with the letter "l" It will be substituted. The adaptive filter 32A receives the reference microphone signal ref and adapts it so that its transfer function Wrh (z) becomes Pr (z) / Srh (z) under an ideal situation and generates an anti-noise signal anti-noiserh . The coefficients of the adaptive filter 32A are generally determined in a least squares sense meaning by using the correlation of the two signals to minimize errors between those components of the reference microphone signal ref that are present in the microphone signal err, And is controlled by the W coefficient control block 31 ⟨/ b⟩ A that determines the response of the adaptive filter 32 ⟨/ b⟩ A. Although the embodiments disclosed herein use an adaptive filter 32 A connected in a feedforward configuration, the techniques disclosed herein are implemented in a noise cancellation system with fixed or programmable filters If the coefficients of the adaptive filter 32A are not preset, selected, or otherwise adapted continuously, and alternatively or in combination with a fixed filter topology, the disclosed disclosure herein The technique to be applied can be applied to a feedback ANC system or a hybrid feedback / feed forward ANC system. The signal provided as input to the W coefficient control block 31A is provided from the copy of the estimate of the response of the path Srh (z) provided by the filter 34B and the output of the combiner 36C including the error microphone signal err And the reference microphone signal ref as shaped by another signal. By converting the reference microphone signal ref with the response SErhCOPY (z), which is a copy of the estimate of the response of the path Srh (z), and minimizing the portion of the error signal correlated with the components of the reference microphone signal ref, The adaptive filter 32A adapts to the desired response of Pr (z) / Srh (z).

[0017] In addition to the error microphone signal err, the secondary path (response SErhCOPY (z) is its copy) with the response SErh (z) as the other signal processed by the W coefficient control block 31A with the output of the filter 34B Includes the inverse quantity of source audio (ds + iar), including downlink audio signal ds and internal audio ian, processed by filter 34A. First, the source audio (ds + iar) is filtered before being provided to the high frequency channel 50A by the high pass filter 35A, which passes only the frequencies provided by the high frequency converter SPKLH or SPKRH. Similarly, the source audio (ds + iar) provided to the low frequency channel 50B is first filtered by a low pass filter 35B, which passes only the frequencies caused by the low frequency converter SPKLL or SPKRL. Accordingly, the high-pass filter 35 A and the low-pass filter 35 B form a crossover for the source audio (ds + iar) such that only the appropriate frequencies are passed through the high frequency channel 50 A and the low frequency channel 50 B, respectively Of the converters SPKLH, SPKLL, or SPKRH, SPKRL. By injecting the inverse quantity of the source audio (ds + iar) filtered by the response SErh (z), the adaptive filter 32 A is prevented from adapting to the relatively large amount of source audio present in the error microphone signal err . By transforming the reciprocal copy of the source audio (ds +

iar) with the estimate of the response of the path Srh (z), the source audio that is removed from the error microphone signal err before processing is the source audio reproduced in the error microphone signal err (Ds + iar), which is the expected version. The amount of source audio is matched since the electrical and acoustic paths of Srh (z) are the paths followed by the source audio (ds + iar) to reach the error microphone E. Filter 34B is not itself an adaptive filter but is tuned to match the response of secondary path adaptive filter 34A such that the response of filter 34B tracks the adaptation of secondary path adaptive filter 34A , Has an adjustable response. To implement the foregoing, the secondary path adaptive filter 34A has coefficients controlled by the SE coefficient control block 33A. Secondary path adaptive filter 34 A processes the low or high frequency source audio (ds + iar) and provides a signal representative of the expected source audio to be delivered to error microphone E. When subtracted from the error microphone signal err, the secondary path adaptive filter 34 A converts the signal forming the error signal e containing the content of the error microphone signal err not attributable to the source audio (ds + iar) to the source audio ds + iar). The combiner 36C removes the filtered source audio (ds + iar) from the error microphone signal err and generates the aforementioned error signal e.

[0018] Each of the high-frequency channel 50A and the low-frequency channel 50B can operate independently so as to generate respective anti-noise signals anti-noiseh and anti-noisel. However, since the error signal e and the reference microphone signal ref can contain frequencies of any frequency within the audio band without band limiting the anti-noise signals anti-noiseh and anti-noisel, their respective high frequencies And components that should not be transmitted to the low frequency converters SPKRH / SPKLH and SPKRL / SPKLL. Therefore, the noise injection technique is used to control the response Wrh (z) of the adaptive filter 32A. A noise source 37 generates an output noise signal nh (z), which is supplied to a copy WrhCOPY (z) of the response Wrh (z) of the adaptive filter 32A provided by the adaptive filter 32B. The combiner 36A adds the noise signal nh (z) to the output adaptive filter 34B, which is provided to the W coefficient control 31A. The noise signal nh (z) as shaped by the filter 32B is added by the combiner 36B to the correlator input to the W coefficient controller 31A such that the noise signal nh (z) is asymmetrically added to the correlation input to the W coefficient control 31A Is subtracted from the output so that the response Wrh (z) of the adaptive filter 32A is biased by the fully correlated input of the noise signal nh (z) to each correlation input to the W coefficient control 31A. The input noise appears in the reference input to the direct W coefficient control 31A and does not appear in the error microphone signal err and is output by the combiner 36B via the combination of noise filtered at the output of the filter 32B to the W coefficient control 31A , The W coefficient control 31 A will adapt W rh (z) to attenuate the frequencies present in n h (z). The content of the noise signal nh (z) does not appear in the anti-noise signal but only in the response Wrh (z) of the adaptive filter 32A, where the noise signal nh (z) will have amplitude reduction at the energy / appear.

[0019] In order to prevent low frequencies from being generated in the anti-noise signal anti-noiseh, the noise source 37 generates noise with a spectrum with energy in the low frequency band, which is a function of W coefficient control 31 A Will attempt to eliminate the obvious source of ambient sound due to the input noise signal nh (z) and will reduce the gain of the adaptive filter 32A within these low frequency bands. For example, a white noise source may be filtered with a response similar to the response of the low pass filter 35B for use as the noise source 37 in the high frequency channel 50A, which is applied to the adaptive filter 32A by a low pass filter 35B In the region of the passband of the low-pass filter. By filtering the white noise source using the same thing for the low frequency channel 50B, that is to say with a response that matches the response of the high pass filter 35A, the crossover is obtained for each anti-noise signal anti-noise h and anti Is effectively formed by the adaptation of adaptive filter 32A in high frequency channel 50A and low frequency channel 50B, which prevents undesirable frequencies in -noisel. A similar construction may be formed around the secondary path adaptive filter 34A, but the input to the secondary path adaptive filter 34A is already removed by the respective one of the filters 35A and 35B to remove from the band energy As it is being filtered, such noise input should not be required to remove unwanted frequencies from the output of the secondary path adaptive filter 34A. One advantage of using noise input rather than additional filtering to remove unwanted crossover energy from anti-noise signals anti-noiseh and anti-noisel is that additional latency is provided to noise source 37 It is not introduced, except for arbitrary latency due to changes in the originating response.

[0020] Referring now to FIG. 4, there is shown a block diagram of an ANC system for implementing an ANC technique as depicted in FIG. 3 and having a processing circuit 40 as may be implemented in the audio integrated circuits 20 A, 20 B of FIG. 2 Although the figures are shown and illustrated as being combined in one circuit, it can also be implemented as two or more processing circuits in communication. The processing circuit 40 includes a processor core 42 coupled to a memory 44 storing program instructions, including a computer program product, which may implement some or all of the ANC techniques described above as well as other signal processing. Optionally, dedicated digital signal processing (DSP) logic 46 may be provided for implementing all or part of the ANC signal processing provided by the processing circuit 40. Processing circuit

40 also includes ADCs 21 A - 21 E, respectively, for receiving input from reference microphone R 1, error microphone E 1, near speech microphone NS, reference microphone R 2, and error microphone E 2. In an alternative embodiment, where one or more of the reference microphone R1, the error microphone E1, the near speech microphone NS, the reference microphone R2, and the error microphone E2 have a digital output or are communicated as a digital signal from a remote ADC, A corresponding one of the ADCs 21 A - 21 E is omitted and the digital microphone signal is directly interfaced to the processing circuit 40. DAC 23 A and amplifier A 1 are also provided by the processing circuit 40 to provide the converter output signal to the converter SPKLH, including anti-noise as described above. Similarly, DACs 23B-23D and amplifiers A2-A4 provide other converter output signals to converter pairs SPKLH, SPKLL, SPKRH, and SPKRL. The converter output signal may be a digital output signal for providing to a module acoustically reproducing the digital output signal.

[0021] While the invention has been particularly shown and described with reference to preferred embodiments thereof, it is to be understood that the foregoing and other variations in form and detail may be made therein without departing from the spirit and scope of the invention May be performed by a person skilled in the art.

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パーソナルオーディオデバイスのための短待ち時間マルチドライバ適応雑音消去（ＡＮＣ）システム

Data originating from sources other than the EPO may not be accurate, complete, or up to date.

The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

A personal audio system, said personal audio system comprising: a source of audio for reproduction, said source of audio providing a source audio signal; a first transducer , Said first transducer comprising a first anti-noise signal for canceling the influence of the source audio signal for reproduction to the listener and the ambient audio sound in the acoustic output of the first transducer, A first transducer for reconstructing the high frequency content of the source audio signal and a second transducer for converting the source audio signal for reproduction to the listener and the second transducer for reproducing the second high frequency content, A second transducer for reconstructing low frequency content with a second anti-noise signal for canceling the influence of ambient audio sounds in the acoustic output of the transducer of the surroundings; At least one microphone for providing at least one microphone signal indicative of a sound; and at least one microphone for providing at least one microphone signal indicative of sound, and means for generating a first anti-noise signal from said at least one microphone signal in response to said at least one microphone signal using a first filter To reduce the presence of the ambient audio sound in the first transducer and the second transducer, wherein the processing circuit uses a second filter to reduce the presence of the ambient audio tone in the first transducer and the second transducer Generating the second anti-noise signal from the at least one microphone signal in response to the at least one microphone signal to determine the presence of the ambient audio sound in the first transducer and the second transducer So that the first filter reduces the presence of the ambient audio sound A first adaptive filter having an adaptive first frequency response and said second filter is a second adaptive filter having a second frequency response adapted to reduce the presence of said ambient audio sound And wherein said processing circuit is operable to limit content of said first anti-noise signal to said first predetermined frequency range by limiting said first frequency response of said first adaptive filter to a first predetermined frequency range Limiting the second frequency response of the second adaptive filter to a second predetermined frequency range by limiting the second frequency response of the second adaptive filter to a second predetermined frequency range by limiting the content of the second anti- 2, wherein the first predetermined frequency range and the second predetermined frequency range are substantially different, whereby the first adaptive filter and the second Wherein the adaptive filter operates as a crossover for separating the at least one microphone signal into a plurality of frequency bands.

Further comprising an error microphone for providing an error microphone signal indicative of ambient audio and acoustic output of the first transducer and ambient audio and acoustic output of the second transducer, wherein the first adaptive filter Comprises a first coefficient generator adapted to minimize the components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit comprises: a first coefficient generator Limiting the adaptation of the first frequency response by modifying the frequency content of the first signal and the second adaptive filter minimizing the components of the reference microphone signal present in the error microphone signal A second coefficient generator adapted to adapt the first coefficient generator to generate a second coefficient signal that is adapted to be applied to the second coefficient generator; The personal audio system according to claim 1, wherein the adaptation of the second frequency response is limited by modifying the content.

The processing circuit places a first additional signal having a first predetermined frequency content within the second predetermined frequency range into the first signal input to the first coefficient generator Thereby modifying the frequency content of said first signal input to said first coefficient generator and said processing circuit having a second predetermined frequency content within said first predetermined frequency range Modifying the frequency content of the second signal input to the second coefficient generator by inputting a second additional signal to the second signal input to the second coefficient generator 3. The personal audio system according to claim 2,

4. The personal audio system according to claim 3, wherein the first additional signal and the second additional signal are noise signals.

Wherein the processing circuit provides a crossover that receives the source audio signal and filters the source audio signal to generate a higher frequency content source audio signal and a lower frequency content source

audio signal, The personal audio system according to claim 1, wherein the processing circuit combines the higher frequency content source audio signal with the first anti-noise signal and combines the lower frequency content source audio signal with the second anti-noise signal system.

The personal audio system according to claim 1, wherein the first converter is a high frequency converter of an ear speaker and the second converter is a low frequency converter of the ear speaker.

A third transducer including a second source audio signal and a third anti-noise for canceling the influence of ambient audio tones in the acoustic output of the third transducer, A third transducer for reconstructing the high frequency content with the signal, a fourth transducer for converting the second source audio signal and the fourth transducer And a fourth transducer for reproducing low frequency content with a fourth anti-noise signal for canceling the influence of ambient audio sound in the acoustic output, wherein the processing circuit uses a third filter By generating said third anti-noise signal and said fourth anti-noise signal from said at least one microphone signal in response to said at least one microphone signal, said third transducer and said fourth conversion Reduces the presence of the ambient audio sound in the at least one microphone signal and the processing circuitry generates a fourth anti-noise signal from the at least one microphone signal in response to the at least one microphone signal using a fourth filter To reduce the presence of the ambient audio sound in the third transducer and the fourth transducer.

A method for canceling the influence of ambient audio sound by a personal audio system, the method comprising: generating at least one microphone signal by measuring ambient audio sound using at least one microphone; Reducing the presence of the ambient audio sound in the first transducer by generating a first anti-noise signal from the at least one microphone signal in response to the at least one microphone signal using a filter of one And generating a second anti-noise signal from said at least one microphone signal in response to said at least one microphone signal using a second filter to produce said ambient audio sound in a second transducer To reduce the existence of Providing a source of audio, the source of audio providing a source audio signal; and using the first converter to convert the source audio signal and the first anti-noise signal Reproducing the high frequency content of the source audio signal and the second anti-noise signal using the first converter, and reproducing the low frequency content of the source audio signal and the second anti-noise signal using the second converter, Filter is a first adaptive filter having a first frequency response that is adapted to reduce the presence of said ambient audio sound and said second filter is adapted to reduce the presence of said ambient audio sound Wherein the first adaptive filter is a second adaptive filter having an adaptive second frequency response, wherein generating the first anti-noise signal comprises comparing the first circumference of the first adaptive filter Comprising limiting the content of the first anti-noise signal to the first predetermined frequency range by limiting the number response to a first predetermined frequency range and generating the second anti-noise signal Wherein limiting the second frequency response of the second adaptive filter to a second predetermined frequency range causes the content of the second anti-noise signal to fall within the second predetermined frequency range Wherein the first predefined frequency range and the second predetermined frequency range are substantially different such that the first adaptive filter and the second adaptive filter are configured to generate at least Operating as a crossover to separate one microphone signal into multiple frequency bands.

Generating an error microphone signal by using an error microphone to measure the ambient audio and acoustic output of the first transducer and the ambient audio and acoustic output of the second transducer Wherein generating the first anti-noise signal comprises generating a first coefficient control to control the first frequency response to minimize a component of a reference microphone signal present in the error microphone signal, Wherein generating the second anti-noise signal comprises adjusting the coefficients of the second frequency response to minimize the component of the reference microphone signal present in the error microphone signal, Wherein the generating of the first anti-noise signal comprises adapting coefficients of a second coefficient generator that controls the first anti-noise signal, Limiting the adaptation of the first frequency response by modifying the frequency content of the first signal input to the number generator, and generating the second anti-noise signal comprises modifying the second coefficient 9. The method according to claim 8, wherein the adaptation of the second frequency response is limited by modifying the frequency content of the second signal input to the generator.

Generating the first anti-noise signal comprises generating a first additional signal having a first predetermined frequency content within the second predetermined frequency range at least one to the first coefficient generator Limiting the adaptation of the first frequency response by injecting one of the first signal inputs and generating the second anti-noise signal comprises applying a second one of the second frequency signals within the first predetermined frequency range, Limiting adaptation of the second frequency response by applying a second

additional signal having predetermined frequency content to at least one second signal input to the second coefficient generator 9.

11. The method of claim 10, wherein the first additional signal and the second additional signal are noise signals.

Implementing a crossover that receives the source audio signal and filters the source audio signal to generate a higher frequency content source audio signal and a lower frequency content source audio signal; 9. The method of claim 8, further comprising: combining a frequency content source audio signal with the first anti-noise signal; and combining the lower frequency content source audio signal with the second anti-noise signal.

9. The method of claim 8, wherein the first transducer is a high frequency transducer of an ear speaker and the second transducer is a low frequency transducer of the ear speaker.

The third converter is used to reproduce the high frequency content of the second source audio signal and the third anti-noise signal in order to counteract the influence of ambient audio sound in the sound output of the third converter And using the fourth converter to cancel the influence of the ambient audio sound in the acoustic output of the fourth transducer, the fourth transducer converts the second source audio signal to the fourth anti-noise signal Reproducing the low frequency content; and generating the third anti-noise signal and the fourth anti-noise signal from the at least one microphone signal in response to the at least one microphone signal using a third filter Reducing the presence of the ambient audio sound in the third transducer and the fourth transducer by using the first filter to reduce the presence of the ambient audio tone in the third transducer and the fourth transducer, Generating the fourth anti-noise signal from the at least one microphone signal in response to at least one microphone signal so as to determine the presence of the ambient audio sound in the third converter and the fourth converter And decreasing the amount of oxygen in the sample.

An integrated circuit for implementing at least a portion of a personal audio system, the integrated circuit comprising: a source of audio for reproduction, the source of the audio providing a source audio signal; a source of audio And a first output for providing a first output signal to a first transducer, the first transducer comprising: a source audio signal for reproduction to a listener; A first output for reproducing high frequency content with a first anti-noise signal for canceling the influence of ambient audio sounds in the acoustic output of one transducer, a first output for converting the second output signal to a second transducer A second output for providing a source audio signal for playback to the listener and a second output for providing to the surrounding audio in the acoustic output of the second transducer, At least one microphone input for providing at least one microphone signal indicative of said ambient audio sound and a second output for reproducing low frequency content with a second anti-noise signal for counteracting sound effects, , Generating said first anti-noise signal from said at least one microphone signal in response to said at least one microphone signal using a first filter, said first transducer and said second Processing circuitry for reducing the presence of said ambient audio sound in a transducer wherein said processing circuitry is adapted to process said at least one microphone signal in response to said at least one microphone signal using a second filter, By generating the anti-noise signal of 2, the first transducer and the second transformation Wherein the first filter is a first adaptive filter having a first frequency response adapted to reduce the presence of the ambient audio sound and the second filter is a first adaptive filter adapted to reduce the presence of the ambient audio sound, Filter is a second adaptive filter having a second frequency response that is adapted to reduce the presence of the ambient audio sound and the processing circuit is adapted to determine the first frequency response of the first adaptive filter By limiting the content of the first anti-noise signal to the first predetermined frequency range by limiting the content of the first anti-noise signal to a first predetermined frequency range, Limiting the content of said second anti-noise signal to said second predetermined frequency range by limiting the frequency response of said first anti-noise signal to a second predetermined frequency range, And the second predetermined frequency range are substantially different so that the first adaptive filter and the second adaptive filter separate the at least one microphone signal into a plurality of frequency bands And a processing circuit that operates as a crossover for performing the processing.

Further comprising an error microphone for providing an error microphone signal indicative of ambient audio and acoustic output of the first transducer and ambient audio and acoustic output of the second transducer, wherein the first adaptive filter Comprises a first coefficient generator adapted to minimize the components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit comprises: a first coefficient generator Limiting the adaptation of the first frequency response by modifying the frequency content of the first signal and the second adaptive filter minimizing the components of the reference microphone signal present in the error microphone signal A second coefficient generator adapted to adapt the

first coefficient generator to generate a second coefficient signal that is adapted to be applied to the second coefficient generator; 16. The integrated circuit of claim 15, wherein adapting the second frequency response is limited by modifying the content.

The processing circuit places a first additional signal having a first predetermined frequency content within the second predetermined frequency range into the first signal input to the first coefficient generator Thereby modifying the frequency content of said first signal input to said first coefficient generator and said processing circuit having a second predetermined frequency content within said first predetermined frequency range The frequency content of the second signal input to the second coefficient generator is modified by inputting the second additional signal to the second signal input to the second coefficient generator , The integrated circuit according to claim 16.

18. The integrated circuit of claim 17, wherein the first additional signal and the second additional signal are noise signals.

Wherein the processing circuit provides a crossover that receives the source audio signal and filters the source audio signal to generate a higher frequency content source audio signal and a lower frequency content source audio signal, 16. The integrated circuit of claim 15, wherein the processing circuit combines the higher frequency content source audio signal with the first anti-noise signal and combines the lower frequency content source audio signal with the second anti-noise signal .

Wherein said first output signal is adapted to be provided to a high frequency converter of an ear speaker and said second output signal is adapted to be provided to a low frequency converter of said ear speaker. An integrated circuit according to item 15.

A third output for providing a third output signal to a third transducer, said third transducer comprising a second source audio signal and a second output audio signal within a sound output of said third transducer And a third anti-noise signal for canceling the influence of the ambient audio sound of the second output signal; a third output for reproducing the high frequency content with the fourth anti- Output, said fourth transducer being configured to generate a second low-pass filtering of the second source audio signal and a fourth anti-noise signal for canceling the influence of the ambient audio sound in the acoustic output of the fourth transducer And a fourth output for reconstructing the frequency content, wherein the processing circuit further comprises means for generating a third anti-noise from the at least one microphone signal in response to the at least one microphone signal using a third filter Signal and said To reduce the presence of the ambient audio sound in the third transducer and the fourth transducer by generating an anti-noise signal of the first transducer and the second transducer, and wherein the processing circuit uses the fourth filter Reducing the presence of the ambient audio sound in the third converter and the fourth converter by generating the fourth anti-noise signal from the at least one microphone signal in response to one microphone signal The integrated circuit according to claim 20.

A personal audio system comprising: a plurality of output transducers; at least one microphone for providing at least one microphone signal indicative of ambient audio sound; a processing circuit for implementing adaptive noise cancellation; Wherein the plurality of adaptive filters generate a plurality of anti-noise signals for a corresponding one of the plurality of output converters, and the plurality of adaptive filters are configured to generate a plurality of anti- Which operates as a crossover for separating the at least one microphone signal into the plurality of frequency bands corresponding to the plurality of output converters by generating the plurality of anti- Audio system.

A method of canceling the influence of ambient audio sound by a personal audio system, the method comprising: generating at least one microphone signal by measuring ambient audio sound using at least one microphone; And generating a plurality of anti-noise signals for providing to a corresponding one of the plurality of output transducers using a corresponding one of the plurality of adaptive filters, The adaptive filter generates the plurality of anti-noise signals in corresponding frequency bands among the plurality of frequency bands to generate the at least one microphone signal in the plurality of frequency bands corresponding to the plurality of output converters To act as a crossover to separate into.

An integrated circuit for mounting at least a portion of a personal audio system, the integrated circuit comprising: a plurality of outputs for providing a plurality of output signals to a corresponding one of the plurality of output transducers; , At least one microphone input for receiving at least one microphone signal indicative of an ambient audio sound and processing circuitry for implementing adaptive noise cancellation wherein the plurality of adaptive filters are adapted to generate a corresponding one of the plurality of outputs

Generating a plurality of anti-noise signals at an output, said plurality of adaptive filters generating said plurality of anti-noise signals within corresponding frequency bands of a plurality of frequency bands, thereby generating said at least one microphone signal And operates as a crossover for separating into the plurality of frequency bands corresponding to the plurality of output converters, Integrated circuit.

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最終頁に続く

(54)【発明の名称】パーソナルオーディオデバイスのための短待ち時間マルチドライバ適応雑音消去（ANC）システム

(57)【特許請求の範囲】
【請求項1】
パーソナルオーディオシステムであって、前記パーソナルオーディオシステムは、
再現のためのオーディオのソースであって、前記オーディオのソースは、ソースオーディオ信号を提供する、オーディオのソースと、
第1の変換器であって、前記第1の変換器は、聴取者への再生のための前記ソースオーディオ信号と、前記第1の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第1の反雑音信号との高周波数コンテンツを再現する、第1の変換器と、
第2の変換器であって、前記第2の変換器は、前記聴取者への再生のための前記ソースオーディオ信号と、前記第2の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第2の反雑音信号との低周波数コンテンツを再現する、第2の変換器と、
前記周囲オーディオ音を示す少なくとも1つのマイクロホン信号を提供するための少なくとも1つのマイクロホンと、
第1のフィルタを使用して前記少なくとも1つのマイクロホン信号に応じて前記少なくとも1つのマイクロホン信号から前記第1の反雑音信号を発生させることにより、前記第1の変換器および前記第2の変換器における前記周囲オーディオ音の存在を低減させる処理回路であって、前記処理回路は、第2のフィルタを使用して前記少なくとも1つのマイクロホン信号に応じて前記少なくとも1つのマイクロホン信号から前記第2の反雑音信号を発生させることにより、前記第1の変換器および第2の変換器における前記周囲オーディオ音の存在を低減させ、前記第1のフィルタは、前記周囲オーディオ音の存在を低減さ

せるように適応する第１の周波数応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の周波数応答を有する第２の適応フィルタであり、前記処理回路は、前記第１の適応フィルタの前記第１の周波数応答を第１の所定の周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の所定の周波数範囲に制限し、前記処理回路は、前記第２の適応フィルタの前記第２の周波数応答を第２の所定の周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の所定の周波数範囲に制限し、前記第１の所定の周波数範囲および前記第２の所定の周波数範囲は、実質的に異なり、これにより、前記第１の適応フィルタおよび前記第２の適応フィルタは、前記少なくとも１つのマイクロホン信号を複数の周波数帯に分離するためのクロスオーバとして動作する、処理回路と
を備える、パーソナルオーディオシステム。

【請求項２】

前記第１の変換器の周囲オーディオ音および音響出力と前記第２の変換器の周囲オーディオ音および音響出力とを示すエラーマイクロホン信号を提供するためのエラーマイクロホンをさらに備え、前記第１の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第１の係数発生器を有し、前記処理回路は、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第２の係数発生器を有し、前記処理回路は、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第２の周波数応答の適応を制限する、請求項１に記載のパーソナルオーディオシステム。

【請求項３】

前記処理回路は、前記第２の所定の周波数範囲内の第１の所定の周波数コンテンツを有する第１の付加的信号を前記第１の係数発生器に入力された前記第１の信号に投入することによって、前記第１の係数発生器に入力された前記第１の信号の周波数コンテンツを改変し、前記処理回路は、前記第１の所定の周波数範囲内の第２の所定の周波数コンテンツを有する第２の付加的信号を前記第２の係数発生器に入力された前記第２の信号に投入することによって、前記第２の係数発生器に入力された前記第２の信号の周波数コンテンツを改変する、請求項２に記載のパーソナルオーディオシステム。

【請求項４】

前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、請求項３に記載のパーソナルオーディオシステム。

【請求項５】

前記処理回路は、前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理することにより、より高い周波数コンテンツソースオーディオ信号とより低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを提供し、前記処理回路は、前記より高い周波数コンテンツソースオーディオ信号を前記第１の反雑音信号と組み合わせ、前記より低い周波数コンテンツソースオーディオ信号を前記第２の反雑音信号と組み合わせる、請求項１に記載のパーソナルオーディオシステム。

【請求項６】

前記第１の変換器は、イヤースピーカの高周波数変換器であり、前記第２の変換器は、前記イヤースピーカの低周波数変換器である、請求項１に記載のパーソナルオーディオシステム。

【請求項７】

第３の変換器であって、前記第３の変換器は、第２のソースオーディオ信号と、前記第３の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第３の反雑音信号との高周波数コンテンツを再現する、第３の変換器と、

第４の変換器であって、前記第４の変換器は、前記第２のソースオーディオ信号と、前記第４の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第４の反雑音信

号との低周波数コンテンツを再現する、第４の変換器と
　をさらに備え、
　前記処理回路は、第３のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第３の反雑音信号および前記第４の反雑音信号を発生させることにより、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させ、前記処理回路は、第４のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第４の反雑音信号を発生させることにより、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させる、請求項６に記載のパーソナルオーディオシステム。
【請求項８】
　パーソナルオーディオシステムによる周囲オーディオ音の影響を打ち消す方法であって、前記方法は、
　少なくとも１つのマイクロホンを用いて周囲オーディオ音を測定することにより、少なくとも１つのマイクロホン信号を生成することと、
　第１のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から第１の反雑音信号を発生させることにより、第１の変換器における前記周囲オーディオ音の存在を低減させることと、
　第２のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から第２の反雑音信号を発生させることにより、第２の変換器における前記周囲オーディオ音の存在を低減させることと、
　再現のためのオーディオのソースを提供することであって、前記オーディオのソースは、ソースオーディオ信号を提供する、ことと、
　前記第１の変換器を用いて、前記ソースオーディオ信号と前記第１の反雑音信号との高周波数コンテンツを再現することと、
　前記第２の変換器を用いて、前記ソースオーディオ信号と前記第２の反雑音信号との低周波数コンテンツを再現することと
　を含み、
　前記第１のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第１の周波数応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の周波数応答を有する第２の適応フィルタであり、
　前記第１の反雑音信号を発生させることは、前記第１の適応フィルタの前記第１の周波数応答を第１の所定の周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の所定の周波数範囲に制限することを含み、前記第２の反雑音信号を発生させることは、前記第２の適応フィルタの前記第２の周波数応答を第２の所定の周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の所定の周波数範囲に制限することを含み、前記第１の所定の周波数範囲および前記第２の所定の周波数範囲は、実質的に異なり、これにより、前記第１の適応フィルタおよび前記第２の適応フィルタは、前記少なくとも１つのマイクロホン信号を複数の周波数帯に分離するためのクロスオーバとして動作する、方法。
【請求項９】
　エラーマイクロホンを用いて、前記第１の変換器の周囲オーディオ音および音響出力と前記第２の変換器の周囲オーディオ音および音響出力とを測定することにより、エラーマイクロホン信号を発生させることをさらに含み、前記第１の反雑音信号を発生させることは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするために、前記第１の周波数応答を制御する第１の係数発生器の係数を適応させることを含み、前記第２の反雑音信号を発生させることは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするために、前記第２の周波数応答を制御する第２の係数発生器の係数を適応させることを含み、前記第１の反雑音信号を発生させる

ことは、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の反雑音信号を発生させることは、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第２の周波数応答の適応を制限する、請求項８に記載の方法。

【請求項１０】

前記第１の反雑音信号を発生させることは、前記第２の所定の周波数範囲内の第１の所定の周波数コンテンツを有する第１の付加的信号を前記第１の係数発生器への少なくとも１つの第１の信号入力に投入することによって、前記第１の周波数応答の適応を制限し、前記第２の反雑音信号を発生させることは、前記第１の所定の周波数範囲内の第２の所定の周波数コンテンツを有する第２の付加的信号を前記第２の係数発生器への少なくとも１つの第２の信号入力に投入することによって、前記第２の周波数応答の適応を制限する、請求項９に記載の方法。

【請求項１１】

前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、請求項１０に記載の方法。

【請求項１２】

前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理することにより、より高い周波数コンテンツソースオーディオ信号とより低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを実装することと、

前記より高い周波数コンテンツソースオーディオ信号を前記第１の反雑音信号と組み合わせることと、

前記より低い周波数コンテンツソースオーディオ信号を前記第２の反雑音信号と組み合わせることと

をさらに含む、請求項８に記載の方法。

【請求項１３】

前記第１の変換器は、イヤースピーカの高周波数変換器であり、前記第２の変換器は、前記イヤースピーカの低周波数変換器である、請求項８に記載の方法。

【請求項１４】

第３の変換器の音響出力内の周囲オーディオ音の影響を打ち消すために、前記第３の変換器を用いて、第２のソースオーディオ信号と第３の反雑音信号との高周波数コンテンツを再現することと、

第４の変換器の音響出力内の周囲オーディオ音の影響を打ち消すために、前記第４の変換器を用いて、前記第２のソースオーディオ信号と第４の反雑音信号との低周波数コンテンツを再現することと、

第３のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第３の反雑音信号および前記第４の反雑音信号を発生させることにより、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させることと、

第４のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第４の反雑音信号を発生させることにより、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させることと

をさらに含む、請求項１３に記載の方法。

【請求項１５】

パーソナルオーディオシステムの少なくとも一部を実装するための集積回路であって、前記集積回路は、

再現のためのオーディオのソースであって、前記オーディオのソースは、ソースオーディオ信号を提供する、オーディオのソースと、

第１の出力信号を第１の変換器に提供するための第１の出力であって、前記第１の変換器は、聴取者への再生のための前記ソースオーディオ信号と、前記第１の変換器の音響出

力内の周囲オーディオ音の影響を打ち消すための第１の反雑音信号との高周波数コンテンツを再現する、第１の出力と、
第２の出力信号を第２の変換器に提供するための第２の出力であって、前記第２の変換器は、前記聴取者への再生のための前記ソースオーディオ信号と、前記第２の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第２の反雑音信号との低周波数コンテンツを再現する、第２の出力と、
前記周囲オーディオ音を示す少なくとも１つのマイクロホン信号を提供するための少なくとも１つのマイクロホン入力と、
第１のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第１の反雑音信号を発生させることにより、前記第１の変換器および前記第２の変換器における前記周囲オーディオ音の存在を低減させる処理回路であって、前記処理回路は、第２のフィルタを使用して前記少なくとも１つのマイクロホン信号に応じて前記少なくとも１つのマイクロホン信号から前記第２の反雑音信号を発生させることにより、前記第１の変換器および前記第２の変換器における前記周囲オーディオ音の存在を低減させ、前記第１のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第１の周波数応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の周波数応答を有する第２の適応フィルタであり、前記処理回路は、前記第１の適応フィルタの前記第１の周波数応答を第１の所定の周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の所定の周波数範囲に制限し、前記処理回路は、前記第２の適応フィルタの前記第２の周波数応答を第２の所定の周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の所定の周波数範囲に制限し、前記第１の所定の周波数範囲および前記第２の所定の周波数範囲は、実質的に異なり、これにより、前記第１の適応フィルタおよび前記第２の適応フィルタは、前記少なくとも１つのマイクロホン信号を複数の周波数帯に分離するためのクロスオーバとして動作する、処理回路と
を備える、集積回路。
【請求項１６】
前記第１の変換器の周囲オーディオ音および音響出力と前記第２の変換器の周囲オーディオ音および音響出力とを示すエラーマイクロホン信号を提供するためのエラーマイクロホンをさらに備え、前記第１の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第１の係数発生器を有し、前記処理回路は、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第２の係数発生器を有し、前記処理回路は、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第２の周波数応答の適応を制限する、請求項１５に記載の集積回路。
【請求項１７】
前記処理回路は、前記第２の所定の周波数範囲内の第１の所定の周波数コンテンツを有する第１の付加的信号を前記第１の係数発生器に入力された前記第１の信号に投入することによって、前記第１の係数発生器に入力された前記第１の信号の周波数コンテンツを改変し、前記処理回路は、前記第１の所定の周波数範囲内の第２の所定の周波数コンテンツを有する第２の付加的信号を第２の係数発生器に入力された前記第２の信号に投入することによって、前記第２の係数発生器に入力された前記第２の信号の周波数コンテンツを改変する、請求項１６に記載の集積回路。
【請求項１８】
前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、請求項１７に記載の集積回路。
【請求項１９】

前記処理回路は、前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理することにより、より高い周波数コンテンツソースオーディオ信号とより低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを提供し、前記処理回路は、前記より高い周波数コンテンツソースオーディオ信号を前記第1の反雑音信号と組み合わせ、前記より低い周波数コンテンツソースオーディオ信号を前記第2の反雑音信号と組み合わせる、請求項15に記載の集積回路。

【請求項20】

前記第1の出力信号は、イヤースピーカの高周波数変換器に提供するように適応され、前記第2の出力信号は、前記イヤースピーカの低周波数変換器に提供するように適応されている、請求項15に記載の集積回路。

【請求項21】

第3の出力信号を第3の変換器に提供するための第3の出力であって、前記第3の変換器は、第2のソースオーディオ信号と、前記第3の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第3の反雑音信号との高周波数コンテンツを再現する、第3の出力と、

第4の出力信号を第4の変換器に提供するための第4の出力であって、前記第4の変換器は、前記第2のソースオーディオ信号と、第4の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第4の反雑音信号との低周波数コンテンツを再現する、第4の出力と

をさらに備え、

前記処理回路は、第3のフィルタを使用して前記少なくとも1つのマイクロホン信号に応じて前記少なくとも1つのマイクロホン信号から前記第3の反雑音信号および前記第4の反雑音信号を発生させることにより、前記第3の変換器および前記第4の変換器における前記周囲オーディオ音の存在を低減させ、前記処理回路は、第4のフィルタを使用して前記少なくとも1つのマイクロホン信号に応じて前記少なくとも1つのマイクロホン信号から前記第4の反雑音信号を発生させることにより、前記第3の変換器および前記第4の変換器における前記周囲オーディオ音の存在を低減させる、請求項20に記載の集積回路。

【請求項22】

パーソナルオーディオシステムであって、前記パーソナルオーディオシステムは、

複数の出力変換器と、

周囲オーディオ音を示す少なくとも1つのマイクロホン信号を提供するための少なくとも1つのマイクロホンと、

適応雑音消去を実装する処理回路と

を備え、

複数の適応フィルタは、前記複数の出力変換器のうちの対応する出力変換器のための複数の反雑音信号を発生させ、前記複数の適応フィルタは、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも1つのマイクロホン信号を前記複数の出力変換器に対応する前記複数の周波数帯に分離するためのクロスオーバとして動作する、パーソナルオーディオシステム。

【請求項23】

パーソナルオーディオシステムによって、周囲オーディオ音の影響を打ち消す方法であって、前記方法は、

少なくとも1つのマイクロホンを用いて周囲オーディオ音を測定することにより、少なくとも1つのマイクロホン信号を発生させることと、

複数の適応フィルタのうちの対応する適応フィルタを使用して、複数の出力変換器のうちの対応する出力変換器に提供するための複数の反雑音信号を発生させることと

を含み、

前記対応する適応フィルタは、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも1つのマイクロホン信号を前記

複数の出力変換器に対応する前記複数の周波数帯に分離するためのクロスオーバとして動作する、方法。
【請求項２４】
パーソナルオーディオシステムの少なくとも一部を実装するための集積回路であって、前記集積回路は、
複数の出力信号を複数の出力変換器のうちの対応する出力変換器に提供するための複数の出力と、
周囲オーディオ音を示す少なくとも１つのマイクロホン信号を受信するための少なくとも１つのマイクロホン入力と、
適応雑音消去を実装する処理回路と
を備え、
複数の適応フィルタは、前記複数の出力のうちの対応する出力において複数の反雑音信号を発生させ、前記複数の適応フィルタは、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも１つのマイクロホン信号を前記複数の出力変換器に対応する前記複数の周波数帯に分離するためのクロスオーバとして動作する、集積回路。
【発明の詳細な説明】
【技術分野】
【０００１】
（発明の分野）
本発明は、概して、異なる周波数帯のための、適応雑音消去（ＡＮＣ）と複数のドライバとを含む、パーソナルオーディオデバイスに関する。
【背景技術】
【０００２】
（発明の背景）
モバイル／携帯電話、コードレス電話等の無線電話、およびＭＰ３プレーヤ等の他の消費者オーディオデバイスが、広く使用されている。明瞭度に関するそのようなデバイスの性能は、周囲音響事象を測定する基準マイクロホンを使用して、ＡＮＣを提供し、次いで、信号処理を使用して、反雑音信号をデバイスの出力に挿入し、周囲音響事象を消去することによって、改良されることができる。
【０００３】
パーソナルオーディオデバイスのために実装されるほとんどのオーディオシステムが、単一の出力変換器に依拠するが、無線電話の筐体上に搭載された変換器、もしくはイヤースピーカが使用されるとき、あるいは無線電話または他のデバイスがステレオスピーカを採用するときの一対の変換器の場合、高品質のオーディオ再現のために、高品質のイヤースピーカにおけるように、高周波および低周波数のための別個の変換器を提供することが望ましくあり得る。しかしながら、そのようなシステムにおいてＡＮＣを実装するとき、低周波数変換器と高周波数変換器との間で信号を分割するクロスオーバによって導入される待ち時間が、遅延をもたらし、これは、動作の増加された待ち時間に起因して、ＡＮＣシステムの有効性を低減させる。
【０００４】
したがって、異なる周波数帯を取り扱う複数の出力変換器を使用しながら、短待ち時間ＡＮＣ動作をもたらす無線電話および／またはイヤースピーカを含む、パーソナルオーディオシステムを提供することが望ましいであろう。
【発明の概要】
【課題を解決するための手段】
【０００５】
（発明の開示）
ＡＮＣを有するパーソナルオーディオデバイスを提供し、異なる周波数帯を取り扱うための複数の出力変換器を採用する上記に述べられた目的は、パーソナルオーディオシステ

ム、動作方法、および集積回路において達成される。

【０００６】

パーソナルオーディオデバイスは、聴取者への再生のためのソースオーディオ信号と、変換器の音響出力における周囲オーディオ音の影響を打ち消すための反雑音信号とを再現するための低周波数出力変換器および高周波数変換器の両方を含む。パーソナルオーディオデバイスはまた、適応雑音消去（ＡＮＣ）機能性を提供する集積回路を含む。方法は、パーソナルオーディオシステムおよび集積回路の動作方法である。基準マイクロホンが、周囲オーディオ音を示す基準マイクロホン信号を提供するために、本デバイス筐体上に搭載される。パーソナルオーディオシステムはさらに、反雑音信号が、その対応する変換器において、周囲オーディオ音の実質的消去を生じさせるように、反雑音信号を基準マイクロホン信号から適応的に発生させるためのＡＮＣ処理回路を含む。適応フィルタが、基準マイクロホン信号をフィルタ処理することによって、反雑音信号を発生させるために使用される。

【０００７】

本発明の前述ならびに他の目的、特徴、および利点は、付随の図面に図示されるように、本発明の好ましい実施形態の以下のより具体的説明から明白となるであろう。

本願明細書は、例えば、以下の項目も提供する。

（項目１）

パーソナルオーディオシステムであって、前記パーソナルオーディオシステムは、

再現のためのオーディオのソースであって、前記オーディオのソースは、ソースオーディオ信号を提供する、オーディオのソースと、

第１の変換器であって、前記第１の変換器は、聴取者への再生のための前記ソースオーディオ信号と、前記第１の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第１の反雑音信号との高周波数コンテンツを再現する、第１の変換器と、

第２の変換器であって、前記第２の変換器は、聴取者への再生のための前記ソースオーディオ信号と、前記第２の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第２の反雑音信号との低周波数コンテンツを再現する、第２の変換器と、

前記周囲オーディオ音を示す少なくとも１つのマイクロホン信号を提供するための少なくとも１つのマイクロホンと、

第１のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第１の反雑音信号および前記第２の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第１の変換器および前記第２の変換器における前記周囲オーディオ音の存在を低減させる、処理回路であって、前記処理回路は、第２のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第２の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第１の変換器および前記第２の変換器における前記周囲オーディオ音の存在を低減させる、処理回路と

を備える、パーソナルオーディオシステム。

（項目２）

前記第１のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第１の応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の適応フィルタである、項目１に記載のパーソナルオーディオシステム。

（項目３）

前記処理回路は、前記第１の適応フィルタの第１の周波数応答を第１の既定周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の既定周波数範囲に制限し、前記処理回路は、前記第２の適応フィルタの第２の応答を第２の既定周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の既定周波数範囲に制限し、前記第１の既定周波数範囲および前記第２の既定周波数範囲は、実質的に異なる、項目１に記載のパーソナルオーディオデバイス。

（項目４）

前記第１の変換器および前記第２の変換器の周囲オーディオ音および音響出力を示すエラーマイクロホン信号を提供するためのエラーマイクロホンをさらに備え、前記第１の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第１の係数発生器を有し、前記処理回路は、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第２の係数発生器を有し、前記処理回路は、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限する、項目３に記載のパーソナルオーディオデバイス。

（項目５）

前記処理回路は、前記第１の既定周波数範囲内における第１の既定周波数コンテンツを有する第１の付加的信号を、前記第１の係数発生器に入力された前記第１の信号に投入することによって、前記第１の係数発生器に入力された前記第１の信号の周波数コンテンツを改変し、前記処理回路は、前記第２の既定周波数範囲内における第２の既定周波数コンテンツを有する第２の付加的信号を前記第２の係数発生器に入力された前記第２の信号に投入することによって、前記第２の係数発生器に入力された前記第２の信号の周波数コンテンツを改変する、項目４に記載のパーソナルオーディオデバイス。

（項目６）

前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、項目５に記載のパーソナルオーディオデバイス。

（項目７）

前記処理回路は、より高い周波数コンテンツソースオーディオ信号と、より低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを提供するために、前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理し、前記処理回路はさらに、前記より高い周波数コンテンツソースオーディオ信号を前記第１の反雑音信号と組み合わせ、前記より低い周波数コンテンツソースオーディオ信号を前記第２の反雑音信号と組み合わせる、項目１に記載のパーソナルオーディオデバイス。

（項目８）

前記第１の変換器は、イヤースピーカの高周波数変換器であり、前記第２の変換器は、前記イヤースピーカの低周波数変換器である、項目１に記載のパーソナルオーディオデバイス。

（項目９）

第３の変換器であって、前記第３の変換器は、第２のソースオーディオ信号と、前記第３の変換器の音響出力内における周囲オーディオ音の影響を打ち消すための第３の反雑音信号との高周波数コンテンツを再現する、第３の変換器と、

第４の変換器であって、前記第４の変換器は、前記第２のソースオーディオ信号と、前記第４の変換器の音響出力内における周囲オーディオ音の影響を打ち消すための第４の反雑音信号との低周波数コンテンツを再現する、第４の変換器と

をさらに備え、

前記処理回路はさらに、第３のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第３の反雑音信号および前記第４の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第３の変換器における前記周囲オーディオ音の存在を低減させ、前記処理回路は、第４のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第４の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第４の変換器における前記周囲オーディオ音の存在を低減させる、項目８に記載のパーソナルオーディオデバイス。

（項目１０）

パーソナルオーディオシステムによる周囲オーディオ音の影響を打ち消す方法であって、前記方法は、

少なくとも１つのマイクロホン信号を生成するために、少なくとも１つのマイクロホンを用いて、周囲オーディオ音を測定することと、
第１に、第１のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、第１の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、第１の変換器における前記周囲オーディオ音の存在を低減させることと、
第２に、第２のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、第２の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、第２の変換器における前記周囲オーディオ音の存在を低減させることと、
再現のためのオーディオのソースを提供することであって、前記オーディオのソースは、ソースオーディオ信号を提供する、ことと、
前記第１の変換器を用いて、前記ソースオーディオ信号と前記第１の反雑音信号との高周波数コンテンツを再現することと、
前記第２の変換器を用いて、前記ソースオーディオ信号と前記第２の反雑音信号との低周波数コンテンツを再現することと
を含む、方法。
（項目１１）
前記第１のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第１の応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の適応フィルタである、項目１０に記載の方法。
（項目１２）
前記第１の発生させることは、前記第１の適応フィルタの第１の周波数応答を第１の既定周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の既定周波数範囲に制限することを含み、前記第２の発生させることはさらに、前記第２の適応フィルタの第２の応答を第２の既定周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の既定周波数範囲に制限することを含み、前記第１の既定周波数範囲および前記第２の既定周波数範囲は、実質的に異なる、項目１０に記載の方法。
（項目１３）
エラーマイクロホン信号を発生させるために、エラーマイクロホンを用いて、前記第１の変換器および前記第２の変換器の周囲オーディオ音および音響出力を測定することをさらに含み、前記第１の発生させることは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするために、前記第１の周波数応答を制御する第１の係数発生器の係数を適応させることを含み、前記第２の発生させることは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするために、第２の周波数応答を制御する第２の係数発生器の係数を適応させることを含み、前記第１の発生させることは、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の発生させることは、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第２の周波数応答の適応を制限する、項目１２に記載の方法。
（項目１４）
前記第１の発生させることは、前記第１の既定周波数範囲内における第１の既定周波数コンテンツを有する第１の付加的信号を前記第１の係数発生器への少なくとも１つの第１の信号入力に投入することによって、前記第１の周波数応答の適応を制限し、前記第２の発生させることは、前記第２の既定周波数範囲内における第２の既定周波数コンテンツを有する第２の付加的信号を前記第２の係数発生器への少なくとも１つの第２の信号入力に投入することによって、前記第２の周波数応答の適応を制限する、項目１３に記載の方法。
（項目１５）
前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、項目１４に記

載の方法。
(項目１６)
より高い周波数コンテンツソースオーディオ信号と、より低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを実装するために、前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理することと、
前記より高い周波数コンテンツソースオーディオ信号を前記第1 の反雑音信号と組み合わせることと、
前記より低い周波数コンテンツソースオーディオ信号を前記第2 の反雑音信号と組み合わせることと
をさらに含む、項目１０に記載の方法。
(項目１７)
前記第1 の変換器は、イヤースピーカの高周波数変換器であり、前記第2 の変換器は、前記イヤースピーカの低周波数変換器である、項目１０に記載の方法。
(項目１８)
第3 の変換器の音響出力内における周囲オーディオ音の影響を打ち消すために、前記第3 の変換器を用いて、第2 のソースオーディオ信号と第3 の反雑音信号との高周波数コンテンツを再現することと、
第4 の変換器の音響出力内における周囲オーディオ音の影響を打ち消すために、前記第4 の変換器を用いて、前記第2 のソースオーディオ信号と第4 の反雑音信号との低周波数コンテンツを再現することと、
第3 のフィルタを使用して、前記少なくとも1 つのマイクロホン信号と一致するように、前記第3 の反雑音信号および前記第4 の反雑音信号を前記少なくとも1 つのマイクロホン信号から発生させ、前記第3 の変換器および前記第4 の変換器における前記周囲オーディオ音の存在を低減させる、ことと、
第4 のフィルタを使用して、前記少なくとも1 つのマイクロホン信号と一致するように、前記第4 の反雑音信号を前記少なくとも1 つのマイクロホン信号から発生させ、前記第3 の変換器および前記第4 の変換器における前記周囲オーディオ音の存在を低減させる、ことと
をさらに含む、項目１７に記載の方法。
(項目１９)
パーソナルオーディオシステムの少なくとも一部を実装するための集積回路であって、前記集積回路は、
再現のためのオーディオのソースであって、前記オーディオのソースは、ソースオーディオ信号を提供する、オーディオのソースと、
第1 の出力信号を第1 の変換器に提供するための第1 の出力であって、前記第1 の変換器は、前記ソースオーディオ信号と、前記第1 の変換器の音響出力内の周囲オーディオ音の影響を打ち消すための第1 の反雑音信号との高周波数コンテンツを再現する、第1 の出力と、
第2 の出力信号を第2 の変換器に提供するための第2 の出力であって、前記第2 の変換器は、聴取者への再生のための第2 のソースオーディオと、第2 のイヤースピーカの音響出力内の周囲オーディオ音の影響を打ち消すための第2 の反雑音信号との両方を含む第2 のオーディオ信号を再現する、第2 の出力と、
前記周囲オーディオ音を示す少なくとも1 つのマイクロホン信号を提供するための少なくとも1 つのマイクロホン入力と、
第1 のフィルタを使用して、前記少なくとも1 つのマイクロホン信号と一致するように、前記第1 の反雑音信号および前記第2 の反雑音信号を前記少なくとも1 つのマイクロホン信号から発生させ、前記第1 の変換器および前記第2 の変換器における前記周囲オーディオ音の存在を低減させる、処理回路であって、前記処理回路は、第2 のフィルタを使用して、前記少なくとも1 つのマイクロホン信号と一致するように、前記第2 の反雑音信号を前記少なくとも1 つのマイクロホン信号から発生させ、前記第1 の変換器および前記第

２の変換器における前記周囲オーディオ音の存在を低減させる、処理回路と
を備える、集積回路。
(項目２０)
前記第１のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第１の応答を有する第１の適応フィルタであり、前記第２のフィルタは、前記周囲オーディオ音の存在を低減させるように適応する第２の適応フィルタである、項目１９に記載の集積回路。
(項目２１)
前記処理回路は、前記第１の適応フィルタの第１の周波数応答を第１の既定周波数範囲に限定することによって、前記第１の反雑音信号のコンテンツを前記第１の既定周波数範囲に制限し、前記処理回路は、前記第２の適応フィルタの第２の応答を第２の既定周波数範囲に限定することによって、前記第２の反雑音信号のコンテンツを前記第２の既定周波数範囲に制限し、前記第１の既定周波数範囲および前記第２の既定周波数範囲は、実質的に異なる、項目１９に記載の集積回路。
(項目２２)
前記第１の変換器および前記第２の変換器の周囲オーディオ音および音響出力を示すエラーマイクロホン信号を提供するためのエラーマイクロホンをさらに備え、前記第１の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第１の係数発生器を有し、前記処理回路は、前記第１の係数発生器に入力された第１の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限し、前記第２の適応フィルタは、前記エラーマイクロホン信号内に存在する基準マイクロホン信号の成分を最小限にするように適応する第２の係数発生器を有し、前記処理回路は、前記第２の係数発生器に入力された第２の信号の周波数コンテンツを改変することによって、前記第１の周波数応答の適応を制限する、項目２１に記載の集積回路。
(項目２３)
前記処理回路は、前記第１の既定周波数範囲内における第１の既定周波数コンテンツを有する第１の付加的信号を、前記第１の係数発生器に入力された前記第１の信号に投入することによって、前記第１の係数発生器に入力された前記第１の信号の周波数コンテンツを改変し、前記処理回路は、前記第２の既定周波数範囲内における第２の既定周波数コンテンツを有する第２の付加的信号を、第２の第１の係数発生器に入力された前記第２の信号に投入することによって、前記第２の係数発生器に入力された前記第２の信号の周波数コンテンツを改変する、項目２２に記載の集積回路。
(項目２４)
前記第１の付加的信号および前記第２の付加的信号は、雑音信号である、項目２３に記載の集積回路。
(項目２５)
前記処理回路は、より高い周波数コンテンツソースオーディオ信号と、より低い周波数コンテンツソースオーディオ信号とを発生させるクロスオーバを提供するために、前記ソースオーディオ信号を受信し、前記ソースオーディオ信号をフィルタ処理し、前記処理回路はさらに、前記より高い周波数コンテンツソースオーディオ信号を前記第１の反雑音信号と組み合わせ、前記より低い周波数コンテンツソースオーディオ信号を前記第２の反雑音信号と組み合わせる、項目１９に記載の集積回路。
(項目２６)
前記第１の変換器は、イヤースピーカの高周波数変換器であり、前記第２の変換器は、前記イヤースピーカの低周波数変換器である、項目１９に記載の集積回路。
(項目２７)
第３の出力信号を第３の変換器に提供するための第３の出力であって、前記第３の変換器は、第２のソースオーディオ信号と、前記第３の変換器の音響出力内における周囲オーディオ音の影響を打ち消すための第３の反雑音信号との高周波数コンテンツを再現する、

第３の出力と、
　第４の出力信号を第４の変換器に提供するための第４の出力であって、前記第４の変換器は、前記第２のソースオーディオ信号と、第４の変換器の音響出力内における周囲オーディオ音の影響を打ち消すための第４の反雑音信号との低周波数コンテンツを再現する、第４の出力と
　をさらに備え、
　前記処理回路はさらに、第３のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第３の反雑音信号および前記第４の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させ、前記処理回路は、第４のフィルタを使用して、前記少なくとも１つのマイクロホン信号と一致するように、前記第４の反雑音信号を前記少なくとも１つのマイクロホン信号から発生させ、前記第３の変換器および前記第４の変換器における前記周囲オーディオ音の存在を低減させる、項目２６に記載の集積回路。
（項目２８）
　パーソナルオーディオシステムであって、前記パーソナルオーディオシステムは、
　複数の出力変換器と、
　周囲オーディオ音を示す少なくとも１つのマイクロホン信号を提供するための少なくとも１つのマイクロホンと、
　適応雑音消去を実装する処理回路と
　を備え、
　複数の適応フィルタは、前記複数の出力変換器のうちの対応する出力変換器のための複数の反雑音信号を発生させ、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも１つのマイクロホン信号を前記複数の出力変換器に対応する前記複数の周波数帯に分離するためのクロスオーバとして動作する、パーソナルオーディオシステム。
（項目２９）
　パーソナルオーディオシステムによって、周囲オーディオ音の影響を打ち消す方法であって、前記方法は、
　少なくとも１つのマイクロホン信号を発生させるために、少なくとも１つのマイクロホンを用いて周囲オーディオ音を測定することと、
　複数の適応フィルタのうちの対応する適応フィルタを使用して、複数の出力変換器のうちの対応する出力変換器に提供するための複数の反雑音信号を発生させることと
　を含み、
　前記対応する適応フィルタは、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも１つのマイクロホン信号を前記複数の出力変換器に対応する前記複数の周波数帯に分離するためのクロスオーバとして動作する、方法。
（項目３０）
　パーソナルオーディオシステムの少なくとも一部を実装するための集積回路であって、前記集積回路は、
　複数の出力信号を複数の出力変換器のうちの対応する出力変換器に提供するための複数の出力と、
　周囲オーディオ音を示す少なくとも１つのマイクロホン信号を受信するための少なくとも１つのマイクロホン入力と、
　適応雑音消去を実装する、処理回路と
　を備え、
　複数の適応フィルタは、前記複数の出力のうちの対応する出力において複数の反雑音信号を発生させ、複数の周波数帯のうちの対応する周波数帯内の前記複数の反雑音信号を発生させることによって、前記少なくとも１つのマイクロホン信号を前記複数の出力変換器

に対応する前記複数の周波数帯に分離するためクロスオーバとして動作する、集積回路。

【図面の簡単な説明】

【０００８】

【図１Ａ】図１Ａは、例示的無線電話１０と、一対のイヤホンＥＢ１およびＥＢ２の例証図である。

【図１Ｂ】図１Ｂは、無線電話１０内部の回路の概略図である。

【図２】図２は、無線電話１０内部の回路のブロック図である。

【図３】図３は、図２のＣＯＤＥＣ集積回路２０ＡのＡＮＣ回路３０を実装するために使用されることができる、種々の例示的ＡＮＣ回路の信号処理回路および機能ブロックを描写する、ブロック図である。

【図４】図４は、ＣＯＤＥＣ集積回路２０内部の信号処理回路および機能ブロックを描写する、ブロック図である。

【発明を実施するための形態】

【０００９】

（発明を実施するための最良モード）

本発明は、無線電話および接続されたイヤホン等のパーソナルオーディオシステム内に実装され得る、雑音消去技法および回路を包含する。パーソナルオーディオシステムは、ソースオーディオ信号を受信し、またはそれを発生させるパーソナルオーディオデバイスの筐体上等のイヤホンまたは他の出力変換器場所において、周囲音響環境を測定し、消去しようとする、適応雑音消去（ＡＮＣ）回路を含む。高品質オーディオ出力を提供するために、対応するソースオーディオの周波数帯を再現する低周波数および高周波数変換器を含む、複数の変換器が、使用される。ＡＮＣ回路は、変換器において周囲音響事象を消去するために、複数の変換器のうちのそれぞれのものに提供される、別個の反雑音信号を発生させる。基準マイクロホンが、周囲音響環境を測定するために提供され、これは、短待ち時間が、発生された反雑音をフィルタ処理するクロスオーバの必要性を排除することによって維持されるように、反雑音信号を発生させる別個の適応フィルタへの入力を提供する。ソースオーディオクロスオーバは、次いで、ソースオーディオ周波数帯特有の成分とその対応する反雑音信号との加算器に先行して設置されることができ、適応フィルタは、その対応する変換器のために適切な周波数範囲内のみにおいて、反雑音を発生させるように制御されることができる。

【００１０】

図１Ａは、無線電話１０と、それぞれ、聴取者の対応する耳５Ａ、５Ｂに取り付けられる、一対のイヤホンＥＢ１およびＥＢ２とを示す。図示される無線電話１０は、本明細書に開示される技法が採用され得る、デバイスの実施例であるが、無線電話１０または後続例証に描写される回路内に図示される要素または構成は全て、要求されるわけではないことを理解されたい。無線電話１０は、有線または無線接続、例えば、ＢＬＵＥＴＯＯＴＨ（登録商標）接続（ＢＬＵＥＴＯＯＴＨ（登録商標）は、Ｂｌｕｅｔｏｏｔｈ（登録商標）　ＳＩＧ，　Ｉｎｃ．の商標である）によって、イヤホンＥＢ１、ＥＢ２に接続される。イヤホンＥＢ１、ＥＢ２はそれぞれ、無線電話１０から受信される遠隔発話、呼出音、記憶されたオーディオプログラム材料、および近端発話（すなわち、無線電話１０のユーザの発話）の投入を含む、ソースオーディオを再現する、それぞれ、対応する一対の変換器ＳＰＫＬＨ／ＳＰＫＬＬおよびＳＰＫＲＨ／ＳＰＫＲＬを有する。変換器ＳＰＫＬＨおよびＳＰＫＲＨは、可聴周波数および変換器ＳＰＫＬＬのより高い範囲を再現する、高周波数変換器または「ツイータ」であり、ＳＰＫＲＬは、オーディオ周波数のより低い範囲を再現する、低周波数変換器または「ウーファ」である。ソースオーディオはまた、無線電話１０が、無線電話１０によって受信されるウェブページまたは他のネットワーク通信からのソースオーディオならびに低バッテリ量および他のシステム事象通知等のオーディオ指標を再現することが要求される、任意の他のオーディオを含む。基準マイクロホンＲ１、Ｒ２は、周囲音響環境を測定するために、個別のイヤホンＥＢ１、ＥＢ２の筐体の表面上に提供される。別の対のマイクロホンである、エラーマイクロホンＥ１、Ｅ２は、イ

ヤホンＥＢ１、ＥＢ２が、耳５Ａ、５Ｂの外側部分内に挿入されると、対応する耳５Ａ、５Ｂに近接する、それぞれの変換器対ＳＰＫＬＨ／ＳＰＫＬＬおよびＳＰＫＲＨ／ＳＰＫＲＬによって再現されるオーディオと組み合わせられた周囲オーディオの測定値を提供することによって、ＡＮＣ動作をさらに改善するために提供される。

【００１１】

無線電話１０は、反雑音信号をＳＰＫＬＨ、ＳＰＫＬＬ、ＳＰＫＲＨ、およびＳＰＫＲＬに投入し、変換器ＳＰＫＬＨ、ＳＰＫＬＬ、ＳＰＫＲＨ、およびＳＰＫＲＬによって再現される遠隔発話および他のオーディオの明瞭度を改善する、適応雑音消去（ＡＮＣ）回路および特徴を含む。無線電話１０内部の例示的回路１４が、信号を基準マイクロホンＲ１、Ｒ２、近接発話マイクロホンＮＳ、およびエラーマイクロホンＥ１、Ｅ２から受信する、オーディオ集積回路２０を含み、無線電話送受信機を含有するＲＦ集積回路１２等の他の集積回路とインターフェースをとる。他の実装では、本明細書に開示される回路および技法は、ＭＰ３プレーヤオンチップ集積回路等のパーソナルオーディオデバイスの全体を実装するための制御回路および他の機能性を含有する、単一集積回路内に組み込まれてもよい。代替として、ＡＮＣ回路は、イヤホンＥＢ１、ＥＢ２の筐体内または無線電話１０とイヤホンＥＢ１、ＥＢ２との間の有線接続に沿って位置するモジュール内に含まれてもよい。例証の目的のために、ＡＮＣ回路は、無線電話１０内に提供されるように説明されるが、前述の変形例は、当業者によって理解可能であり、イヤホンＥＢ１、ＥＢ２、無線電話１０、および第３のモジュール間に要求される、結果として生じる信号は、要求に応じて、それらの変形例のために容易に判定されることができる。近接発話マイクロホンＮＳは、無線電話１０の筐体に提供され、無線電話１０から他の会話参加者に伝送される、近端発話を捕捉する。代替として、近接発話マイクロホンＮＳは、イヤホンＥＢ１、ＥＢ２の一方の筐体の外側表面上、イヤホンＥＢ１、ＥＢ２の一方に添着された支持部材上、あるいは無線電話１０とイヤホンＥＢ１、ＥＢ２の一方または両方との間に位置する付属物上に提供されてもよい。

【００１２】

図１Ｂは、対応するイヤホンＥＢ１、ＥＢ２内に位置するオーディオ集積回路２０Ａ、２０Ｂ内のＡＮＣ処理回路によってフィルタ処理される、周囲オーディオ音周囲１、周囲２の測定値を提供する、基準マイクロホンＲ１、Ｒ２に結合されるように、ＡＮＣ処理を含む、オーディオ集積回路２０Ａ、２０Ｂの簡略化された概略図を示す。オーディオ集積回路２０Ａ、２０Ｂは、代替として、無線電話１０内の集積回路２０等の単一集積回路内に組み合わせられてもよい。オーディオ集積回路２０Ａ、２０Ｂは、増幅器Ａ１－Ａ４のうちの関連付けられた１つによって増幅され、対応する変換器対ＳＰＫＬＨ／ＳＰＫＬＬおよびＳＰＫＲＨ／ＳＰＫＲＬに提供される、その対応するチャネルのための出力を発生させる。オーディオ集積回路２０Ａ、２０Ｂは、（特定の構成に応じて、有線または無線）信号を基準マイクロホンＲ１、Ｒ２、近接発話マイクロホンＮＳ、およびエラーマイクロホンＥ１、Ｅ２から受信する。オーディオ集積回路２０Ａ、２０Ｂはまた、図１Ａに示される無線電話送受信機を含有する、ＲＦ集積回路１２等の他の集積回路とインターフェースをとる。他の構成では、本明細書に開示される回路および技法は、ＭＰ３プレーヤオンチップ集積回路等のパーソナルオーディオデバイスの全体を実装するための制御回路および他の機能性を含有する、単一集積回路内に組み込まれてもよい。代替として、複数の集積回路は、例えば、無線接続が、イヤホンＥＢ１、ＥＢ２のそれぞれから無線電話１０に提供されるとき、および／またはＡＮＣ処理の一部または全部が、イヤホンＥＢ１、ＥＢ２または無線電話１０をイヤホンＥＢ１、ＥＢ２に接続するケーブルに沿って配置されるモジュール内で行なわれるとき、使用されてもよい。

【００１３】

一般に、本明細書で図示されるＡＮＣ技法は、基準マイクロホンＲ１、Ｒ２に衝突する、周囲音響事象（変換器ＳＰＫＬＨ、ＳＰＫＬＬ、ＳＰＫＲＨ、およびＳＰＫＲＬならびに／または近端発話とは対照的に）を測定し、また、エラーマイクロホンＥ１、Ｅ２に衝突する、同一の周囲音響事象を測定する。集積回路２０Ａ、２０ＢのＡＮＣ処理回路は、

個々に、対応する基準マイクロホンＲ１、Ｒ２の出力から発生される反雑音信号を適応し、対応するエラーマイクロホンＥ１、Ｅ２において、周囲音響事象の振幅を最小限にする特性を有する。音響経路P_L（ｚ）は、基準マイクロホンＲ１からエラーマイクロホンＥ１に延在するため、オーディオ集積回路２０Ａ内のＡＮＣ回路は、本質的に、それぞれ、オーディオ集積回路２０Ａのオーディオ出力回路の応答と、変換器ＳＰＫＬＨおよびＳＰＫＬＬの音響／電気伝達関数とを表す、電気音響経路S_{LH}（ｚ）およびS_{LL}（ｚ）の影響を除去した状態で組み合わせられた音響経路P_L（ｚ）を推定している。推定される応答は、耳５Ａの近接性および構造ならびにイヤホンＥＢ１に近接し得る他の物理的物体およびヒト頭部構造によって影響される、特定の音響環境内における変換器ＳＰＫＬＨおよびＳＰＫＬＬとエラーマイクロホンＥ１との間の結合を含む。同様に、オーディオ集積回路２０Ｂは、それぞれ、オーディオ集積回路２０Ｂのオーディオ出力回路の応答および変換器ＳＰＫＲＨおよびＳＰＫＲＬの音響／電気伝達関数を表す、電気音響経路S_{RH}（ｚ）、およびS_{RL}（ｚ）の影響を除去した状態で組み合わせられた音響経路P_R（ｚ）を推定する。

【００１４】

次に、図２を参照すると、イヤホンＥＢ１、ＥＢ２および無線電話１０内の回路が、ブロック図に示される。図２に示される回路はさらに、前述の他の構成にも適用されるが、無線電話１０内のＣＯＤＥＣ集積回路２０と他のユニットとの間の信号伝達は、オーディオ集積回路２０Ａ、２０Ｂが、無線電話１０の外部、例えば、対応するイヤホンＥＢ１、ＥＢ２内に位置するとき、ケーブルまたは無線接続によって提供される。そのような構成では、集積回路２０Ａ－２０Ｂを実装する単一集積回路２０と、エラーマイクロホンＥ１、Ｅ２、基準マイクロホンＲ１、Ｒ２、ならびに変換器ＳＰＫＬＨ、ＳＰＫＬＬ、ＳＰＫＲＨ、およびＳＰＫＲＬとの間の信号伝達は、オーディオ集積回路２０が、無線電話１０内に位置するとき、有線または無線接続によって提供される。図示される実施例では、オーディオ集積回路２０Ａ、２０Ｂは、別個かつ実質的に同じ回路として示され、したがって、オーディオ集積回路２０Ａのみ、以下に詳細に説明される。

【００１５】

オーディオ集積回路２０Ａは、基準マイクロホン信号を基準マイクロホンＲ１から受信し、基準マイクロホン信号のデジタル表現ｒｅｆを生成するアナログ／デジタルコンバータ（ＡＤＣ）２１Ａを含む。オーディオ集積回路２０Ａはまた、エラーマイクロホン信号をエラーマイクロホンＥ１から受信し、エラーマイクロホン信号のデジタル表現ｅｒｒを生成するためのＡＤＣ２１Ｂと、近接発話マイクロホン信号を近接発話マイクロホンＮＳから受信し、近接発話マイクロホン信号のデジタル表現ｎｓを生成するためのＡＤＣ２１Ｃとを含む。（オーディオ集積回路２０Ｂは、前述のように、無線または有線接続を介して、近接発話マイクロホン信号のデジタル表現ｎｓをオーディオ集積回路２０Ａから受信する。）オーディオ集積回路２０Ａは、結合器２６Ａの出力を受信する、デジタル／アナログコンバータ（ＤＡＣ）２３Ａの出力を増幅させる、増幅器Ａ１から変換器ＳＰＫＬＨを駆動させるための出力を発生させる。結合器２６Ｃが、左チャネル内部オーディオ信号ｉａｌと、ソースオーディオｄｓとを組み合わせ、これは、無線周波数（ＲＦ）集積回路２２から受信される。結合器２６Ａは、ソースオーディオ$ds_h + ia_{l_h}$を組み合わせ、これは、左チャネルＡＮＣ回路３０によって発生された高周波数帯反雑音信号$anti\text{-}noise_{l_h}$を伴う結合器２６Ｃの出力の高周波数帯成分であり、通例、基準マイクロホン信号ｒｅｆ内の雑音と同一の極性を有し、したがって、結合器２６Ａによって減算される。結合器２６Ａはまた、無線電話１０のユーザが、ダウンリンク発話ｄｓに適切に関連して、その自身の音声を聞き取れるように、近接発話信号ｎｓの減衰された高周波数部分、すなわち、側音情報st_hを組み合わせる。近接発話信号ｎｓはまた、ＲＦ集積回路２２に提供され、アンテナＡＮＴを介して、アップリンク発話としてサービスプロバイダに伝送される。同様に、左チャネルオーディオ集積回路２０Ａは、変換器ＳＰＫＬＬを駆動させるための出力を増幅器Ａ２から発生させ、これは、結合器２６Ｂの出力を受信するデジタル／アナログ変換器（ＤＡＣ）２３Ｂの出力を増幅する。結合器２６Ｂは、ソー

スオーディオds_l+ia_lを組み合わせ、これは、ＡＮＣ回路３０によって発生された低周波数帯反雑音信号$anti-noise_l$を伴う結合器２６Ｃの出力の低周波数帯成分であり、通例、基準マイクロホン信号ｒｅｆ内の雑音と同一の極性を有し、したがって、結合器２６Ｂによって減算される。結合器２６Ｂはまた、近接発話信号ｎｓの減衰された部分、すなわち、側音低周波数情報st_lを組み合わせる。

【００１６】

次に、図３を参照すると、図２のオーディオ集積回路２０Ｂを実装するために使用され得るような、ＡＮＣ回路３０内部の詳細の実施例が、示される。同一回路が、オーディオ集積回路２０Ａを実装するために使用され、以下に留意されるようなチャネル標識図内部に対する変更を伴う。それぞれ、高周波数チャネル５０Ａおよび低周波数チャネル５０Ｂが、反雑音信号$anti-noise_{rh}$および$anti-noise_{rl}$を発生させるために、提供される。信号および応答標識が、右チャネルを示す文字「ｒ」を含有する、以下の説明では、文字は、図２のオーディオ集積回路２０Ａの内部に実装されるような図３に従って、別の回路内の左チャネルを示す「ｌ」と置換されるであろう。信号および応答が、高周波数チャネル５０Ａにおける低周波数のための文字「ｈ」で標識化される場合、低周波数チャネル５０Ｂ内の対応する要素は、文字「ｌ」で標識化される信号および応答と置換されるであろう。適応フィルタ３２Ａが、基準マイクロホン信号ｒｅｆを受信し、理想的状況下、その伝達関数$W_{rh}(z)$を$P_r(z)/S_{rh}(z)$になるように適応させ、反雑音信号$anti-noise_{rh}$を発生させる。適応フィルタ３２Ａの係数は、概して、最小二乗平均的意味において、マイクロホン信号ｅｒｒ内に存在する基準マイクロホン信号ｒｅｆのそれらの成分間のエラーを最小限にする、２つの信号の相関を使用して、適応フィルタ３２Ａの応答を判定する、Ｗ係数制御ブロック３１Ａによって制御される。本明細書に開示される実施例は、フィードフォワード構成で接続された適応フィルタ３２Ａを使用するが、本明細書に開示される技術は、固定またはプログラム可能フィルタを有する雑音消去システムにおいて実装されることができ、適応フィルタ３２Ａの係数が、事前設定される、選択される、または別様に連続的に適応されない場合、また、代替として、または固定フィルタトポロジと組み合わせた場合、本明細書に開示される技術は、フィードバックＡＮＣシステムまたはハイブリッドフィードバック／フィードフォワードＡＮＣシステムに適用されることができる。Ｗ係数制御ブロック３１Ａへの入力として提供される信号は、フィルタ３４Ｂによって提供される経路$S_{rh}(z)$の応答の推定値のコピーと、エラーマイクロホン信号ｅｒｒを含む結合器３６Ｃの出力から提供される別の信号とによって成形されるような基準マイクロホン信号ｒｅｆである。基準マイクロホン信号ｒｅｆを経路$S_{rh}(z)$の応答の推定値のコピーである、応答$SE_{rhCOPY}(z)$で変換し、基準マイクロホン信号ｒｅｆの成分と相関するエラー信号の部分を最小限にすることによって、適応フィルタ３２Ａは、$P_r(z)/S_{rh}(z)$の所望の応答に適応する。

【００１７】

エラーマイクロホン信号ｅｒｒに加え、Ｗ係数制御ブロック３１Ａによってフィルタ３４Ｂの出力とともに処理される他の信号として、応答$SE_{rh}(z)$を（応答$SE_{rhCOPY}(z)$は、そのコピーである）有する、２次経路フィルタ３４Ａによって処理される、ダウンリンクオーディオ信号ｄｓおよび内部オーディオｉａｎを含む、ソースオーディオ$(ds+ia_r)$の逆数量を含む。まず、ソースオーディオ$(ds+ia_r)$が、高域フィルタ３５Ａによって高周波数チャネル５０Ａに提供される前に、フィルタ処理され、これは、高周波数変換器ＳＰＫＬＨまたはＳＰＫＲＨによってもたらされた周波数のみ通過させる。同様に、まず、低周波数チャネル５０Ｂに提供されるソースオーディオ$(ds+ia_r)$は、低域フィルタ３５Ｂによってフィルタ処理され、これは、低周波数変換器ＳＰＫＬＬまたはＳＰＫＲＬによってもたらされた周波数のみを通過させる。したがって、高域フィルタ３５Ａおよび低域フィルタ３５Ｂは、適切な周波数のみが、それぞれ、高周波数チャネル５０Ａおよび低周波数チャネル５０Ｂに通過されるように、ソースオーディオ$(ds+ia_r)$に対するクロスオーバを形成し、それぞれの変換器ＳＰＫＬＨ、

ＳＰＫＬＬ、またはＳＰＫＲＨ、ＳＰＫＲＬに適切な帯域幅を有する。応答$SE_{rh}(z)$によってフィルタ処理されたソースオーディオ（ｄｓ＋ｉａ$_r$）の逆数量を投入することによって、適応フィルタ３２Ａは、エラーマイクロホン信号ｅｒｒ内に存在する比較的に大量のソースオーディオに適応しないように防止される。経路$S_{rh}(z)$の応答の推定値でソースオーディオ（ｄｓ＋ｉａ$_r$）の逆数コピーを変換することによって、処理前にエラーマイクロホン信号ｅｒｒから除去されるソースオーディオは、エラーマイクロホン信号ｅｒｒにおいて再現されるソースオーディオ（ｄｓ＋ｉａ$_r$）の予期されるバージョンに整合するはずである。ソースオーディオ量は、$S_{rh}(z)$の電気および音響経路が、エラーマイクロホンＥに到達するために、ソースオーディオ（ｄｓ＋ｉａ$_r$）によって辿られる経路であるため、整合する。フィルタ３４Ｂは、それ自体は、適応フィルタではないが、フィルタ３４Ｂの応答が、２次経路適応フィルタ３４Ａの適応を追跡するように、２次経路適応フィルタ３４Ａの応答に整合するように同調される、調節可能応答を有する。前述を実装するために、２次経路適応フィルタ３４Ａは、ＳＥ係数制御ブロック３３Ａによって制御される係数を有する。２次経路適応フィルタ３４Ａは、低または高周波数ソースオーディオ（ｄｓ＋ｉａ$_r$）を処理し、エラーマイクロホンＥに送達される予期されるソースオーディオを表す、信号を提供する。２次経路適応フィルタ３４Ａは、それによって、エラーマイクロホン信号ｅｒｒから減算されると、ソースオーディオ（ｄｓ＋ｉａ$_r$）に起因しないエラーマイクロホン信号ｅｒｒのコンテンツを含有する、エラー信号ｅを形成する信号をソースオーディオ（ｄｓ＋ｉａ$_r$）から発生させるように適応される。結合器３６Ｃは、フィルタ処理されたソースオーディオ（ｄｓ＋ｉａ$_r$）をエラーマイクロホン信号ｅｒｒから除去し、前述のエラー信号ｅを発生させる。

【００１８】

高周波数チャネル５０Ａおよび低周波数チャネル５０Ｂのそれぞれは、それぞれの反雑音信号ａｎｔｉ－ｎｏｉｓｅ$_h$およびａｎｔｉ－ｎｏｉｓｅ$_l$を発生させるように、独立して動作することができる。しかしながら、エラー信号ｅおよび基準マイクロホン信号ｒｅｆが、反雑音信号ａｎｔｉ－ｎｏｉｓｅ$_h$およびａｎｔｉ－ｎｏｉｓｅ$_l$を帯域限定せずに、オーディオ帯域内の任意の周波数の周波数を含有し得るため、それらのそれぞれの高周波数および低周波数変換器ＳＰＫＲＨ／ＳＰＫＬＨおよびＳＰＫＲＬ／ＳＰＫＬＬに送信されるべきではない成分を含有し得る。したがって、雑音投入技術が、適応フィルタ３２Ａの応答$W_{rh}(z)$を制御するために使用される。雑音ソース３７が、適応フィルタ３２Ｂによって提供される適応フィルタ３２Ａの応答$W_{rh}(z)$のコピー$W_{rhCOPY}(z)$に供給される、出力雑音信号$n_h(z)$を発生させる。結合器３６Ａが、雑音信号$n_h(z)$をＷ係数制御３１Ａに提供される、出力適応フィルタ３４Ｂに追加する。フィルタ３２Ｂによって成形されるような雑音信号$n_h(z)$は、雑音信号$n_h(z)$が、Ｗ係数制御３１Ａへの相関入力に非対称的に追加されるように、結合器３６Ｂによって結合器３６Ｃの出力から減算され、その結果、適応フィルタ３２Ａの応答$W_{rh}(z)$が、Ｗ係数制御３１Ａへの各相関入力に雑音信号$n_h(z)$の完全に相関性がある投入によってバイアスされる。投入雑音が、直接Ｗ係数制御３１Ａへの基準入力内に現れ、エラーマイクロホン信号ｅｒｒにおいて現われず、結合器３６Ｂによって、フィルタ３２Ｂの出力においてフィルタ処理された雑音の組み合わせを介して、Ｗ係数制御３１Ａへの他の入力においてのみ現れるため、Ｗ係数制御３１Ａは、$n_h(z)$内に存在する周波数を減衰させるために、$W_{rh}(z)$を適応させるであろう。雑音信号$n_h(z)$のコンテンツは、反雑音信号内に現われず、雑音信号$n_h(z)$がエネルギーを有する周波数／帯域において振幅減少を有するであろう適応フィルタ３２Ａの応答$W_{rh}(z)$においてのみ現れる。

【００１９】

低周波数が、反雑音信号ａｎｔｉ－ｎｏｉｓｅ$_h$内で発生されることを防止するために、雑音ソース３７は、低周波数帯においてエネルギーを有するスペクトルを有する、雑音を発生させ、これは、Ｗ係数制御３１Ａに、投入雑音信号$n_h(z)$に起因する周囲音響音の明白なソースを消去しようとして、これらの低周波数帯内において適応フィルタ３２

Ａの利得を減少させるであろう。例えば、白色雑音ソースが、高周波数チャネル５０Ａ内の雑音ソース３７としての使用のための低域フィルタ３５Ｂの応答に類似する応答によってフィルタ処理され得、これは、適応フィルタ３２Ａに、低域フィルタ３５Ｂの通過帯域の領域内において低利得を有させるであろう。低周波数チャネル５０Ｂのために同一のこと、すなわち、高域フィルタ３５Ａの応答に合致する応答を用いて白色雑音ソースをフィルタ処理することによって、クロスオーバが、それぞれの反雑音信号ａｎｔｉ－ｎｏｉｓｅ$_h$およびａｎｔｉ－ｎｏｉｓｅ$_l$内の望ましくない周波数を防止する、高周波数チャネル５０Ａおよび低周波数チャネル５０Ｂ内における適応フィルタ３２Ａの適応によって効果的に形成される。類似する構築が、２次経路適応フィルタ３４Ａの周囲に形成され得るが、２次経路適応フィルタ３４Ａへの入力が、帯域エネルギーから除去するために、フィルタ３５Ａおよび３５Ｂのうちのそれぞれ１つによってすでにフィルタ処理されているため、そのような雑音投入は、望ましくない周波数を２次経路適応フィルタ３４Ａの出力から除去するために必要とされないはずである。望ましくないクロスオーバエネルギーを反雑音信号ａｎｔｉ－ｎｏｉｓｅ$_h$およびａｎｔｉ－ｎｏｉｓｅ$_l$から除去するために、付加的フィルタ処理ではなく、雑音投入を使用する利点の１つは、付加的待ち時間が、雑音ソース３７に起因する応答の変更に起因する任意の待ち時間以外、導入されないことである。

【００２０】

次に、図４を参照すると、図３に描写されるようなＡＮＣ技法を実装し、図２のオーディオ集積回路２０Ａ、２０Ｂ内に実装され得るような処理回路４０を有するためのＡＮＣシステムのブロック図が、示され、１つの回路内に組み合わせられるように図示されるが、相互通信する２つ以上の処理回路としても実装され得る。処理回路４０は、前述のＡＮＣ技法の一部または全部ならびに他の信号処理を実装し得る、コンピュータプログラム製品を含む、プログラム命令が記憶されたメモリ４４に結合される、プロセッサコア４２を含む。随意に、専用デジタル信号処理（ＤＳＰ）論理４６が、処理回路４０によって提供されるＡＮＣ信号処理の一部、または代替として、全部を実装するために提供されてもよい。処理回路４０はまた、それぞれ、基準マイクロホンＲ１、エラーマイクロホンＥ１、近接発話マイクロホンＮＳ、基準マイクロホンＲ２、およびエラーマイクロホンＥ２から入力を受信するために、ＡＤＣ２１Ａ－２１Ｅを含む。基準マイクロホンＲ１、エラーマイクロホンＥ１、近接発話マイクロホンＮＳ、基準マイクロホンＲ２、およびエラーマイクロホンＥ２のうちの１つ以上が、デジタル出力を有する、または遠隔ＡＤＣからデジタル信号として通信される、代替実施形態では、ＡＤＣ２１Ａ－２１Ｅのうちの対応する１つは、省略され、デジタルマイクロホン信号は、直接、処理回路４０にインターフェースがとられる。ＤＡＣ２３Ａおよび増幅器Ａ１もまた、前述のような反雑音を含む、変換器出力信号を変換器ＳＰＫＬＨに提供するために、処理回路４０によって提供される。同様に、ＤＡＣ２３Ｂ－２３Ｄおよび増幅器Ａ２－Ａ４は、他の変換器出力信号を変換器対ＳＰＫＬＨ、ＳＰＫＬＬ、ＳＰＫＲＨ、およびＳＰＫＲＬに提供する。変換器出力信号は、デジタル出力信号を音響的に再現するモジュールに提供するためのデジタル出力信号であってもよい。

【００２１】

本発明は、特に、その好ましい実施形態を参照して図示および説明されたが、形態および詳細における前述ならびに他の変形例も本発明の精神および範囲から逸脱することなく、本明細書において行なわれてもよいことが、当業者によって理解されるであろう。

【 図１ Ａ 】

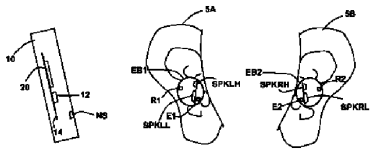


Fig. 1A

【 図１ Ｂ 】

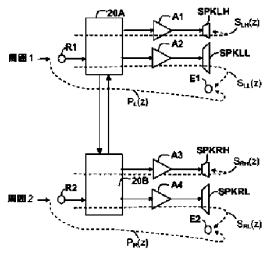


Fig. 1B

【 図２ 】

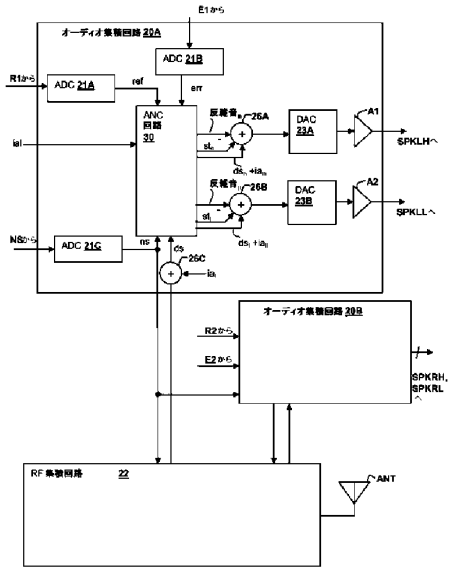


Fig. 2

【 図３ 】

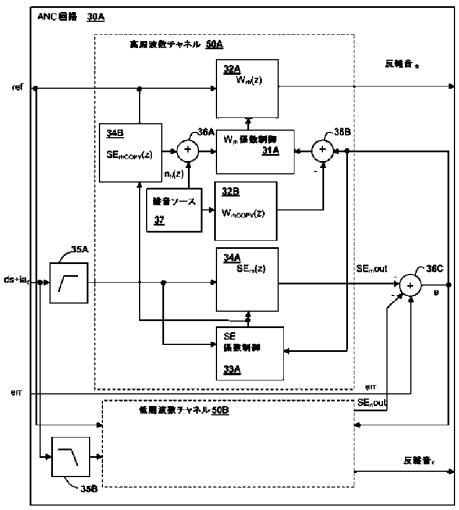


Fig. 3

【 図４ 】

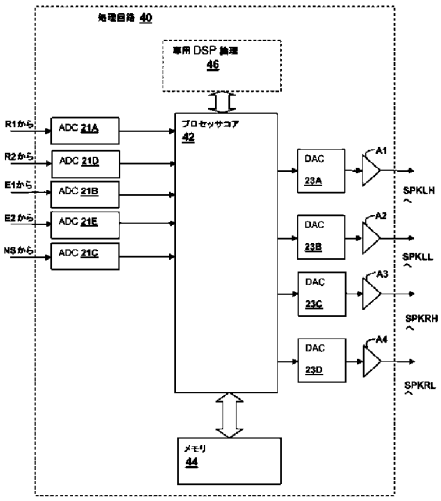


Fig. 4

フロントページの続き

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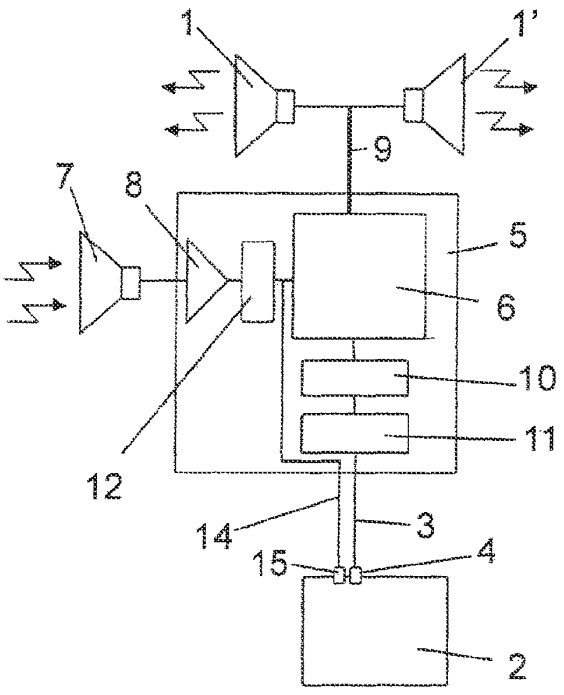
(81) Estados designados *(a menos que se indique otra cosa, para toda clase de protección nacional admisible)*: AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

(84) Estados designados *(a menos que se indique otra cosa, para toda clase de protección regional admisible)*: ARIPO

[Continúa en la página siguiente]

(54) Title: MULTIFUNCTION HEADPHONES

(54) Título: AURICULARES MULTIFUNCIÓN



(57) Abstract: Specially designed to enable the user to listen simultaneously to the audio signal coming from a reproduction device (2), and to ambient sound, the headphones are formed from standard speakers (1-1') associated with a control circuit (5) which includes an audio signals mixer (6), the input of which is associated with the audio line (3) coming from the reproduction equipment (2) and at least one microphone suitably placed to capture the ambient sound, including the corresponding amplification stage (8), in such a way that both signals are simultaneously reproduced through said speakers (1-1'). The control circuit is programmed to auto-level the audio signals, and has activation/disablement means (11-12) for the same. The improvements consist in providing the headphones with an internal supply source (16) which converts the device into an autonomous element, which can be used independently of the reproduction equipment (2-2'), for example as an audiphone. Another improvement consists in integrating in the control circuit (5) a wireless communications module (17), which can replace the cable of connection (3) to the reproduction equipment (2) or form an additional element, which extends the connection options for the device, although in this case the headphones have been designed to simultaneously receive several audio signals from different devices (2-2') which may be activated or disabled by means of the respective connection switches (11-11').

[Continúa en la página siguiente]

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(BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), euroasiática (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), europea (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, MT, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Publicada:

- *con informe de búsqueda internacional*
- *antes de la expiración del plazo para modificar las reivindicaciones y para ser republicada si se reciben modificaciones*

(57) Resumen: Especialmente concebidos para permitir a su usuario escuchar simultáneamente la señal de audio proveniente de un dispositivo reproductor (2), así como el sonido ambiente que le rodea, los auriculares están constituidos a partir de los clásicos altavoces (1-1') asociados a un circuito de control (5) en el que se establece un mezclador (6) de señales de audio, mezclador (6) cuya entrada está asociada a la línea de audio (3) proveniente del equipo reproductor (2) y a al menos un micrófono (7) convenientemente dispuesto para captar el sonido ambiental, incluyendo la correspondiente etapa de amplificación (8), de manera que a través de los citados altavoces (1-1') se reproducen simultáneamente ambas señales. El circuito de control está programado para autonivelar las señales de audio, contando con medios de activación/inhabilitación (11-12) de las mismas. Las mejoras consiste en dotar a los auriculares de una fuente de alimentación interna (16), que conviertan al dispositivo en un elemento autónomo, que puede ser utilizado independientemente del equipo reproductor (2-2'), por ejemplo a modo de audífono. Otra mejora consiste en la incorporación al circuito de control (5) de un módulo de comunicaciones inalámbricas (17), que puede sustituir al cable de conexión (3) al equipo reproductor (2) o bien constituir un elemento adicional, que amplíe las posibilidades de conexionado del dispositivo, si bien en este caso se ha previsto que los auriculares puedan recibir de forma simultánea varias señales de audio, provenientes de diferentes dispositivos (2-2') y habilitables/deshabilitables a través de respectivos interruptores de corte (11-11').

AURICULARES MULTIFUNCIÓN

DESCRIPCIÓN

OBJETO DE LA INVENCIÓN

La presente invención se refiere a unos auriculares, de los destinados permitir la escucha de señales de audio provenientes de equipos de reproducción de señales de audio tales como, ordenadores, radios, mp3's, "walkman", y similares.

El objeto de la invención es proporcionar unos auriculares que permitan a su usuario escuchar paralelamente tanto la señal de audio proveniente del equipo reproductor de que se trate, así como el sonido ambiente de los alrededores en donde se encuentre el usuario.

Es asimismo objeto de la invención que el dispositivo permita regular el volumen de entrada de una y otra señal a voluntad, hasta el punto de inhibir momentáneamente cualquiera de las dos.

La invención resulta de especial aplicación en el ámbito de las video-llamadas o video-conferencias, pudiendo no obstante aplicarse diferentes dispositivos como los anteriormente citados.

Las mejoras de la invención tienen por objeto que el dispositivo pueda amplificar las señales de audio correspondientes al sonido ambiente sin depender del dispositivo reproductor de audio al que se conecta normalmente.

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Es también objeto de la~~s mejoras~~ dotar a los auriculares de un sistema de comunicaciones inalámbricas con el reproductor de audio anteriormente citado, en orden a proporcionar un dispositivo más cómodo de portar, evitando las molestias que supone la dependencia de cables, o en su caso, permitiendo controlar la entrada de audio de un dispositivo inalámbrico y un dispositivo al que se conecta a través del correspondiente cable, de forma simultánea.

ANTECEDENTES DE LA INVENCIÓN

Es frecuente el uso de auriculares a la hora de escuchar la radio, música, etc. a través de dispositivos reproductores tanto fijos como portátiles, como pueden ser reproductores de CD's, MP3, radios, mini-disc, ordenadores, cadenas de música, etc.

El problema que presentan este tipo de dispositivos es que presentan una configuración que aísla el oído del usuario frente al sonido ambiente exterior, propiedad que puede no ser deseable en muchos casos, como por ejemplo en el caso de reuniones en las que participan múltiples personas, y en las que algunas de ellas emplean auriculares para realizar video-conferencias, escuchar traducciones de algún participante, etc, lo que obliga a estar continuamente quitándose y poniéndose los citados auriculares, de manera que cuando el usuario los tiene puestos no es capaz de seguir las conversaciones que se estén manteniendo a su alrededor.

Por otro lado, y en el ámbito de la escucha de señales acústicas provenientes de dispositivos reproductores portátiles, el citado aislamiento del usuario frente a los sonidos de su entorno puede resultar en ocasiones peligroso, al no advertir éste cualquier alarma o advertencia de naturaleza

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sonora.

Tratando de obviar este problema, en ciertos tipos de auriculares en los que el auricular izquierdo es independiente físicamente del auricular derecho, es factible que su usuario se adapte únicamente uno de los dos auriculares para poder estar atento al sonido ambiente proximal, pero esta solución impide que el mismo pueda escuchar la señal de audio del reproductor en estéreo, reduciendo sensiblemente la calidad de la misma, por lo que resulta una solución poco satisfactoria.

En la Patente de Invención P200603276 se describen unos auriculares especialmente concebidos para permitir a su usuario escuchar simultáneamente la señal de audio proveniente de un dispositivo reproductor, así como el sonido ambiente que le rodea, de manera que los auriculares están constituidos a partir de los clásicos altavoces, asociados a un circuito de control en el que se establece un mezclador de señales de audio, mezclador cuya entrada está asociada a la línea de audio proveniente del equipo reproductor y a al menos un micrófono convenientemente dispuesto para captar el sonido ambiental, incluyendo la correspondiente etapa de amplificación, de manera que a través de los citados altavoces se reproducen simultáneamente ambas señales.

Dicho circuito de control está programado para autonivelar las señales de audio, si bien su usuario a través de los correspondientes interruptores de corte puede habilitar/deshabilitar una u otra señal manualmente.

El problema que presenta este dispositivo es que depende eléctricamente del dispositivo reproductor de que se trate, de manera que el mismo no puede ser utilizado independientemente, como por ejemplo a modo

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de audífono, para amplificar las señales de audio del entorno.

Igualmente, los auriculares se relacionan con el dispositivo reproductor de que se trate a través de un cable, que en ocasiones puede resultar molesto, limitando los movimientos del usuario.

DESCRIPCIÓN DE LA INVENCIÓN

Los auriculares que la invención propone resuelven de forma plenamente satisfactoria la problemática anteriormente expuesta, permitiendo realizar de forma simultánea la escucha de una señal de audio proviniente de un reproductor paralelamente a la señal de audio correspondiente al ambiente de los alrededores del dispositivo.

Para ello, y de forma más concreta, se ha previsto que partiendo de la estructuración clásica de unos auriculares convencionales, sean del tipo que sean, el altavoz o altavoces estén asociados a un circuito electrónico, dotado de un mezclador de señales, circuito al que accede por un lado la señal de sonido proveniente del equipo reproductor de que se trate a través de la correspondiente línea, y por otro la señal proviniente de al menos un micrófono destinado a recibir los sonidos que se emitan en las proximidades del dispositivo.

De acuerdo con otra de las características de la invención, el citado circuito de control contará con medios de regulación del volumen así como de activación/desactivación de cada una de las señales.

Asimismo, el circuito de control estará programado para realizar la

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auto-nivelación del volumen de los auriculares, de manera dicho circuito se encarga de eliminar los sonidos externos que no superen un valor de decibelios preestablecido como nivel de activación de la señal externa y auto nivelación, regulando igualmente el nivel del volumen de la señal de audio proviniente del dispositivo reproductor, reestableciendo el nivel original de forma progresiva a término de la señal de entrada de audio externa proviniente del micrófono.

Por último cabe destacar que el micrófono o micrófonos podrán integrarse en la carcasa en la que se integra cada altavoz, o bien de forma independiente, de manera que dicho micrófono puede ser aprovechado para su empleo en video-conferencias, video-llamadas y similares, de manera que el circuito de control incorpore una línea de entrada para transmisión de la señal al equipo de que se trate.

Se consigue de esta forma un dispositivo que permite escuchar conjuntamente el sonido proviniente de un reproductor de audio así el proviniente del exterior, cómodo, sencillo de usar y totalmente automático.

Las mejoras que la invención propone resuelven de forma plenamente satisfactoria la problemática anteriormente expuesta, pudiendo constituir un elemento física y energéticamente independiente del dispositivo reproductor de que se trate, de manera que puede ser utilizado exclusivamente como audífono, o bien estar asociado a más de un dispositivo reproductor, a través de conexiones inalámbricas y/o alámbricas, cuyas señales podrán ser igualmente habilitadas/inhabilitadas de forma manual a través de los correspondientes interruptores de corte.

Para ello se ha previsto que, a partir de la estructura básica descrita en la patente de invención P200603276, el circuito de control incorpore una fuente de alimentación eléctrica interna, que permita a los auriculares funcionar

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independientemente del dispositivo reproductor, de manera que, como ya se ha apuntado con anterioridad, los auriculares pueden utilizarse simplemente para amplificar las señales de audio correspondientes al sonido ambiente que rodea al usuario, a modo de audífono.

De acuerdo con otra de las mejoras previstas para la invención, se ha previsto que dicho circuito de control incorpore un módulo de transmisión de datos inalámbrico, bien por radiofrecuencias o bien por infrarrojos, en sustitución del cable de conexión entre los auriculares y el dispositivo reproductor, si bien los auriculares podrían disponer de ambas opciones, a fin de ser compatibles con un mayor número de dispositivos así como para permitir la transferencia de señales de audio simultánea con dos o más dispositivos.

De ésta forma, el circuito de control o mezclador podría aunar más de dos señales de audio, incluyendo los correspondientes interruptores de corte, de manera que, por ejemplo podría utilizarse simultáneamente para escuchar una señal de audio proviniente de un reproductor convencional, conectado a los auriculares, por ejemplo a través del cable, el sonido ambiental a través del micrófono asociado a los auriculares, y la señal proviniente de un tercer dispositivo reproductor, como por ejemplo un elemento inalámbrico dotado de un micrófono, como los utilizados para su instalación en habitaciones en las que se encuentren alojados bebés, en orden a permitir escuchar lo que sucede en dicho habitáculo.

Como es evidente, las funciones previstas en la patente P200603276 entre los auriculares y el reproductor podrán realizarse igualmente a través del módulo de comunicaciones inalámbricas, como por ejemplo la grabación de la señal de audio captada por el micrófono asociado a dichos auriculares, o su recepción para el empleo del dispositivo en dispositivos que permitan realizar videollamadas, así como cualquier otra aplicación.

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Se consigue de esta manera un dispositivo enormemente polivalente, que permite mezclar y regular múltiples señales de audio, pudiendo igualmente constituir un elemento físicamente independiente y autónomo.

DESCRIPCIÓN DE LOS DIBUJOS

Para complementar la descripción que se está realizando y con objeto de ayudar a una mejor comprensión de las características del invento, de acuerdo con un ejemplo preferente de realización práctica del mismo, se acompaña como parte integrante de dicha descripción, un juego de dibujos en donde con carácter ilustrativo y no limitativo, se ha representado lo siguiente:

La figura 1.- Muestra una representación esquemática de los distintos componentes que participan en unos auriculares multifunción realizados de acuerdo con el objeto de la invención.

La figura 2.- Muestra una vista en perspectiva de una posible realización práctica de la invención.

La figura 3.- Muestra una vista en perspectiva dos realizaciones prácticas distintas de la invención, en las que el dispositivo se materializa en un único auricular.

Para complementar la descripción que se está realizando y con objeto de ayudar a una mejor comprensión de las características del invento, de acuerdo con un ejemplo de realización práctica del mismo, se acompaña como parte integrante de dicha descripción, una hoja única de planos en la que con carácter ilustrativo y no limitativo y en su única figura, se ha representado de

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forma esquemática un diagrama de bloques de los auriculares en el que se pueden observar las diferentes mejoras objeto de la invención

REALIZACIÓN PREFERENTE DE LA INVENCIÓN

A la vista de las figuras reseñadas y en especial de la figura 1, puede observarse como en los auriculares de la invención participan, como es convencional, una pareja de altavoces (1-1'), que se integran en la carcasa del correspondiente auricular, altavoces asociados a un equipo reproductor de audio (2), a través de un cable o línea de audio (3) con el complementario conector (4).

Pues bien, a partir de la estructuración convencional anteriormente descrita, la invención centra sus características en el hecho de que entre los altavoces (1-1') y la línea de audio (3) se establece un circuito de control (5) en el que se integra un mezclador de señales de audio (6).

Al citado mezclador confluyen las señales de la línea de audio (3) del reproductor (2), así como la señal proviniente de al menos un micrófono (7) previa amplificación (8) de la señal.

Así pues, el circuito a la salida (9) mezcla ambas señales según los parámetros de programación interna, de manera tal que el usuario escucharía ambas señales si no dispusiera de unos auriculares y estuviera escuchando dicha señal de audio a través de uno o más altavoces convencionales.

Complementariamente y tal y como se ha dicho con anterioridad, el circuito de control (4) estará programado para realizar de forma automática la auto-nivelación de las señales, incorporando filtros internos que eliminen lo que

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podría considerarse como ruido ambiental, constituido por sonidos inferiores a un nivel mínimo preestablecido.

El circuito de control cuenta asimismo con medios de regulación (10) del volumen de la señal de audio proveniente del equipo reproductor, así como interruptores de corte (11) y (12) para inhabilitación de una u otra señal entrante al mezclador (6) según se estime conveniente.

El micrófono (7) puede estar integrado en la carcasa (7-7') del auricular, tal como muestran las realizaciones mostradas en la figura 3, o bien establecerse en cualquier otra zona de los auriculares, como por ejemplo a la altura del cuello del usuario, tal como muestra la figura 2, pudiéndose utilizar dicho micrófono para la captura de la señal de la voz del usuario, en cuyo caso el circuito de control (5) dispondrán de una línea de entrada (14) con su correspondiente conector (15) para tratamiento de la señal en el equipo (2) de que se trate.

A la vista de la figura reseñada puede observarse como las mejoras que se preconizan parten de la estructuración básica de la patente P200603276 en la que participan, una pareja de altavoces (1-1'), que se integran en la carcasa del correspondiente auricular, asociados a un circuito de control (5), dotado de mandos de regulación del nivel de volumen (10), circuito en el que se integra un mezclador de señales de audio (6), una de las cuales proviene de un micrófono (7) asociado a un amplificador (8) y al correspondiente interruptor de corte (12).

Pues bien, a partir de la estructuración anteriormente descrita, una de las mejoras consiste en la incorporación de una fuente de alimentación (16) asociada al circuito de control (5), tal como una pila, batería o similar, en orden a dotar al dispositivo de un carácter autónomo, independiente del reproductor o

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reproductores (2-2') al que pueda estar asociado.

Tal y como se ha comentado con anterioridad, y de acuerdo con otra de las mejoras de la invención, el circuito de control puede incorporar un módulo inalámbrico de transmisión de datos (17), para comunicación con un dispositivo reproductor (2') de similares características, módulo (17) asociado a un regulador de volumen (10') y al un interruptor de corte (11'), de manera que esta especial estructuración puede sustituir al regulador de volumen (10) y al interruptor de corte (11) que se asociaban al cable (3) de conexión al reproductor (2), o bien los auriculares pueden incorporar ambos elementos, en orden a ampliar las posibilidades de conexión del dispositivo a diferentes equipos, así como permitir la escucha simultánea de señales provenientes de distintos dispositivos (2-2') a la vez que del micrófono (7), como ya se ha apuntado, pudiendo emplearse para múltiples aplicaciones.

A dicho módulo de comunicaciones inalámbricas (17) estará igualmente asociada la señal de entrada del micrófono (7), a través de una conexión (14'),en orden a que la misma pueda ser enviada de forma inalámbrica al dispositivo (2') para su procesamiento.

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REIVINDICACIONES

1ª.- Auriculares multifunción, que siendo del tipo de los que incorporan al menos un altavoz (1-1') asociado a una salida de señal de audio de un equipo reproductor (2) a través de la correspondiente línea de audio (3), se caracteriza porque incluye un circuito de control (5) en el que se establece un mezclador (6) de señales de audio, mezclador (6) cuya entrada está asociada a la citada línea de audio (3) proveniente del equipo reproductor (2) y a al menos un micrófono (7) convenientemente dispuesto para captar el sonido ambiental, con su correspondiente etapa de amplificación (8), mezclador de audio (6) a cuya salida se establecen el citado altavoz o altavoces (1-1') reproductores de ambas señales.

2ª.- Auriculares multifunción, según reivindicación 1ª, caracterizados porque el circuito de control (5) cuenta con interruptores de corte (11-12) de inhabilitación de la señal proviniente del micrófono (7) y del equipo reproductor (2), un regulador de volumen (10), así como un software interno de programación de la auto-nivelación del volumen de ambas señales.

3ª.- Auriculares multifunción, según reivindicaciones anteriores, caracterizado porque el circuito de control (5) cuenta con una línea (14) de audio auxiliar que comunica la señal amplificada del micrófono (7) con el equipo reproductor (2).

4ª.- Mejoras introducidas en la patente de invención P200603276, consistente en unos auriculares multifunción, en los que participa al menos un altavoz (1-1'), que se integra en la carcasa del correspondiente auricular, asociado a un circuito de control (5), dotado de mandos de regulación del nivel de volumen (10), circuito en el que se integra un mezclador de señales de audio (6), una de las cuales proviene de un micrófono (7) asociado a un amplificador

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(8) y al correspondiente interruptor de corte (12), caracterizadas porque el circuito de control (5) incorpora una fuente de alimentación independiente (16), tal como una pila, batería o similar.

5ª.- Mejoras introducidas en la patente de invención P200603276, según reivindicación 1ª, caracterizadas porque el circuito de control (5) incorpora adicionalmente o en sustitución del cable de conexión (3) de los auriculares a un equipo reproductor de audio (2), un módulo de comunicaciones inalámbricas (17) para recepción de señales de audio provenientes de equipos (2') compatibles con esta tecnología.

6ª.- Mejoras introducidas en la patente de invención P200603276, según reivindicaciones anteriores, caracterizadas porque en el caso de que los auriculares incorporen tanto el cable de conexión (3) como el módulo de comunicaciones inalámbricas (17) asociados al mezclador (6), cada uno de estos elementos dispondrá de su correspondiente interruptor de corte (11-11').

7ª.- Mejoras introducidas en la patente de invención P200603276, según reivindicaciones anteriores, caracterizadas porque la señal de audio proveniente del micrófono (7) es transmitida a través del módulo de comunicaciones (17) mediante una conexión (14').

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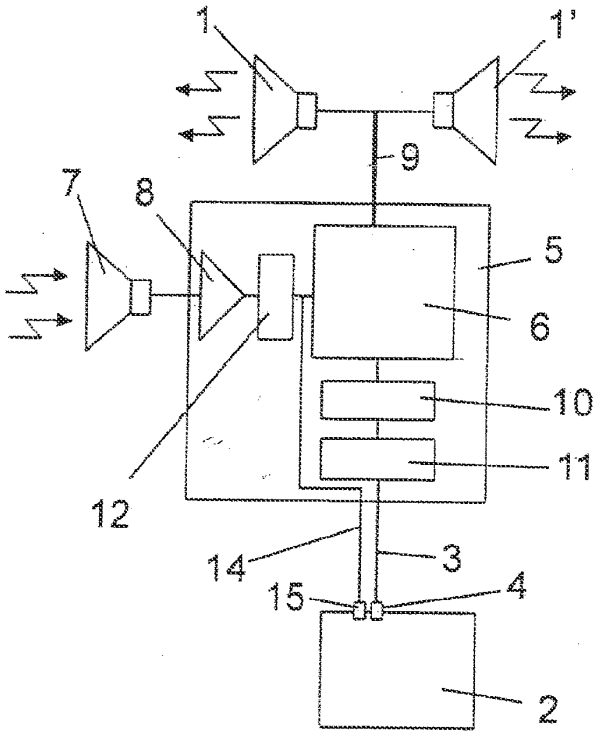


FIG. 1

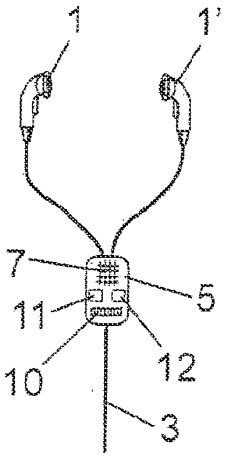


FIG. 2

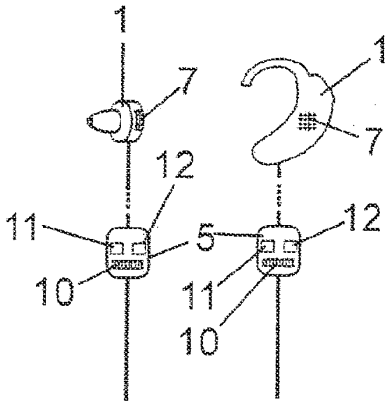


FIG. 3

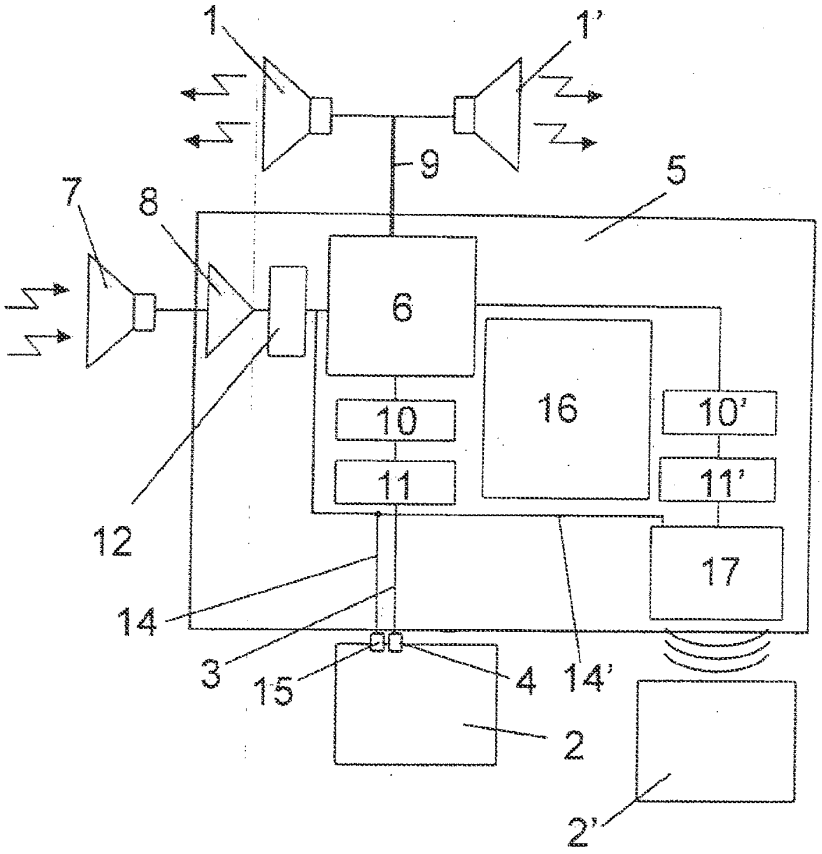


FIG. 4

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/ ES 2007/000765

A. CLASSIFICATION OF SUBJECT MATTER

see extra sheet

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

CIBEPAT,EPODOC,WPI: headphone+, headset, earphone+, ambient, microphone+, mix+, stereo, channel, AGC, noise, filter, compensat+, gain control

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 2003035551 A1 (LIGHT et al.) 20.02.2003 Abstract; figures 2, 5 and 7; paragraphs 12, 13, 16, 23 and 28.	1-4
X	US 2006182287 A1 (SCHULEIN et al.) 17.08.2006, Paragraphs 7 a 11, 14, 19, 26, 57 and 58.	1-4

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance.

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure use, exhibition, or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other documents , such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
18.April.2008 (18.04.2008)	**(30/05/2008)**
Name and mailing address of the ISA/ O.E.P.M. Paseo de la Castellana, 75 28071 Madrid, España. Facsimile No. 34 91 3495304	Authorized officer A. Catalina Gallego Telephone No. +34 91 349 85 51

Form PCT/ISA/210 (second sheet) (April 2007)

INTERNATIONAL SEARCH REPORT

International application No.

PCT/ ES 2007/000765

CLASSIFICATION OF SUBJECT MATTER

H04R 5/033 (2006.01)
H04R 5/04 (2006.01)
H04R 5/00 (2006.01)
H04R 3/00 (2006.01)

INFORME DE BÚSQUEDA INTERNACIONAL

Solicitud internacional nº
PCT/ ES 2007/000765

A. CLASIFICACIÓN DEL OBJETO DE LA SOLICITUD

Ver hoja adicional
De acuerdo con la Clasificación Internacional de Patentes (CIP) o según la clasificación nacional y CIP.

B. SECTORES COMPRENDIDOS POR LA BÚSQUEDA

Documentación mínima buscada (sistema de clasificación seguido de los símbolos de clasificación)
H04R

Otra documentación consultada, además de la documentación mínima, en la medida en que tales documentos formen parte de los sectores comprendidos por la búsqueda

Bases de datos electrónicas consultadas durante la búsqueda internacional (nombre de la base de datos y, si es posible, términos de búsqueda utilizados)
CIBEPAT,EPODOC,WPI: headphone+, headset, carphone+, ambient, microphone+, mix+, stereo, channel, AGC, noise, filter, compensat+, gain control

C. DOCUMENTOS CONSIDERADOS RELEVANTES

Categoría*	Documentos citados, con indicación, si procede, de las partes relevantes	Relevante para las reivindicaciones nº
X	US 2003035551 A1 (LIGHT et al.) 20.02.2003 Resumen; figuras 2, 5 y 7; párrafos 12, 13, 16, 23 y 28.	1-4
X	US 2006182287 A1 (SCHULEIN et al.) 17.08.2006, Párrafos 7 a 11, 14, 19, 26, 57 y 58.	1-4

☐ En la continuación del Recuadro C se relacionan otros documentos ☐ Los documentos de familias de patentes se indican en el Anexo

* Categorías especiales de documentos citados:

"A" documento que define el estado general de la técnica no considerado como particularmente relevante.

"E" solicitud de patente o patente anterior pero publicada en la fecha de presentación internacional o en fecha posterior.

"L" documento que puede plantear dudas sobre una reivindicación de prioridad o que se cita para determinar la fecha de publicación de otra cita o por una razón especial (como la indicada).

"O" documento que se refiere a una divulgación oral, a una utilización, a una exposición o a cualquier otro medio.

"P" documento publicado antes de la fecha de presentación internacional pero con posterioridad a la fecha de prioridad reivindicada.

"T" documento ulterior publicado con posterioridad a la fecha de presentación internacional o de prioridad que no pertenece al estado de la técnica pertinente pero que se cita por permitir la comprensión del principio o teoría que constituye la base de la invención.

"X" documento particularmente relevante; la invención reivindicada no puede considerarse nueva o que implique una actividad inventiva por referencia al documento aisladamente considerado.

"Y" documento particularmente relevante; la invención reivindicada no puede considerarse que implique una actividad inventiva cuando el documento se asocia a otro u otros documentos de la misma naturaleza, cuya combinación resulta evidente para un experto en la materia.

"&" documento que forma parte de la misma familia de patentes.

Fecha en que se ha concluido efectivamente la búsqueda internacional.	Fecha de expedición del informe de búsqueda internacional
18.Abril.2008 (18.04.2008)	**30 de Mayo de 2008 (30/05/2008)**
Nombre y dirección postal de la Administración encargada de la búsqueda internacional O.E.P.M. Paseo de la Castellana, 75 28071 Madrid, España. Nº de fax 34 91 3495304	Funcionario autorizado A. Catalina Gallego Nº de teléfono +34 91 349 85 51

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INFORME DE BÚSQUEDA INTERNACIONAL

Solicitud internacional nº

PCT/ ES 2007/000765

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H04R 5/033 (2006.01)
H04R 5/04 (2006.01)
H04R 5/00 (2006.01)
H04R 3/00 (2006.01)

Specially designed to enable the user to listen simultaneously to the audio signal coming from a reproduction device (2), and to ambient sound, the headphones are formed from standard speakers (1-1') associated with a control circuit (5) which includes an audio signals mixer (6), the input of which is associated with the audio line (3) coming from the reproduction equipment (2) and at least one microphone suitably placed to capture the ambient sound, including the corresponding amplification stage (8), in such a way that both signals are simultaneously reproduced through said speakers (1-1'). The control circuit is programmed to auto-level the audio signals, and has activation/disablement means (11-12) for the same. The improvements consist in providing the headphones with an internal supply source (16) which converts the device into an autonomous element, which can be used independently of the reproduction equipment (2-2'), for example as an audiphone. Another improvement consists in integrating in the control circuit (5) a wireless communications module (17), which can replace the cable of connection (3) to the reproduction equipment (2) or form an additional element, which extends the connection options for the device, although in this case the headphones have been designed to simultaneously receive several audio signals from different devices (2-2') which may be activated or disabled by means of the respective connection switches (11-11').

[Translation]

[0001] MULTIFUNCTION HEADPHONES

[0002] DESCRIPTION

[0003] OBJECT OF THE INVENTION

[0004] The present invention relates to headphones, intended to allow listening to audio signals coming from audio signal reproduction equipment such as computers, radios, mp3's, "walkman". and the like.

[0005] The object of the invention is to provide headphones that allow the user to simultaneously listen to both the audio signal coming from the player in question, as well as the ambient sound of the surroundings where the user is.

[0006] It is also an object of the invention that the device allows the input volume of one and the other signal to be adjusted at will, to the point of momentarily inhibiting either of the two.

[0007] The invention is especially applicable in the field of video calls or video conferences, however, different devices such as those mentioned above can be applied.

[0008] The improvements of the invention are intended for the device to be able to amplify the audio signals corresponding to the ambient sound without depending on the audio player device to which it is normally connected.

[0009] SUBSTITUTE SHEET (Rogta 26)

It is also the object of the improvements- to provide the headphones with a wireless communications system with the aforementioned audio player, in order to provide a more comfortable device to carry, avoiding the inconvenience that dependence on cables entails, or in its case, allowing to control the audio input of a wireless device and a device to which it is connected through the corresponding cable, simultaneously.

[0010] BACKGROUND OF THE INVENTION

[0011] Headphones are often used when listening to the radio, music, etc. through playing devices both fixed and portable, such as CD's, MP3 players . radios, mini-disc, computers, stereos, etc.

[0012] The problem with this type of device is that it has a configuration that isolates the user's ear from external ambient sound, a property that may not be desirable in many cases, such as in the case of meetings in which multiple people participate, and in which some of them use headphones to make video-conferences, listen to translations of some participant, etc. which forces to be continuously taking off and putting on the aforementioned headphones, so that when the user has them on, he is not able to follow the conversations that are taking place around him.

[0013] On the other hand, and in the field of listening to acoustic signals coming from portable playback devices, the aforementioned isolation of the user from the sounds of his environment can sometimes be dangerous, as he does not notice any alarm or warning of a

[0014] HCJA DE REPLACEMENT (Rule £ Vj

sound.

[0015] Trying to avoid this problem, in certain types of earphones in which the left earphone is physically independent from the right earphone, it is feasible that its user adapts only one of the two earphones in order to be aware of the proximal ambient sound, but this solution it prevents it from being able to listen to the player's audio signal in stereo, significantly reducing its quality, which is why it is an unsatisfactory solution.

[0016]In the Patent of Invention P20G603276, headphones specially designed to allow the user to listen simultaneously to the audio signal coming from a reproducing device, as well as the ambient sound that surrounds him, are described, so that the headphones are made from the classic loudspeakers,

associated with a control circuit in which an audio signal mixer is established, a mixer whose input is associated with the audio line coming from the player equipment and at least one microphone conveniently arranged to capture the ambient sound, including the corresponding amplification stage, so that both signals are reproduced simultaneously through the aforementioned loudspeakers.
[0017] Said control circuit is programmed to self-level the audio signals, although its user, through the corresponding cut-off switches, can enable/ disable one or another signal manually.
[0018] The problem with this device is that it is electrically dependent on the player device in question, so that it cannot be used independently, such as as a
[0019] SUBSTITUTE SHEET (Rule 20)
'Hearing aid, to amplify audio signals from the environment.
[0020] Similarly, the headphones are related to the player device. that it is through a cable, which can sometimes be annoying, limiting the user's movements.
[0021] DESCRIPTION OF THE INVENTION
[0022] 10
[0023] The headphones that the invention proposes fully satisfactorily resolve the aforementioned problems, allowing simultaneous listening to an audio signal coming from a player parallel to the audio signal corresponding to the listening environment.
[0024] 15 the surroundings of the device.
[0025] For this, and more specifically, it has been planned that, starting from the classic structure of conventional headphones, whatever type they may be, the speaker or speakers are associated with an electronic circuit, equipped with Q a signal mixer, circuit to the that accesses on the one hand the sound signal coming from the reproducing equipment in question through the corresponding line, and on the other the signal coming from at least one microphone
[0026] 'intended to receive the sounds that are emitted in the vicinity of the device.
[0027] 25
[0028] According to another of the characteristics of the invention, the aforementioned control circuit will have means for regulating the volume as well as for activating/deactivating each one of the signals.
[0029] ' iθ Likewise, the control circuit will be programmed to perform the
[0030] SUBSTITUTE SHEET (Rule 20)
self-leveling of the volume of the headphones, in such a way that said circuit is responsible for eliminating external sounds that do not exceed a pre-established decibel value as the level of activation of the external signal and self-levelling, also regulating the volume level of the signal of audio coming from the reproducing device, restoring the original level progressively at the end of the external audio signal coming from the microphone.
[0031] Finally, it should be noted that the microphone or microphones may be integrated into the housing in which each speaker is integrated, or independently, so that said microphone can be used for use in video-conferences, video-calls and similar, so that the control circuit incorporates an input line for transmission of the signal to the equipment in question. 5 In this way, a device is achieved that allows the sound coming from an audio player to be listened to together, as well as the sound coming from outside, comfortable, simple to use and totally automatic.
[0032]The improvements 'proposed by the invention solve in a fully

satisfactory way the previously exposed problem, being able to constitute an element physically and energetically independent of the reproducing device in question, so that it can be used exclusively as a hearing aid, or be associated with more of a player device, through wireless and/or wired connections, whose signals can also be enabled/disabled manually through the corresponding cut-off switches.
[0033] For this, it has been foreseen that, based on the basic structure described in the invention patent P200603276. the control circuit incorporates an internal power supply, which allows the headphones to work
[0034] SUBSTITUTE SHEET (Rule 26)
independently of the reproducing device, so that, as has been pointed out previously, the headphones can be used simply to amplify the audio signals corresponding to the ambient sound that surrounds the user, as a hearing aid.
[0035] According to another of the improvements provided for the invention , it has been planned that said control circuit incorporates a wireless data transmission module, either by radio frequencies or by infrared, replacing the connection cable between the headphones and the device. player, although the headphones could have both options, in order to be compatible with a greater number of devices as well as to allow the simultaneous transfer of audio signals with two or more devices.
[0036] In this way, the control circuit or mixer could combine more than two audio signals. including the corresponding cut-off switches. so that, for example, it could be used simultaneously to listen to an audio signal coming from a conventional player, connected to the headphones, for example through the cable, the ambient sound through the microphone associated with the headphones, and the signal coming from a third player device, such as a wireless element equipped with a microphone, such as those used for installation in rooms where babies are housed, in order to allow what is happening in said room to be heard.
[0037] As is evident, the functions provided for in patent P200603276- between the headphones and the player can also be carried out through the wireless communications module, such as the recording of the audio signal captured by the microphone associated with said headphones, or its reception for the use of the device in devices that allow video calls. as well as any other application.
[0038] SUBSTITUTE SHEET (Rule 26)
- In this way, an enormously versatile device is achieved, which allows multiple audio signals to be mixed and regulated. being able to also constitute a physically independent and autonomous element .
[0039] DESCRIPTION OF THE DRAWINGS
[0040] To complement the description that is being made and in order to help a better understanding of the characteristics of the invention. In accordance with a preferred example of a practical embodiment thereof, a set of drawings is attached as an integral part of said description, where the following has been represented for illustrative and non-limiting purposes:
[0041] Figure L- Shows a schematic representation of the different components that participate in multifunction headphones made according to the object of the invention.
[0042] Figure 2.- Shows a perspective view of a possible practical embodiment of the invention.
[0043] Figure 3.- Shows a perspective view of two different practical embodiments of the invention, in which the device is embodied in a single headset.
[0044]To complement the description that is being made and in order to help a better understanding of the characteristics of the invention,

according to an example of its practical embodiment, a single sheet of plans is attached as an integral part of said description. that with an illustrative and non-limiting character and in its only figure, has been represented in

[0045] SUBSTITUTE SHEET (Rule 26)

schematically a block diagram of the headphones in which the different improvements object of the invention can be observed

[0046] PREFERRED EMBODIMENT OF THE INVENTION

[0047] In view of the figures outlined and especially of figure 1, it can be seen how a pair of loudspeakers (1-1') participate, as is conventional, in the headphones of the invention. which are integrated into the housing of the corresponding headset, speakers associated with an audio player (2) , through a cable or audio line (3) with the complementary connector (4).

[0048] Well, based on the conventional structure described above, the invention focuses its characteristics on the fact that between the loudspeakers (1-1 ") and the audio line (3), a control circuit (5) is established in which an audio signal mixer (6) is integrated.

[0049] The signals from the audio line (3) of the player (2) flow into the aforementioned mixer. as well as the signal coming from at least one microphone (7) after amplification (8) of the signal.

[0050] Thus, the circuit at the output (9) mixes both signals according to the internal programming parameters, in such a way that the user would hear both signals if they did not have headphones and were listening to said audio signal through one or more speakers. conventional.

[0051] In addition, and as has been said previously, the control circuit (4) will be programmed to automatically carry out the self-leveling of the signals, incorporating internal filters that eliminate what

[0052] SUBSTITUTE SHEET (Rule 26)

could be considered as environmental noise, made up of sounds below a pre-established minimum level,

[0053] The control circuit also has means (10) for regulating the volume of the audio signal coming from the reproducing equipment, as well as cut-off switches (11) and (12) for disabling one or another signal entering the mixer (6) as deemed appropriate.

[0054] The microphone (7) can be integrated into the casing (7-7') of the headset, as shown in the embodiments shown in Figure 3, or it can be established in any other area of the headset, such as at the height of the neck. of the user, as shown in figure 2, being able to use said microphone to capture the user's voice signal, in which case the control circuit (5) will have an input line (14) with its corresponding connector (15) for signal processing in the equipment (2) in question.

[0055] In view of the outlined figure, it can be seen how the improvements that are recommended paiten of the basic structure of patent P200603276 in which they participate, a pair of loudspeakers (1-T). which are integrated into the casing of the corresponding headset, associated with a control circuit (5), equipped with volume level adjustment knobs (10). circuit in which an audio signal mixer (or) is integrated, one of which comes from a microphone (7) associated with an amplifier (8) and the corresponding cutoff switch (12).

[0056] Well, based on the structure described above, one of the improvements consists of incorporating a power supply (16) associated with the control circuit (5). such as a battery, battery or similar, in order to provide the device with an autonomous character, independent of the player or

[0057] SUBSTITUTE SHEET (Real 9K\

players (2-2 :) to which it may be associated.

[0058]As previously mentioned, and in accordance with another of the

improvements of the invention, the control circuit can incorporate a wireless data_transmission module (17), for communication with a player device (2). ')-of simi!ares ~ features. module (17) associated with a volume regulator (10') and a cut-off switch (1 1'). so that this special structuring can replace the volume regulator (10) and the cut-off switch (11) that were associated with the cable (3) connecting to the player (2). or the headphones can incorporate both elements, in order to expand the possibilities of connecting the device to different equipment, as well as allowing the simultaneous listening of signals coming from different devices (2-2') as well as from the microphone (7) . As has already been pointed out, it can be used for multiple applications.

[0059] The input signal of the microphone (7) will also be associated with said wireless communications module (17), through a connection i 14'). so that it can be sent wirelessly to the device (2'). for processing.

[0060] SUBSTITUTE SHEET (Rule 26)

[Translation]

R E I V I N D I C A T I O N S

Ia.- Multi-function headphones, which being of the type that incorporate at least one loudspeaker (1-1 ') associated with an audio signal output from a reproducing equipment (2) through the corresponding audio line (3). it is characterized in that it includes a control circuit (5) in which an audio signal mixer (6) is established. mixer (6) whose input is associated with the aforementioned audio line (3) coming from the player equipment (2) and at least one microphone (7) conveniently arranged to capture the ambient sound, with its corresponding amplification stage (8), audio mixer (6) at whose output the aforementioned loudspeaker or loudspeakers (1-T) reproducing both signals are established.

2a.- Multifunction headphones. according to claim Ia, characterized in that the control circuit (5) has cut-off switches (11-12) for disabling the signal coming from the microphone (7) and from the reproducing equipment (2). a volume regulator (10), as well as an internal software for programming the auto-leveling of the volume of both signals.

3rd.- Headphones muitifimción, according to previous claims, characterized in that the control circuit (5) has an auxiliary audio line (14) that communicates the amplified signal of the microphone (7) with the player equipment (2).

4s.- Improvements introduced in the invention patent P200603276, consisting of multifunction headphones. in which at least one speaker (1-1 ') participates. which is integrated into the casing of the corresponding headset, associated with a control circuit (5), equipped with volume level adjustment knobs (10). circuit in which an audio signal mixer (6) is integrated. one of which comes from a microphone (7) associated with an amplifier

SUBSTITUTE SHEET (Regia 26) (8) and a] corresponding cut-off switch (12), characterized in that the control circuit (5) incorporates an independent power supply (16), such as a battery, battery or the like.

5*,- Improvements introduced in the invention patent P200603276, according to -claim Ia, characterized in that the control circuit (5) additionally incorporates or replaces the connection cable (3) of the headphones to an audio player equipment (2). a wireless communications module (17) for receiving audio signals from equipment (T) compatible with this technology .

6«.- Improvements introduced in the invention patent P2Ü0603276, according to previous reiλ'indications, characterized in that in the event that the headphones incorporate both the connection cable (3) and the wireless communications module (17) associated with the mixer (6), each of these elements will have its corresponding cut-off switch (1 1-11 ').

7s. - Improvements introduced in the invention patent P200603276. according to previous claims, characterized in that the audio signal from the microphone (7) is transmitted through the communications module (17) through a connection (14").

SUBSTITUTE SHEET (Rule 26)

Espacenet

KR20080111004A HEADSET WITH AMBIENT SOUND

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Inventors: BOERSMA JOHAN [NL]

Classifications:
IPC ***H04B1/40;***

CPC **H04B1/40 (KR); H04M1/05 (EP,US); H04M1/6058 (EP,US); H04R1/10 (EP,US); H04R1/1041 (EP,US);** H03G3/32 (EP,US); H04R1/1016 (EP,US); H04R2420/01 (EP,US); H04R2420/07 (EP,US); H04R5/033 (EP,US);

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HEADSET WITH AMBIENT SOUND

Abstract

A headset that includes an input that receives an audio signal from an electronic equipment, and at least one earpiece having an electro-acoustic speaker for converting electrical signals to audible sound. The headset further includes at least one microphone that receives ambient sound and converts the ambient sound to an ambient sound signal. In addition, the headset includes a processing circuit formed as an integral unit with the input and the at least one earpiece. The processing circuit includes circuitry that combines the audio signal from the electronic equipment with the ambient sound signal from the at least one microphone and provides the combined signals to the at least one earpiece such that the electro-acoustic speaker reproduces the ambient sound in combination with audio from the audio signal. ® KIPO & WIPO 2009

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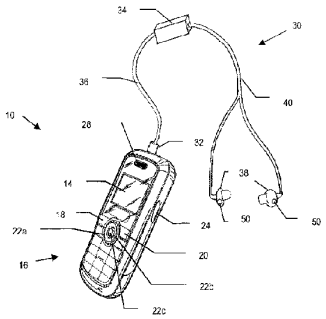
전체 청구항 수 : 총 26 항

(54) 주변 음향을 갖는 헤드셋

(57) 요 약

헤드셋은 전자 장비로부터의 오디오 신호를 수신하는 입력부, 및 전기 신호를 가청 음향으로 변환시키기 위한 전자-어쿠스틱 스피커를 갖는 적어도 하나의 이어피스를 포함한다. 헤드셋은 또한 주변 음향을 수신하고, 그 주변 음향을 주변 음향 신호로 변환하는 적어도 하나의 마이크로폰을 포함한다. 게다가, 헤드셋은 입력부 및 적어도 하나의 이어피스를 갖는 통합 유닛으로서 형성된 프로세싱 회로를 포함한다. 프로세싱 회로는 전자 장비로부터의 오디오 신호를 적어도 하나의 마이크로폰으로부터 주변 음향 신호와 결합하고, 결합된 신호를 적어도 하나의 이어피스에 제공하여 전자-어쿠스틱 스피커가 주변 음향과 오디오 신호로부터의 오디오를 결합하여 재생하도록 하는 회로를 포함한다.

대 표 도 - 도1



특허청구의 범위

청구항 1

오디오를 재생하는 헤드셋으로서,

전자 장비로부터 오디오 신호를 수신하는 입력부와,

전기 신호를 가청의 음향으로 변환하기 위한 전자-어쿠스틱(electro-acoustic) 스피커를 갖는 적어도 하나의 이어피스(earpiece)와,

주변 음향을 수신하고 주변 음향을 주변 음향 신호로 변환하는 적어도 하나의 마이크로폰과,

상기 입력부 및 적어도 하나의 상기 이어피스와의 통합 유닛으로 형성되는 프로세싱 회로 - 상기 프로세싱 회로는, 상기 전자 장비로부터의 오디오 신호를 적어도 하나의 상기 마이크로폰으로부터의 주변 음향 신호와 결합하고, 결합된 신호를 적어도 하나의 상기 이어피스에 제공하여, 전자-어쿠스틱 스피커가 오디오 신호로부터의 오디오와 결합된 상기 주변 음향을 재생하도록 하는 회로를 포함함 - 를 포함하는 오디오 재생용 헤드셋.

청구항 2

제1항에 있어서, 상기 입력부는 상기 전자 장비에 직접적으로 접속되도록 구성된 전기 커넥터를 포함하는 오디오 재생용 헤드셋.

청구항 3

제2항에 있어서, 상기 전기 커넥터는 오디오 플러그 형식의 커넥터를 포함하는 오디오 재생용 헤드셋.

청구항 4

제2항에 있어서, 상기 프로세싱 회로는 제1 오디오 케이블을 통해 상기 전기 커넥터에 연결되고 제2 오디오 케이블을 통해 적어도 하나의 상기 이어피스에 연결되며, 이들 모두는 통합 유닛의 일부분인 오디오 재생용 헤드셋.

청구항 5

제2항에 있어서, 상기 전자 장비는 이동 전화기인 오디오 재생용 헤드셋.

청구항 6

제1항에 있어서, 상기 입력부는 상기 전자 장비에 포함된 인터페이스에 통신하는 저전력 무선 인터페이스를 포함하는 오디오 재생용 헤드셋.

청구항 7

제6항에 있어서, 상기의 저전력 무선 인터페이스는 블루투스 인터페이스인 오디오 재생용 헤드셋.

청구항 8

제6항에 있어서, 상기 전자 장비는 이동 전화기인 오디오 재생용 헤드셋.

청구항 9

제1항에 있어서, 상기 헤드셋은 상기 오디오 신호 및 상기 주변 음향으로부터 오디오를 재생하기 위한 전자-어쿠스틱 스피커를 각각 포함하는 한 쌍의 이어피스를 포함하는 오디오 재생용 헤드셋.

청구항 10

제9항에 있어서, 상기 헤드셋은 한 쌍의 상기 이어피스의 각각에 포함된 각각의 한 쌍의 마이크로폰을 포함하는 오디오 재생용 헤드셋.

청구항 11

제1항에 있어서, 적어도 하나의 상기 이어피스는 이어버드(earbud) 스타일의 이어피스인 오디오 재생용 헤드셋.

청구항 12

제11항에 있어서, 상기 이어버드 스타일의 이어피스는 적어도 하나의 마이크로폰을 포함하는 오디오 재생용 헤드셋.

청구항 13

제12항에 있어서, 상기 이어버드 스타일의 이어피스는 사용자의 귀에 삽입하되, 귀의 외이(auricle)에 의해 형성된 공동(cavity) 내에 상기 마이크로폰이 위치하도록 구성된 오디오 재생용 헤드셋.

청구항 14

오디오를 재생하는 헤드셋으로서,

전자 장비로부터 오디오 신호를 수신하는 입력부와,

전기 신호를 가청의 음향으로 변환하기 위한 전자-어쿠스틱 스피커를 갖는 적어도 하나의 이어피스와,

주변 음향을 수신하고 주변 음향을 주변 음향 신호로 변환하는 적어도 하나의 마이크로폰과,

상기 전자 장비로부터의 오디오 신호를 적어도 하나의 상기 마이크로폰으로부터의 주변 음향 신호와 결합하고 결합된 신호를 적어도 하나의 상기 이어피스에 제공하여, 전자-어쿠스틱 스피커가 오디오 신호로부터의 오디오와 결합된 상기 주변 음향을 재생하도록 하는 회로를 포함하는 프로세싱 회로,

를 포함하고,

적어도 하나의 상기 이어피스는 적어도 하나의 상기 마이크로폰을 포함하며, 상기 이어피스는 사용자의 귀에 삽입하되, 귀의 외이에 의해 형성된 공동 내에 마이크로폰이 위치하도록 구성되는 오디오 재생용 헤드셋.

청구항 15

제14항에 있어서, 상기 헤드셋은 각각의 전자-어쿠스틱 스피커 및 그에 상응하는 마이크로폰을 각각 포함하는 한 쌍의 이어피스를 포함하며, 각각의 상기 전자-어쿠스틱 스피커는 상응하는 상기 마이크로폰에 의해 수신되는 상기 주변 음향을 재생하는 오디오 재생용 헤드셋.

청구항 16

제14항에 있어서, 상기 프로세싱 회로는, 상기 프로세싱 회로가 상기 오디오 신호와 상기 주변 음향 신호를 결합하는 비율을 사용자가 조정가능하게 하는 혼합 회로를 포함하는 오디오 재생용 헤드셋.

청구항 17

제16항에 있어서, 상기 프로세싱 회로는 상기 결합된 신호의 볼륨을 상기 비율과는 독립적으로 사용자가 조정가능하게 하는 볼륨 조정 회로를 포함하는 오디오 재생용 헤드셋.

청구항 18

제16항에 있어서, 상기 프로세싱 회로는 상기 비율이 일정하게 되도록 입력 신호의 진폭 변화를 보상하는 게인 제어 회로를 포함하는 오디오 재생용 헤드셋.

청구항 19

제14항에 있어서, 상기 프로세싱 회로는 입력 신호를 실질적인 변경없이 적어도 하나의 상기 이어피스로 통과하도록 하는 바이패스 회로를 포함하는 오디오 재생용 헤드셋.

청구항 20

제14항에 있어서, 상기 프로세싱 회로는 상기 입력부와 적어도 하나의 상기 이어피스를 구비한 통합 유닛으로 구성되는 오디오 재생용 헤드셋.

청구항 21

오디오를 재생하는 헤드셋으로서,

전자 장비로부터 오디오 신호를 수신하는 입력부와,

전기 신호를 가청의 음향으로 변환하기 위한 전자-어쿠스틱 스피커를 갖는 적어도 하나의 이어피스와,

주변 음향을 수신하고 주변 음향을 주변 음향 신호로 변환하는 적어도 하나의 마이크로폰과,

상기 전자 장비로부터의 오디오 신호를 적어도 하나의 상기 마이크로폰으로부터의 주변 음향 신호와 결합하고, 결합된 신호를 적어도 하나의 상기 이어피스에 제공하여, 전자-어쿠스틱 스피커가 오디오 신호로부터의 오디오와 결합된 상기 주변 음향을 재생하도록 하는 회로를 포함하는 프로세싱 회로,

를 포함하고,

상기 프로세싱 회로는, 상기 프로세싱 회로가 상기 오디오 신호와 상기 주변 음향 신호를 결합하는 비율을 사용자가 조정가능하게 하는 혼합 회로를 포함하는 오디오 재생용 헤드셋.

청구항 22

제21항에 있어서, 상기 프로세싱 회로는 상기 결합된 신호의 볼륨을 상기 비율과는 독립적으로 사용자가 조정가능하게 하는 볼륨 조정 회로를 포함하는 오디오 재생용 헤드셋.

청구항 23

제21항에 있어서, 상기 프로세싱 회로는 상기 비율이 일정하게 되도록 입력 신호의 진폭 변화를 보상하는 게인 제어 회로를 포함하는 오디오 재생용 헤드셋.

청구항 24

제21항에 있어서, 상기 프로세싱 회로는 입력 신호를 실질적인 변경없이 적어도 하나의 상기 이어피스로 통과하도록 하는 바이패스 회로를 포함하는 오디오 재생용 헤드셋.

청구항 25

제21항에 있어서, 적어도 하나의 상기 이어피스는 적어도 하나의 상기 마이크로폰을 포함하며, 상기 이어피스는 사용자의 귀에 삽입하되, 귀의 외이에 의해 형성된 공동 내에 마이크로폰이 위치하도록 구성되는 오디오 재생용 헤드셋.

청구항 26

제21항에 있어서, 상기 프로세싱 회로는 상기 입력부와 적어도 하나의 상기 이어피스를 구비한 통합 유닛으로 구성되는 오디오 재생용 헤드셋.

명 세 서

기 술 분 야

<I> 본 발명은 일반적으로 전자기기에 관한 것이며, 더욱 특별히 전자기기용 헤드셋에 관한 것이다.

배 경 기 술

<2> 휴대용 전자 디바이스와 같은 전자기기는 수십 년 동안 인기가 있었지만, 그 인기는 계속해서 증가하고 있다. 많은 현대의 휴대용 전자 디바이스는 어쿠스틱(acoustic) 및/또는 비디오(video) 신호들의 기록 또는 재생용으로 적합하거나 의도된다. 예를 들면, 휴대용 CD나 DVD 플레이어, MPEG 플레이어, MP-3 플레이어 등은 매우 다양한 형식의 개인 오락을 제공한다. 오디오(audio) 및/또는 비디오 오락이든지, 입의 사용자의 취향을 만족시키는 수많은 휴대용 전자 디바이스들이 있다.

<3> 유사하게, 이동 전화기, 호출기, 커뮤니케이터(communicator), 휴대용 전자 장치들, 예를 들어, 전자 수첩, 퍼스널 디지털 어시스턴트(personal digital assistants)(PDAs), 스마트폰 등은 또한 더 대중적이 되고 있다.

Exhibit 1004
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그러한 디바이스들은 사용자가 타인과 소통하며, 동일한 디바이스 내에서 여러 번 데이터를 저장 및 조작하고, 텍스트를 작성하는 것 등을 가능하게 한다.

<4> 특별한 어플리케이션(들)에 따르면, 헤드셋은 종종 휴대용 전자 디바이스와 결합하여 사용된다. 헤드셋은 귀 내부에 위치하는 이어피스(earpiece) 하우징(귀 삽입형(in-ear) 헤드폰)이나 귀 외부에 위치하는 이어피스 하우징(귀 접촉형(over-the-ear) 헤드폰) 각각에 형성된 하나 이상의 확성기들을 일반적으로 포함한다. 헤드셋은 예를 들면, 이동 전화 사용자가 핸즈 프리 도의(hands free discussion)에 참여하도록 해준다. 미디어 플레이어의 경우에, 헤드셋을 사용하면, 사용자는 타인을 방해하지 않고 개인적인 청취 경험을 즐길 수 있다.

<5> 그러나, 종래의 헤드셋은 단점을 갖고 있다. 단독으로 또는 헤드셋에서 재생된 음악이나 다른 오디오와 결합하여 이어피스는 모든 환경음을 완전히 차단할 수 있다. 일반적으로, 재생되는 오디오의 볼륨이 높으면 높을수록, 외부의 환경음을 사용자가 짐짐 더 인식하지 못하게 될 것이다. 사용자가 이 유형의 분리를 원할 수도 있는 시기가 있는 반면, 사용자가 주변 음향을 들을 수 없는 것이 바람직하지 않거나 위험하기까지 할 수 있는 시기가 또한 있다. 예를 들면, 주행 중(in traffic) 헤드셋을 끼고 있는 사용자는 경찰이나 비상용 상비 사이렌을 들을 수 없을지도 모르며, 그에 잠재적으로 위험한 상황을 만들고 있을 수도 있다. 사회적인 상황에서, 헤드셋을 끼는 사용자들은 여전히 자신의 주변의 음향을 듣기를 원할지도 모른다.

<6> 과거의 헤드셋은 헤드셋에 의해 재생되는 오디오뿐만 아니라 주변 음향을 유지하는 것을 시도해 왔다. 그러한 헤드셋은 주변 음향을 수집하기 위하여 설계된 하나 이상의 마이크로폰들이 있는 헤드셋을 포함해 왔다. 그러면, 주변 음향은 오디오 신호와 결합되어 헤드셋에 의해 재생된다.

<7> 그럼에도 불구하고, 원하는 오디오와 함께 사용자에게 편안하고 편리한 주변 음향의 재생을 제공하는 것에 있어서는 진술한 시도들은 부족했다. 예를 들면, 종래의 헤드셋은 다양한 유형의 전자 장비에 사용되는 능력이 국한되어 왔다. 더욱이, 종래의 헤드셋은 재생된 오디오에 주변 음향을 결합하기 위한 충분한 제어를 사용자에게 제공하지 않았다. 게다가, 종래의 헤드셋은 주변 음향의 공간적 특성들을 인식하고 받아들이는 사용자의 능력을 제한했다.

<8> <발명의 개요>

<9> 종래의 헤드셋에 관련하여 전술한 단점들을 고려하여, 당 분야에서 다양한 형식의 전자 장비와 사용하기에 적합한 헤드셋이 매우 필요하다. 더욱이, 당 분야에서 주변 음향을 재생되는 오디오와 결합하는 방법을 사용자가 제어할 수 있는 헤드셋이 매우 필요하다. 게다가, 당 분야에서 주변 음향의 공간적 특성들을 보존하는 헤드셋이 매우 필요하다.

<10> 발명의 양상에 따르면, 오디오 재생용 헤드셋이 제공된다. 헤드셋은 전자 장비로부터의 오디오 신호를 수신하는 입력부, 및 전기 신호를 가청 음향으로 변환시키기 위한 전자-어쿠스틱 스피커를 갖는 적어도 하나의 이어피스를 포함한다. 헤드셋은 또한 주변 음향을 수신하고, 그 주변 음향을 주변 음향 신호로 변환하는 적어도 하나의 마이크로폰을 포함한다. 게다가, 헤드셋은 입력부 및 적어도 하나의 이어피스를 갖는 통합 유닛으로서 형성된 프로세싱 회로를 포함한다. 프로세싱 회로는 전자 장비로부터의 오디오 신호를 적어도 하나의 마이크로폰으로부터 주변 음향 신호와 결합하고, 결합된 신호를 적어도 하나의 이어피스에 제공하여 전자-어쿠스틱 스피커가 주변 음향과 오디오 신호로부터의 오디오를 결합하여 재생하도록 하는 회로를 포함한다.

<11> 특별한 양상에 따르면, 입력부는 전자 장비로 직접적으로 연결하도록 구성되는 전기 커넥터(connector)를 포함한다.

<12> 다른 양상에 따르면, 전기 커넥터는 오디오 플러그(audio plug)형식 커넥터를 포함한다.

<13> 다른 양상에 따르면, 프로세싱 회로는 제1 오디오 케이블을 통한 전기 커넥터와 제2 오디오 케이블을 통한 적어도 하나의 이어피스에 연결되며, 모두 통합 유닛의 일부분이다.

<14> 다른 양상에 따르면, 전자 장비는 이동 전화기이다.

<15> 다른 양상에 따르면, 입력부는 전자 장비에 포함된 인터페이스와 통신하는 저전력의 무선 인터페이스를 포함한다.

<16> 다른 양상에 따르면, 저전력의 무선 인터페이스는 블루투스 인터페이스이다.

<17> 다른 양상에 따르면, 전자 장비는 이동 전화기이다.

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<18> 다른 양상에 따르면, 헤드셋은 오디오 신호 및 주변 음향으로부터의 오디오를 재생하기 위한 전자-어쿠스틱 스피커를 각각 포함하는 한 쌍의 이어피스를 포함한다.

<19> 다른 양상에 따르면, 헤드셋은 한 쌍의 이어피스 중 각각의 것에 포함된 한 쌍의 마이크로폰을 각각 포함한다.

<20> 다른 양상에 따르면, 적어도 하나의 이어피스는 이어버드-스타일(earbud-style)의 이어피스다.

<21> 다른 양상에 따르면, 이어버드-스타일 이어피스는 적어도 하나의 마이크로폰을 포함한다.

<22> 다른 양상에 따르면, 이어버드 스타일의 이어피스는 사용자의 귀에 삽입하되, 귀의 외이(auricle)에 의해 형성된 공동(cavity) 내에 마이크로폰이 위치하도록 구성된다.

<23> 다른 양상에 따르면, 오디오 재생용 헤드셋은 다시 제공된다. 헤드셋은 전자 장비로부터 오디오 신호를 수신하는 입력부, 전기 신호를 가청의 음향으로 변환시키기 위해 전자-어쿠스틱 스피커를 구비한 적어도 하나의 이어피스, 및 주변 음향을 수신하고 주변 음향을 주변 음향 신호로 변환시키는 적어도 하나의 마이크로폰을 포함한다. 헤드셋은 또한 프로세싱 회로 - 상기 프로세싱 회로는, 전자 장비로부터의 오디오 신호를 적어도 하나의 마이크로폰으로부터의 주변 음향 신호와 결합하고, 결합된 신호를 적어도 하나의 상기 이어피스에 제공하여, 전자-어쿠스틱 스피커가 오디오 신호로부터의 오디오와 결합된 상기 주변 음향을 재생하도록 하는 회로를 포함함 - 를 포함한다. 적어도 하나의 이어피스는 적어도 하나의 마이크로폰을 포함하며, 이어피스는 사용자의 귀에 삽입하되, 귀의 외이에 의해 형성된 공동 내에 마이크로폰이 위치하도록 구성된다.

<24> 다른 양상에 따르면, 상기 헤드셋은 각각의 전자-어쿠스틱 스피커 및 그에 상응하는 마이크로폰을 각각 포함하는 한 쌍의 이어피스를 포함하며, 각각의 상기 전자-어쿠스틱 스피커는 상응하는 상기 마이크로폰에 의해 수신되는 상기 주변 음향을 재생한다.

<25> 다른 양상에 따르면, 상기 프로세싱 회로는, 상기 프로세싱 회로가 상기 오디오 신호와 상기 주변 음향 신호를 결합하는 비율을 사용자가 조정가능하게 하는 혼합 회로를 포함한다.

<26> 다른 양상에 따르면, 상기 프로세싱 회로는 상기 결합된 신호의 볼륨을 상기 비율과는 독립적으로 사용자가 조정가능하게 하는 볼륨 조정 회로를 포함한다.

<27> 다른 양상에 따르면, 상기 프로세싱 회로는 상기 비율이 일정하게 되도록 입력 신호의 진폭 변화를 보상하는 게인 제어 회로를 포함한다.

<28> 다른 양상에 따르면, 상기 프로세싱 회로는 입력 신호를 실질적인 변경없이 적어도 하나의 상기 이어피스로 통과하도록 하는 바이패스 회로를 포함한다.

<29> 다른 양상에 따르면, 상기 프로세싱 회로는 상기 입력부와 적어도 하나의 상기 이어피스를 구비한 통합 유닛으로 구성된다.

<30> 본 발명의 다른 양상에 따르면, 전자 장비로부터 오디오 신호를 수신하는 입력부, 전기 신호를 가청의 음향으로 변환시키기 위해 전자-어쿠스틱 스피커를 구비한 적어도 하나의 이어피스를 포함하는 오디오 재생용 헤드셋은 제공된다. 헤드셋은 또한 주변 음향을 수신하고 주변 음향을 주변 음향 신호로 변환시키는 적어도 하나의 마이크로폰을 포함한다. 프로세싱 회로는 전자 장비로부터의 오디오 신호를 적어도 하나의 마이크로폰으로부터의 주변 음향 신호와 결합하고 결합된 신호를 적어도 하나의 이어피스에 제공하여, 전자-어쿠스틱 스피커가 오디오 신호로부터의 오디오와 결합된 주변 음향을 재생하도록 하는 회로를 포함한다. 상기 프로세싱 회로는, 상기 프로세싱 회로가 상기 오디오 신호와 상기 주변 음향 신호를 결합하는 비율을 사용자가 조정가능하게 하는 혼합 회로를 포함한다.

<31> 다른 양상에 따르면, 상기 프로세싱 회로는 상기 결합된 신호의 볼륨을 상기 비율과는 독립적으로 사용자가 조정가능하게 하는 볼륨 조정 회로를 포함한다.

<32> 다른 양상에 따르면, 프로세싱 회로는 비율이 상수로 남는 입력부 신호의 진폭의 변화를 보상하는 게인 제어 회로를 포함한다.

<33> 다른 양상에 따르면, 상기 프로세싱 회로는 입력 신호를 실질적인 변경없이 적어도 하나의 상기 이어피스로 통과하도록 하는 바이패스 회로를 포함한다.

<34> 다른 양상에 따르면, 적어도 하나의 상기 이어피스는 적어도 하나의 상기 마이크로폰을 포함하며, 상기 이어피스는 사용자의 귀에 삽입하되, 귀의 외이에 의해 형성된 공동 내에 마이크로폰이 위치하도록 구성된다.

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<35> 다른 양상에 따르면, 상기 프로세싱 회로는 상기 입력부와 적어도 하나의 상기 이어피스를 구비한 통합 유닛으로 구성된다.

<36> 본 발명은, 전술하고 관련된 목적의 달성을 위하여, 본 명세서에서 청구 사항들에 충분히 후술되고 특별히 지적된 특징들을 포함한다. 다음의 기술 및 부가된 도면들은 발명의 명백하고 구체적인 실시예들을 상세히 설명한다. 그러나, 이 실시예들은 상기 발명의 원리가 사용될 수도 있는 몇몇의 다양한 방법들로만 나타낸다. 발명의 다른 목적, 장점 및 새로운 특징들은 도면에 관련하여 고려될 때 본 발명의 이하의 상술로 명백해 질 것이다.

<37> 본 명세서에 사용되는 "포함한다/포함하는"("comprises/comprising")이라는 용어는 정해진 특징, 완성물, 단계 또는 요소들의 존재를 구체화하기 위한 것이지 하나 이상의 다른 특징들, 완성물들, 단계들, 요소들 또는 그룹들의 그룹들의 존재 및 부가를 배제하는 것은 아니라는 것이 강조된다.

발명의 상세한 설명

<43> 본 발명은 도면을 참조하여 설명되며, 도면 전체에 걸쳐 유사한 구성 요소에는 유사한 참조 부호가 부여된다.

<44> 여기서 언급된 바와 같은 "전자 장비"라는 용어는 디지털 음악 재생기(예를 들어, CD, MP3, AC-3, .wma 등.), 비디오 재생기(예를 들어, DVD, MPEG, .wma 등.), 사진 재생기(예를 들면, JPEG 등.) - 이에 한정되는 것은 아님 - 와 같은 다양한 형식의 미디어 재생기를 포함하는 개인 휴대용 전자제품을 포함한다. 전자 장비는 다양한 미디어 파일들을 저장용 메모리 형식으로 불리는, 매우 인기있는 "플래쉬 메모리" 및 "하드 드라이브" 형식의 미디어 플레이어 뿐만 아니라 CD와 DVD 플레이어와 같은 전통적인 미디어 플레이어도 포함한다. 전자 장비는 미디어 플레이어를 포함하는 멀티 기능의 디바이스 뿐만 아니라, 전용 미디어 플레이어도 포함한다. 그러한 멀티 기능의 디바이스는 휴대용 무선 통신 장비를 포함한다. "이동 무선 단말기"로 언급되기도 하는 "휴대용 무선 통신 장비"라는 용어는, 이동 전화기, 호출기, 커뮤니케이터, 예를 들어 전자 수첩, 퍼스널 디지털 어시스턴트(PDAs), 스마트폰 등과 같은 모든 장비들을 포함한다. 본 명세서에서, 상기 발명은 이동 전화기에 관련하여 우선적으로 기술된다. 그러나, 본 발명이 이동 전화기에 국한하려고 하는 것은 아니고 어떠한 형식의 전자 장비라도 될 수 있다는 것이 이해될 것이다. 더욱이, 본 발명은 이동 전화기와 같은 전자 장비를 통하여 음악을 듣는 것과 관련하여 우선적으로 기술되지만, 본 발명은 더 일반적으로 음향이 재생되는 어떠한 형식의 전자 장비에도 적용할 수 있다는 것이 이해될 것이다.

<45> 우선 도 1을 참조하면, 본 발명에 따른 전자 장비(10)가 도시되어 있다. 여기서 기술된 예시적 실시예에서, 전자 장비는 이동 전화기(10)이다. 이동 전화기(10)는 "벽돌" 또는 "블록" 디자인 형식의 하우징을 갖는 것으로 도시되어 있지만, 대합 또는 슬라이드 형식의 하우징과 같은 다른 형식의 하우징들을 본 발명의 범위에서 벗어나지 않고 이용할 수도 있다는 것이 이해될 것이다.

<46> 이동 전화기(10)는 디스플레이(14)와 키패드(16)를 포함한다. 종래와 같이, 디스플레이(14)는 작동 상태, 시간, 전화 번호들, 연락 정보, 사용자가 이동 전화기(10)의 다양한 특징들을 이용할 수 있도록 하는 다양한 네비게이션 메뉴들 등과 같은 정보를 사용자에게 표시한다. 유사하게, 키패드(16)는 다양한 사용자 입력 작동들을 제공한다는 점에서는 종래의 것과 동일하다. 예를 들어, 키패드(16)는 전형적으로 전화 번호, 전화 목록, 연락 정보, 주석 등과 같은 영숫자 정보를 입력할 수 있는 영숫자 키들을 포함한다. 더욱이, 키패드(16)는 통화 설기 또는 받기를 위한 "통화 발신(call send)"키(18), 및 통화 정지 또는 "끝기"를 위한 "통화 정지(call-end)"키(20)와 같은 특수한 기능 키들을 전형적으로 포함한다. 특수 기능 키들은 예를 들어, 디스플레이(14)에 디스플레이되는 메뉴를 조종하여 다른 전화 기능, 프로파일, 셋징 등을 종래와 같이 신택하도록 하는 메뉴 네비게이션(navigation) 키들 (22a), (22b) 및 (22c)를 또한 포함할 수도 있다. 키패드(16) 내에 포함된 다른 키들은 웹 브라우저 런치(web browser launch) 키, 카메라 키 등과 같은 다양한 다른 키들 뿐만 아니라 볼륨 키(24), 온/오프(on/off) 파워 키(26)도 포함할 수도 있다.

<47> 도 1의 특정 실시예에서, 이동 전화기(10)는 디스플레이(14) 및 이와 별도의 키패드(16)를 포함한다. 대안적 실시예에서, 디스플레이(14)는 자체가 하나 이상의 키들을 포함하는 터치스크린을 포함할 수도 있다. 다른 실시예에서, 디스플레이(14)는 매우 국한된 키패드(16)를 포함하거나 키패드(16)를 전혀 포함하지 않도록 하기 위해 전화기(10)를 작동하는데 사용되는 모든 또는 거의 모든 키들을 포함하는 터치스크린을 포함할 수 있다. 알 수 있는 바와 같이, 키패드(16) 또는 터치스크린/디스플레이(14)가 포함된 키의 특정 형태 및 기능은 광의의 의미에서 본 발명과 밀접한 관계가 있는 것은 아니다.

<48> 이동 전화기(10)는 전통적인 이동 전화기 기능과 더불어 다양한 기능들을 수행하는 것이 가능한 멀티 기능의 디

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바이스이다. 예를 들어, 본 발명에 따른 이동 전화기(10)는 또한 미디어 플레이어의 기능을 한다. 더 구체적으로, 이동 전화기(10)는 오디오 파일(예를 들어, MP3, .wma, AC-3), 비디오 파일(예를 들어, MPEG, .wma 등), 정지 영상(예를 들어, .pdf, JPEG, .bmp 등)과 같은 다른 형식의 미디어 객체들을 재생할 수 있다. 미디어 객체는 선형적으로 이동 전화기(10) 내의 비휘발성 메모리에 저장된다. 이동 전화기(10)는 본 발명에 따른 스피커(28) 또는 스테레오 헤드셋(30)과 같은 악세서리를 통하여 오디오 파일을 재생한다. 스테레오 헤드셋(30)은 적절한 커넥터를 통하여 이동 전화기(10)의 오디오 출력부에 플러그가 꽂아지거나 연결될 수도 있다.

<49> 후술될 이유들로 인해서, 헤드셋(30)은 오디오 출력부를 갖는 전자 장비를 사용하는 종래의 헤드셋과 관련된 전술된 단점들을 극복한다. 예를 들어, 헤드셋(30)은 다양한 형식의 전자 장비와 사용하는데 적당하다. 헤드셋(30)은 사용자가 재생되는 오디오를 듣도록 하면서도 여전히 주변 음향을 들을 수 있도록 하기 위해 프로세싱 회로에 의존하거나 전자 장비(10) 내에서 제어할 필요는 없다. 더욱이, 헤드셋(30)은 주변 음향을 재생되는 오디오에 결합하는 방식을 제어하는 기능을 사용자에게 제공한다. 사용자는 재생되는 오디오 및 주변 음향의 각 수준을 제어할 수 있기 때문에, 자신의 선호에 맞게 청취 경험을 조정할 수 있도록 해준다. 게다가, 헤드셋(30)은 주변 음향의 공간 특성을 보존한다. 그 결과, 사용자는 주변 음향이 발원하는 방향과 같은 방위를 계속해서 알 수 있다. 이 특징들은 종래의 헤드셋과 비교하여 안전성을 증가시키고 사용자의 청취 경험을 개선시킨다.

<50> 예시적 실시예에서의 헤드셋(30)은 이동 전화기(10)로부터 오디오 신호를 수신하는 입력부(32)를 포함한다. 입력부(32)는 종래의 오디오 잭/플러그(jack/plug) 또는 임의의 다른 형식의 적절한 전기 커넥터를 포함한다. 오디오 신호는 스테레오가 아닌 신호(예를 들어, 모노(mono), 멀티-채널(multi-channel)등)일 수도 있지만, 오디오 신호는 좌우 채널용 오디오를 포함하는 스테레오 신호가 바람직하다.

<51> 더욱이, 헤드셋(30)은 오디오 케이블(36)을 통해서 입력부(32)에서 오디오 신호를 수신하는 프로세서(34)를 포함한다. 게다가, 헤드셋(30)은 오디오 케이블(40)을 통해 프로세서(34)에 접속된 한 쌍의 이어피스(38)를 포함한다. 이어피스(38)가 전기 신호를 오디오 음향으로 변환하기 위한 전자-어쿠스틱 확성기(미도시됨)를 이어피스 하우징 내에 각각 포함한다는 점에서 각각의 이어피스(38)는 종래의 것과 동일하다. 프로세서(34)는 이동 전화기(10)의 출력부로부터의 좌우 채널 스테레오 신호를 각각의 이어피스(38)에 제공한다.

<52> 모범 실시예에서, 이어피스(38)는 도 2에 도시한 바와 같이 이어버드(38)와 같은 각각 "귀 삽입" 형식의 이어피스이다. 본 명세서에서 언급되는 바와 같이, "귀 삽입" 형식의 이어피스(38)는 사용자가 귀의 외이(44)에 의해 형성된 공동(42) 내에 일반적으로 착용하는 형식이다. 이어버드(38)는 공동(42) 및/또는 귀 관으로의 개구(46) 내에 아늑하게 맞는다. 반대로, "귀 접촉" 형식의 이어피스는 사용자가 외이(44)에 의해 형성된 공동(42)의 외부 또는 상부에 일반적으로 착용한다.

<53> 계속해서 도 1 및 도 2를 참조하면, 이어피스(38) 각각은 주변 음향을 모으기 위한 마이크로폰(50)을 포함한다. 마이크로폰(50)은 이어피스(38)를 사용자가 착용했을 때 마이크로폰(50)이 외이(44)에 의해 형성된 공동(42) 내에 위치하고 그 공동(42)에 노출되도록 이어피스 하우징 내에 위치한다. 따라서, 도 2의 화살표(52)로 도시한 바와 같이, 외이(44)에 의해 수집된 주변 음향 파동은 일반적으로 마이크로폰(50) 쪽으로 집중된다. 그 결과, 마이크로폰(50)은 음향의 공간 특성들을 보존하면서 주변 음향을 모은다. 즉, 인간은 귀의 공간 및 방향에 기반하여 음향이 발원하는 방향을 감지할 수 있는 것과 같이, 마이크로폰(50)은 동일한 주변 음향을 모은다. 반대로, 귀 접촉 형식의 이어피스 내에 위치한 마이크로폰은 그러한 공간 특성을 잃는다. 마이크로폰이 계속해서 이격되어 있는 한, 외이(44)에 의해 형성된 공동 내에 위치하고 있지 않기 때문에 귀의 방향과 관련한 특성은 잃게 된다.

<54> 더욱이, 도 2는 각각의 이어피스(38)는 전술한 확성기(56)를 어떻게 포함하는지를 기술한다. 확성기(56)는 종래의 형식(예를 들어, 압전 소자 등)일 수 있다. 도 2에 도시한 바와 같이, 확성기(56)는 오디오 케이블(40)을 통해 프로세서(34)로부터의 오디오 신호를 수신한다. 반대로, 마이크로폰(50)은 주변 음향을 수신하고 그러한 음향을, 오디오 케이블(40)을 통해 프로세서(34)에 제공되는 주변 음향 신호로 변환한다.

<55> 도 3 및 도 4에 따라 더 자세히 후술되는 바와 같이, 프로세서(34)는 이동 전화기(10)로부터 수신되는 오디오 신호를 각각의 마이크로폰(50)에 의해 제공되는 주변 음향 신호와 결합하는 기능을 한다. 프로세서(34)는 각각의 이어피스(38) 내의 확성기(56)에 결합된 신호를 제공한다. 그러한 방식으로, 각각의 이어피스(38)는 이동 전화기(10)로부터 오디오 신호와 함께 그에 상응하는 마이크로폰(50)에 의해 모아진 주변 음향을 재생할 수 있다. 따라서 주변 음향의 공간 특성이 보존된다.

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<56> 도 1을 다시 간략히 참조하면, 바람직한 실시예에서의 헤드셋(30)은 통합형 또는 내장형 유닛이다. 구체적으로, 입력부(32), 프로세서(34) 및 이어피스(38)는 오디오 케이블(36, 40)을 통해 다른 것에 고정적으로 연결된다. 따라서, 헤드셋(30)은 특정 전자 장비 또는 임의의 다른 악세사리 및/또는 부속물에 반드시 의존할 필요가 없는 내장형 악세사리를 나타낸다. 헤드셋(30)은 주변 음향을 획득하고 그 주변 음향을 전자 장비로부터의 오디오와 결합하여 원하는 청취 경험을 제공하는 자체의 프로세서(34)를 포함한다.

<57> 도 3을 참조하면, 프로세서(30)의 예시적 실시예가 도시되어 있다. 도 4에 따라 더 자세히 후술되는 바와 같이, 프로세서(30)는 사용자가 프로세서(34)를 온 및 오프할 수 있도록 하는 온/오프 스위치(70)를 포함한다. 온 상태에서는, 헤드셋(30)은 주변 음향을 모으고 그 주변 음향을 이동 전화기(10)로부터 오디오 신호와 결합하여 둘 다 사용자가 들을 수 있도록 하는 기능을 한다. 오프 상태에서는, 기본적으로 종래의 헤드셋과 같이 프로세서(34)는 이동 전화기로부터 오디오 신호를 직접적으로 이어피스(58)에 보낸다. 온/오프 스위치(70)는 슬라이드 스위치(slide switch)로 도시되어 있지만, 어떠한 형식의 적당한 스위치도 본 발명의 범위에서 벗어나지 않고 가상적으로 사용될 수도 있는 것이 이해될 것이다.

<58> 프로세서(34)는 볼륨 제어 스위치(72) 및 혼합 스위치(74)를 또한 포함한다. 볼륨 제어 스위치(72)는 사용자가 이동 전화기(10)의 볼륨 출력을 직접적으로 조정하지 않고서도 조정하지 않고서도 헤드셋(30)에 의해 재생되는 오디오 볼륨을 제어 가능하게 한다. 혼합 스위치(74)는 프로세서(34)가 이동 전화기(10)로부터의 오디오 신호와 마이크로폰(50)으로부터 주변 음향 신호를 결합하는 비율을 사용자가 조정 가능하게 한다. 교환 스위치를 제어함으로써, 사용자는 이어피스(38)가 오디오 신호에 관한 주변 음향을 더 또는 덜 제공하는 지의 여부를 선택할 수 있다. 상기 상황에 따르면, 사용자는 확성기(56)에 제공되는 주변 음향의 양을 실질적으로 감소시킴으로써(예를 들어, 혼잡한 기차 내), 대부분 또는 전부의 환경음들을 차단하도록 선택할 수도 있다. 사용자는 환경에 어느 정도의 접촉을 유지하는 것이 바람직한 상황(예를 들어, 운행 중에 있는 동안) 내에서, 비록 감소된 수준이긴 하지만, 낮은 수준의 주변 음향을 유지하는 것을 선택할 수도 있다. 더욱이, 사용자는 환경에 완전한 접촉을 유지하는 것이 바람직한 상황(예를 들어, 누군가의 거실에서) 하에서 보다 높은 수준의 주변 음향을 유지하는 것을 선택할 수도 있다.

<59> 도 3에 도시된 바와 같이 스위치(72, 74)는 특정 화살표를 누름에 따라 볼륨을 조정하고 위 또는 아래로 혼합하는 기능을 하는 멤브레인(membrane) 스위치이다. 그러나, 당해 분야에 능숙한 사람이라면 어떠한 다른 형식의 적당한 스위치도 본 발명의 범위에서 벗어나지 않고 가상적으로 또한 사용될 수 있었다는 것을 곧 이해할 수 있을 것이다. 예를 들어, 스위치(72) 및/또는 스위치(74)는 회전식 스위치, 슬라이드 스위치 등으로 대체될 수 있었다. 이해될 수 있는 바와 같이, 본 명세서에 기술된 임의의 또는 모든 회로에 함께 스위치(72 및/또는 74)는 아날로그 회로, 디지털 회로, 또는 그들의 임의의 조합일 수도 있다. 본 발명은 아날로그 디자인에 관하여 주로 기술되었지만, 회로는 대안으로 디지털 디자인 또는 그들의 조합을 통해 구현될 수 있었다는 것을 용이하게 이해할 수 있을 것이다.

<60> 도 4는 프로세서(34)의 개략적인 도면이다. 프로세서(34)는 단일 채널을 나타내기 위하여 단순화된다. 그러나, 당해 분야에 능숙한 사람이라면 각각의 이어피스(38)에 제공되는 좌우 채널 스테레오 신호들을 다루기 위한 구체적 실시예가 도 4에 나타난 회로의 복제를 단순히 포함한다는 것임을 이해할 수 있을 것이다.

<61> 이동 전화기(10)는 오디오 케이블(36) 내의 라인(80)을 통해 프로세서(34)에 오디오 신호를 제공한다. 라인(36) 상의 오디오 신호는 프로세서(34) 내에 포함된 가산기(82)로의 입력이다. 가산기(82)의 출력은 가산기(82)의 출력을 증폭하는 조정가능한 게인 증폭기(84)로의 입력이다. 증폭기의 출력은 오디오 케이블(40) 내의 라인(86)을 통해 이어피스(38)에 제공된다.

<62> 마이크로폰(50)에 의해 모아진 주변 음향은 오디오 케이블(40)로부터 라인(88) 상의 주변 음향 신호로써 프로세서(34)에의 입력된다. 라인(88) 상의 주변 음향 신호는 주변 음향 신호를 증폭하기 위해 조정가능한 게인 증폭기(90)에의 입력된다. 증폭기(90)는 전술한 혼합 스위치(74) 형태의 수준 조절 스위치를 사용하여 게인이 조절되는 조정가능한 게인 증폭기(91)에 주변 음향 신호를 출력한다. 이 실시예에서 가급적이면 가감저항기인 혼합 스위치(74)는 증폭기(91)로부터의, 가산기(82)의 다른 입력부에 제공되는 증폭된 주변 음향 신호의 수준을 조절한다.

<63> 따라서, 가산기(82)는 주변 음향 신호를 이동 전화기(10)로부터의 오디오 신호와 결합하는 기능을 한다. 혼합 스위치(74)를 조정함으로써, 사용자는 가산기(82)의 출력으로 표시되는 주변 음향 신호대 오디오 신호의 상대 진폭 또는 비율을 조절할 수 있다. 따라서, 사용자는 주변 음향이 오디오 신호에 비교하여 나타나는 정도를 조

절할 수 있다.

<64> 이 실시예에서 또한 바람직하게는 가감저항기인 볼륨 스위치(72)는 증폭기(84)의 게인을 조정하는 기능을 한다. 따라서, 사용자는 결합된 오디오 및 주변 음향 신호를 증폭하는 증폭기(84)의 게인을 조절할 수도 있다. 그 결과, 사용자는 헤드셋에 의해 제생되는 음향의 전체적인 볼륨을 조절할 수도 있다.

<65> 바람직한 실시예에서, 프로세서(34)는 이동 전화기(10)로부터 라인(80) 상에 수신되는 오디오 신호의 수준을 검출하는 기능을 하는 신택적 수준 검출기 회로(92)를 더 포함할 수도 있다. 수준 검출기 회로(92)는 라인(80) 상의 오디오 신호의 수준에 비례하는 라인(94) 상의 게인 제어 신호를 출력하도록 설계된다. 라인(94) 상의 게인 조절 신호는 조정가능한 게인 증폭기(90)의 게인을 조정하는 것을 제공한다. 그 결과, 프로세서(34)는 수신되는 오디오 신호의 수준에서의 변화에 비례하여 주변 음향 신호의 수준을 자동으로 조정한다. 이 방법에서, 사용자가 이동 전화기(10) 또는 다른 전자 장비의 오디오 출력 볼륨을 조정하더라도, 오디오 신호 및 주변 음향 신호의 혼합 비율은 실질적으로 일정한 상태로 있게 된다. 미도시되었지만, 수준 검출기(92)는 (예를 들어, 오디오 신호에의 갑작스러운 조용한/시끄러운 주기에 기인한 주변 음향 수준의 갑작스러운 변화를 피하기 위해) 바람직하게는 증폭기(90)의 응답을 평활화하기 위한 저역 통과 필터를 포함한다.

<66> 프로세서(34)는 남아있는 다른 회로에 전원(100)에서의 전력을 온 및 오프하는 전술한 스위치(70)를 포함한다. 전원(100)은 전지와 같이, 바람직하게는 내장형이다. 대안으로, 프로세서(34)는 전자 장비 자체에서 전력을 받도록 설계된다. 스위치(70)는 선택적 스위치(70a)의 일부분으로서 또한 포함할 수도 있다. 상기 스위치(70a)는 효과적으로 개방 회로가 되도록 작동 중에는 일반적으로 개방된다(스위치(70)은). 스위치(70)가 오프인 경우, 스위치(70a)는 닫힌다. 그 결과, 라인(80) 상의 오디오 신호는 프로세서(34) 내의 회로를 우회하고 라인(86) 상의 이어피스(38)에 직접적으로 출력된다.

<67> 더욱이, 프로세서(34)는 헤드셋(30)을 사용하여 이동 전화기(10)와 핸즈 프리 통신하는 동안 사용자의 음성을 모으는 마이크로폰(102)을 포함할 수도 있다. 마이크로폰(102)은 종래의 기술을 사용하여 라인(104) 상의 이동 전화기(10)에 음성 신호를 제공한다.

<68> 도 5는 프로세서(34)의 대체 실시예를 기술한다. 이 실시예에서, 입력부(32)는 블루투스 송수신기(106)와 같은 저전력 무선 인터페이스로 대체된다. 블루투스 송수신기(106)는 헤드셋(30)과 유사하게 갖추어진 이동 전화기(10) 또는 다른 전자 장비 간에 종래의 통신을 수행하도록 프로그래밍될 수 있다. 그러한 실시예에서, 오디오 케이블(36)은 필요하지 않다.

<69> 더욱이, 본 명세서에서 기술된 실시예들은 본 발명의 범위로부터 벗어나지 않고 변경될 수 있다. 예를 들어, 프로세서(34)를 이어피스(38)와 입력부(32) 사이에 일렬로 접속된 하우징의 일부로서 도시하고 있지만, 프로세서(34)는 입력부(32)를 형성하는 잭(jack)의 일부분 내에 및/또는 이어피스(38)의 하우징 내에 대신 설치할 수도 있다. 대안으로, 블루투스 실시예에서, 프로세서(34)는 공지된 블루투스 헤드셋과 유사하게 사용자의 귀 뒤쪽에 맞도록 설계된 작은 하우징 내에 결합될 수도 있다.

<70> 본 발명은 바람직한 실시예들을 도시하고 기술하였지만, 당해 분야에 능숙한 다른 사람들이 본 명세서를 읽고 이해함으로써 등가물 및 수정이 가능하다는 것은 자명하다. 본 발명은 그러한 모든 등가물 및 수정을 포함하며, 이후의 청구 범위에 의해서만 국한된다.

도면의 간단한 설명

<38> 도 1은 본 발명의 실시예에 따른 이동 전화와 헤드셋의 주변도이다.

<39> 도 2는 본 발명의 실시예에 따라 외이의 공동 내에 위치한 마이크로폰을 포함하는 이어버드의 단면도이다.

<40> 도 3은 본 발명의 실시예에 따른 헤드셋의 프로세서의 정면도이다.

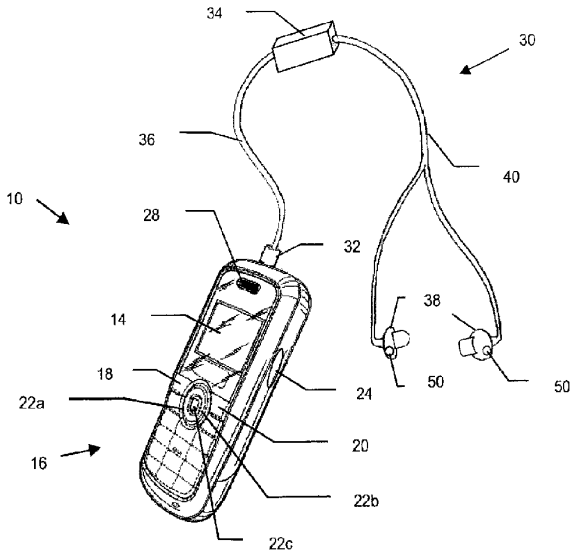
<41> 도 4는 본 발명의 실시예에 따른 도 3의 프로세서의 간략 회로 도면이다.

<42> 도 5는 본 발명의 다른 실시예에 따른 프로세서의 간략 회로 도면이다.

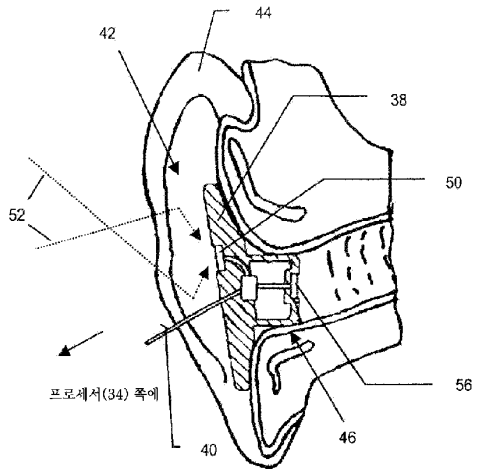
Exhibit 1004
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도면

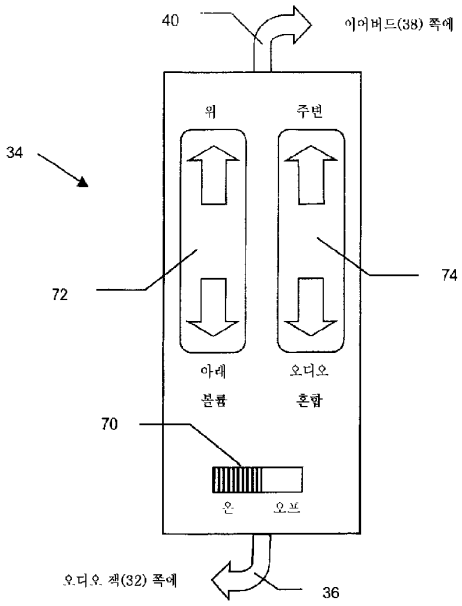
도면1


34
30
36
40
10
28
32
38
14
18
24
50
50
22a
20
16
22b
22c

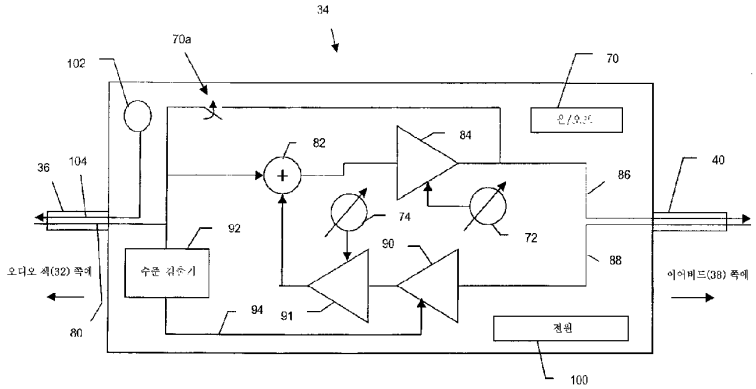
도면2


44
42
38
50
52
56
프로세서(34) 쪽에
40
46

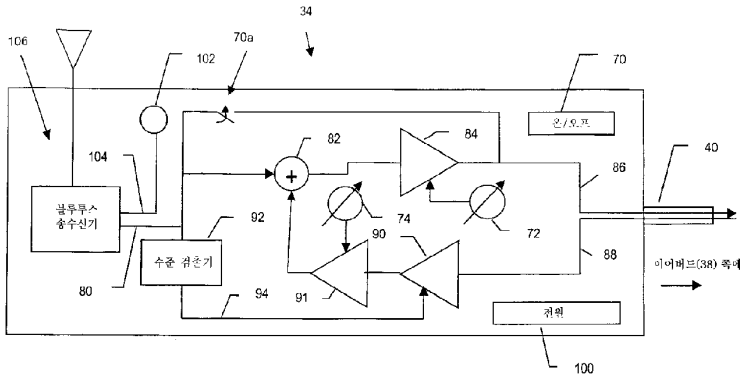
도면3


40
어어버드(38) 쪽에
위
주변
34
72
74
아래
볼륨
오디오
혼합
70
온
오프
오디오 잭(32) 쪽에
36

도면4


34
70a
102
70
82
84
36
104
40
86
74
72
92
90
88
오디오 잭(32) 쪽에
수분 검출기
어어버드(38) 쪽에
94
91
80
전원
100

도면5


34
106
70a
102
70
온/오프
82
84
40
104
86
블루투스
송수신기
92
74
90
72
88
수줄 검출기
이어버드(38) 폭에
80
94
91
권한
100

KR20080111004A HEADSET WITH AMBIENT SOUND

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

Headset with ambient sound {HEADSET WITH AMBIENT SOUND}

[0001] BACKGROUND OF THE INVENTION 1. Field of the Invention The present invention relates generally to electronic devices, and more particularly to headsets for an electronic device.

[0002] Electronic devices such as portable electronic devices have been popular for decades, but their popularity continues to grow. Many modern portable electronic devices are suitable for or intended for recording or reproducing acoustic and / or video signals. For example, portable CD or DVD players, MPEG players, and MP-3 players offer a wide variety of personal entertainment. There are a number of portable electronic devices that satisfy the taste of any user, whether audio and / or video entertainment.

[0003] Similarly, mobile phones, pagers, communicators, portable electronic devices, such as electronic organizers, personal digital assistants (PDAs), smart phones, and the like, are also becoming more popular. Such devices enable the user to communicate with others, store and manipulate data multiple times within the same device, create text, and the like.

[0004] According to a particular application (s), a headset is often used in conjunction with a portable electronic device. The headset may be one or more of the earpiece housings (ear-in-ear headphones) located within the ear or ear-piece housings (ear-ear-type headphones) Loudspeakers. The headset allows, for example, a mobile phone user to participate in a hands free discussion. In the case of a media player, using a headset allows the user to enjoy a personal listening experience without disturbing others.

[0005] However, the conventional headset has disadvantages. Combined with music or other audio played solo or on a headset, the earpiece can completely isolate all environmental sounds. Generally, the higher the volume of audio being played back, the more the user will not be able to perceive the external environment sound. There are also times when the user may desire this type of separation, while there are times when it is undesirable or even dangerous for the user not to be able to hear the ambient sound. For example, a user wearing a headset in traffic may not be able to listen to police or emergency equipment sirens, and may be creating potentially dangerous situations. In a social situation, users wearing headsets may still want to hear the sound of their surroundings.

[0006] The headset of the past has tried to maintain the ambient sound as well as the audio played by the headset. Such a headset has included a headset with one or more microphones designed to acquire ambient sound. The ambient sound is then combined with the audio signal and played back by the headset.

[0007] Nevertheless, the above-mentioned attempts have been lacking in providing comfortable and convenient reproduction of ambient sound to the user with the desired audio. For example, conventional headsets have been limited in their ability to be used in various types of electronic equipment. Moreover, the conventional headset did not provide the user with sufficient control to combine ambient sound with the reproduced audio. In addition, conventional headsets have limited the user's ability to recognize and accept spatial characteristics of ambient sound.

[0008] SUMMARY OF THE INVENTION [

[0009] In view of the above-mentioned disadvantages associated with conventional headsets, there is a great need in the art for headsets that are suitable for use with various types of electronic equipment. Moreover, there is a great need in the art for a headset that allows the user to control how to combine ambient sound with reproduced audio. In addition, there is a great need in the art for a headset that preserves the spatial properties of ambient sound.

[0010] According to an aspect of the invention, a headset for audio reproduction is provided. The headset includes at least one earpiece having an input for receiving an audio signal from the electronic device and an electro-acoustic speaker for converting the electrical signal to audible sound. The headset also includes at least one microphone for receiving ambient sound and converting the ambient sound to an ambient sound signal. In addition, the headset includes a processing circuit formed as an integrated unit having an input and at least one earpiece. The processing circuitry combines the audio signal from the electronic equipment with at least one microphone with a peripheral acoustic signal and provides the combined signal to at least one earpiece so that the electronic- acoustic speaker combines the audio from the ambient audio and the audio signal Reproducing circuit.

[0011] According to a particular aspect, the input includes an electrical connector configured to connect directly to the electronic equipment.

[0012] According to another aspect, the electrical connector includes an audio plug type connector.

[0013] According to another aspect, the processing circuit is connected to the electrical connector through the first audio cable and to the at least one earpiece through the second audio cable, all of which are part of the integrated unit.

[0014] According to another aspect, the electronic equipment is a mobile telephone.

[0015] According to another aspect, the input includes a low-power wireless interface that communicates with an interface included in the electronic device.

[0016] According to another aspect, the low-power wireless interface is a Bluetooth interface.

[0017] According to another aspect, the electronic equipment is a mobile telephone.

[0018] According to another aspect, the headset includes a pair of ear pieces each including an audio signal and an electro-acoustic speaker for reproducing audio from the ambient sound.

[0019] According to another aspect, the headset includes a pair of microphones included in each of the pair of ear pieces.

[0020] According to another aspect, the at least one earpiece is an earbud-style earpiece.

[0021] According to another aspect, the earbud-style earpiece includes at least one microphone.

[0022] According to another aspect, an earbud style earpiece is configured to be inserted into a user's ear, with the microphone positioned within a cavity formed by an auricle of the ear.

[0023] According to another aspect, a headset for audio reproduction is provided again. The headset includes an input for receiving an audio signal from the electronic equipment, at least one earpiece having an electro-acoustic speaker for converting the electrical signal into audible sound, and a headphone for receiving the ambient sound and converting the ambient sound to an ambient sound signal And at least one microphone. The headset also includes a processing circuit, wherein the processing circuitry combines an audio signal from the electronic equipment with a peripheral acoustic signal from at least one microphone and provides the combined signal to at least one earpiece, And circuitry for playing back the ambient sound combined with audio from the audio signal. The at least one earpiece includes at least one microphone, wherein the earpiece is configured to be inserted into a user & apos; s ear, wherein the microphone is positioned within a cavity formed by the outer ear of the ear.

[0024] According to another aspect, the headset comprises a pair of earpieces each comprising a respective electro-acoustic speaker and a corresponding microphone, each said electro- Sound is reproduced.

[0025] According to another aspect, the processing circuit includes a mixing circuit that allows the user to adjust the rate at which the processing circuit combines the audio signal and the ambient sound signal.

[0026] According to another aspect, the processing circuit includes a volume adjustment circuit that allows the user to adjust the volume of the combined signal independently of the ratio.

[0027] According to another aspect, the processing circuit includes a gain control circuit that compensates for the amplitude variation of the input signal such that the ratio is constant.

[0028] According to another aspect, the processing circuit includes a bypass circuit that allows the input signal to pass through at least one earpiece without substantial modification.

[0029] According to another aspect, the processing circuit comprises an integrated unit having the input section and at least one earpiece.

[0030] According to another aspect of the present invention there is provided a headset for audio reproduction comprising an input for receiving an audio signal from an electronic device, and at least one earpiece with an electro-acoustic speaker for converting the electrical signal into audible sound . The headset also includes at least one microphone for receiving ambient sound and converting the ambient sound to an ambient sound signal. The processing circuitry combines an audio signal from the electronic device with a peripheral acoustic signal from at least one microphone and provides a combined signal to at least one earpiece so that the electronic- And the like. The processing circuitry includes a mixing circuit that allows the user to adjust the rate at which the processing circuitry combines the audio signal and the ambient sound signal.

[0031] According to another aspect, the processing circuit includes a volume adjustment circuit that allows the user to adjust the volume of the combined signal independently of the ratio.

[0032] According to another aspect, the processing circuit includes a gain control circuit that compensates for a change in the amplitude of the input signal that remains constant as a constant.

[0033] According to another aspect, the processing circuit includes a bypass circuit that allows the input signal to pass through at least one earpiece without substantial modification.

[0034] According to another aspect, the at least one earpiece includes at least one microphone, and the earpiece is configured to be inserted into a user's ear, wherein the microphone is positioned within the cavity formed by the ear's outer ear.

[0035] According to another aspect, the processing circuit comprises an integrated unit having the input section and at least one earpiece.

[0036] BRIEF DESCRIPTION OF THE DRAWINGS The present invention, in order to achieve the foregoing and related ends, includes the features hereinafter fully described and particularly pointed out in the claims. The following description and the annexed drawings set forth in detail certain illustrative embodiments of the invention. However, these embodiments represent only a few of the various ways in which the principles of the invention may be used. Other objects, advantages and novel features of the invention will become apparent from the following description of the invention when considered in connection with the drawings.

[0037] The term & quot; comprises / comprising & quot ;, as used herein, is intended to specify the presence of stated features, integers, steps or elements, but not to limit the presence or addition of one or more other features, It is emphasized that this does not exclude the presence or addition of elements or groups thereof.

[0038] 1 is a peripheral view of a mobile telephone and a headset according to an embodiment of the present invention.

[0039] 2 is a cross-sectional view of an earbud including a microphone located within the cavity of the ear canal in accordance with an embodiment of the present invention.

[0040] 3 is a front view of a processor of a headset according to an embodiment of the present invention.

[0041] Figure 4 is a simplified circuit diagram of the processor of Figure 3 in accordance with an embodiment of the present invention.

[0042] 5 is a simplified circuit diagram of a processor according to another embodiment of the present invention.

[0043] BRIEF DESCRIPTION OF THE DRAWINGS The invention will now be described with reference to the drawings, in which like elements are given like reference numerals throughout the drawings.

[0044] The term & quot; electronic equipment & quot; as referred to herein is intended to encompass digital music players (e.g., CD, MP3, AC-3, .wma etc.), video players (e.g., DVD, MPEG, , A photo player (e.g., JPEG, etc.), a variety of media players such as, but not limited to: Electronic equipment includes traditional media players such as CD and DVD players, as well as media players in the form of very popular "flash memory" and "hard drive" formats, called memory formats for storing various media files. The electronic equipment includes not only a multi-function device including a media player but also a dedicated media player. Such multi-function devices include portable wireless communication equipment. The term & quot; portable wireless communication equipment & quot ;, also referred to as a & quot; mobile wireless terminal & quot; includes all equipment such as mobile phones, pagers, communicators, e.g., electronic organizers, personal digital assistants (PDAs) In this specification, the invention is primarily described with reference to a mobile telephone. However, it is to be understood that the present invention is not intended to be limited to mobile telephones, but may be any type of electronic equipment. Moreover, while the present invention is described primarily with respect to listening to music through electronic equipment such as a mobile telephone, it will be appreciated that the present invention is more generally applicable to any type of electronic equipment in which sound is reproduced.

[0045] Referring first to Figure 1, an electronic device 10 in accordance with the present invention is shown. In the exemplary embodiment described herein, the electronic equipment is a mobile telephone 10. Although mobile phone 10 is shown having a housing of the "brick" or "block" design type, it is understood that other types of housings, such as confrontation or slide type housings, may be used without departing from the scope of the present invention will be.

[0046] The mobile phone 10 includes a display 14 and a keypad 16. As is conventional, the display 14 displays information such as operational status, time, phone numbers, contact information, various navigation menus that allow the user to utilize various features of the mobile phone 10, and the like. Similarly, the keypad 16 is the same as the conventional one in that it provides various user input operations. For example, the keypad 16 typically includes alphanumeric keys for entering alphanumeric information such as phone numbers, phone listings, contact information, comments, and the like. Moreover, the keypad 16 can be used to provide a "call send" key 18 for making or receiving a call, and a "call-end" key 20 for stopping or "hanging" Typically include special function keys. The special function keys may include, for example, menu navigation keys 22a, 22b, and 22c (not shown) for navigating menus displayed on the display 14 to allow other phone functions, profiles,). & Lt; / RTI &) Other keys included in the keypad 16 include a volume key 24 and an on / off power key 26 as well as various other keys such as a web browser launch key, a camera key, You may.

[0047] In the particular embodiment of Figure 1, the mobile telephone 10 includes a display 14 and a separate keypad 16 thereon. In an alternative embodiment, the display 14 itself may include a touch screen that includes one or more keys. In another embodiment, the display 14 includes a touch screen that includes all or nearly all of the keys used to operate the phone 10 to include a highly localized keypad 16 or not to include the keypad 16 at all . As will be appreciated, the particular form and function of the key including the keypad 16 or the touch screen / display 14 are not necessarily in the broadest sense of the invention.

[0048] The mobile phone 10 is a multi-function device capable of performing various functions in addition to a conventional mobile phone function. For example, the mobile telephone 10 according to the present invention also functions as a media player. More specifically, the mobile telephone 10 can be an audio file (e.g., MP3, .wma, AC-3), a video file (e.g., MPEG, , JPEG, .bmp, etc.). The media objects are typically stored in non-volatile memory within the mobile telephone 10. [The mobile phone 10 reproduces the audio file through an accessory such as the speaker 28 or the stereo headset 30 according to the present invention. The stereo headset 30 may be plugged into or connected to the audio output of the mobile phone 10 via an appropriate connector.

[0049] For reasons that will be described later, the headset 30 overcomes the aforementioned disadvantages associated with a conventional headset using electronic equipment having an audio output. For example, the headset 30 is suitable for use with various types of electronic equipment. The headset 30 need not rely on the processing circuitry or control within the electronic equipment 10 to allow the user to hear the audio being played back and still be able to hear ambient sounds. Furthermore, the headset 30 provides the user with the ability to control the manner in which ambient sound is coupled to the audio being played back. The user can

control each level of audio being played and ambient sound, allowing the user to adjust the listening experience to his or her preference. In addition, the headset 30 preserves the spatial characteristics of ambient sound. As a result, the user can continuously obtain the same orientation as the direction in which the ambient sound originates. These features increase safety and improve the user's listening experience compared to conventional headsets.

[0050] The headset 30 in the illustrative embodiment includes an input 32 for receiving an audio signal from the mobile telephone 10. The input 32 includes a conventional audio jack / plug or any other suitable type of electrical connector. Although the audio signal may be a non-stereo signal (e.g., mono, multi-channel, etc.), the audio signal is preferably a stereo signal including audio for left and right channels.

[0051] Moreover, the headset 30 includes a processor 34 that receives audio signals at the input 32 via an audio cable 36. [In addition, the headset 30 includes a pair of ear pieces 38 connected to the processor 34 via an audio cable 40. Each earpiece 38 is the same as the conventional one in that the ear piece 38 includes an e-acoustic loudspeaker (not shown) for converting an electric signal into an audio sound, respectively, in the ear piece housing. Processor 34 provides left and right channel stereo signals from the output of mobile phone 10 to each earpiece 38.

[0052] In an exemplary embodiment, the earpiece 38 is an earpiece of the & quot; ear insert & quot; type, such as the ear bud 38, as shown in Fig. As mentioned herein, the earpiece 38 of the & quot; ear insert & quot; type is a type that the wearer wears generally within the cavity 42 formed by the ear outer ear 44. The ear bud 38 snugly fits within the cavity 42 and / or the opening 46 into the ear canal. Conversely, the ear piece of the "ear contact" type is generally worn by the user on the outside or top of the cavity 42 formed by the outer ear 44.

[0053] 1 and 2, each of the ear pieces 38 includes a microphone 50 for collecting ambient sound. The microphone 50 is positioned within the earpiece housing such that when the user wears the earpiece 38, the microphone 50 is located within the cavity 42 defined by the external ear 44 and is exposed to the cavity 42. Thus, as shown by the arrow 52 in Fig. 2, the ambient acoustic waves collected by the outer ear 44 are generally concentrated towards the microphone 50. [As a result, the microphone 50 collects ambient sound while preserving the spatial characteristics of the sound. That is, the microphone 50 collects the same ambient sound as the human being can sense the direction in which the sound originates based on the ear space and direction. Conversely, a microphone located within the earpiece of the ear contact type loses such spatial characteristics. As long as the microphone is continuously spaced apart, the characteristics associated with the ear direction are lost because it is not located within the cavity formed by the outer ear 44.

[0054] 2 further illustrates how each earpiece 38 includes the loudspeaker 56 described above. The loudspeaker 56 may be a conventional type (for example, a piezoelectric element, etc.). As shown in FIG. 2, the loudspeaker 56 receives audio signals from the processor 34 via the audio cable 40. Conversely, the microphone 50 receives ambient sound and converts such sound to ambient sound signals provided to the processor 34 via the audio cable 40.

[0055] The processor 34 functions to combine the audio signals received from the mobile telephone 10 with the ambient sound signals provided by the respective microphones 50, as will be described in more detail below with respect to Figures 3 and 4. [The processor 34 provides a signal coupled to the loudspeaker 56 in each earpiece 38. In such a manner, each earpiece 38 can reproduce ambient sound collected by the microphone 50 corresponding to the audio signal from the mobile telephone 10. Thus, the spatial characteristics of the ambient sound are preserved.

[0056] Referring briefly to Figure 1 again, the headset 30 in the preferred embodiment is an integrated or built-in unit. Specifically, input 32, processor 34, and earpiece 38 are fixedly connected to one another via audio cables 36, 40. Thus, the headset 30 represents a built-in accessory that does not have to depend on certain electronic equipment or any other accessory and / or accessories. The headset 30 includes its own processor 34 that acquires ambient sound and combines its ambient sound with audio from electronic equipment to provide the desired listening experience.

[0057] Referring to FIG. 3, an exemplary embodiment of processor 30 is shown. 4, processor 30 includes an on / off switch 70 that allows a user to turn processor 34 on and off. In the on state, the headset 30 functions to collect ambient sounds and to combine their ambient sounds with the audio signals from the mobile telephone 10 so that both are audible to the user. In the off state, the processor 34 basically sends an audio signal from the mobile phone directly to the earpiece 58, like a conventional headset. Although the on / off switch 70 is shown

as a slide switch, it will be understood that any suitable switch of any type may be used virtually without departing from the scope of the present invention.

[0058] The processor 34 also includes a volume control switch 72 and a mixing switch 74. The volume control switch 72 allows the user to control the audio volume reproduced by the headset 30 without adjusting the volume output of the mobile phone 10 directly or without adjusting it. The mixing switch 74 allows the user to adjust the rate at which the processor 34 combines the audio signal from the mobile telephone 10 and the ambient sound signal from the microphone 50. [By controlling the swap switch, the user can select whether the earpiece 38 provides more or less ambient sound with respect to the audio signal. According to this situation, the user may choose to block most or all environmental sounds by substantially reducing the amount of ambient sound provided to the loudspeaker 56 (e.g., in a congested train). The user may choose to maintain a low level of ambient sound, albeit at a reduced level, within a situation where it is desirable to maintain some level of contact with the environment (e.g., while in service). Moreover, the user may choose to maintain a higher level of ambient sound under circumstances where it is desirable to maintain full contact with the environment (e.g., in someone's living room).

[0059] As shown in FIG. 3, the switches 72 and 74 are membrane switches that function to adjust the volume and mix up or down by pushing a specific arrow. However, one of ordinary skill in the art will readily understand that any other suitable switch could also be used virtually without departing from the scope of the present invention. For example, switch 72 and / or switch 74 could be replaced by a rotary switch, a slide switch, or the like. As can be appreciated, the switches 72 and / or 74 together with any or all of the circuits described herein may be analog circuits, digital circuits, or any combination thereof. While the present invention has been described primarily with respect to analog design, it will be readily understood that the circuitry could alternatively be implemented through digital design or a combination thereof.

[0060] 4 is a schematic diagram of the processor 34. As shown in FIG. Processor 34 is simplified to represent a single channel. However, those skilled in the art will appreciate that the specific embodiment for handling the left and right channel stereo signals provided to each earpiece 38 merely includes a duplication of the circuit shown in FIG.

[0061] The mobile telephone 10 provides an audio signal to the processor 34 via a line 80 in an audio cable 36. The audio signal on line 36 is an input to adder 82 contained within processor 34. The output of the adder 82 is an input to an adjustable gain amplifier 84 that amplifies the output of the adder 82. The output of the amplifier is provided to the earpiece 38 through a line 86 in the audio cable 40.

[0062] The ambient sound collected by the microphone 50 is input to the processor 34 from the audio cable 40 as a peripheral acoustic signal on the line 88. The ambient acoustic signal on line 88 is input to an adjustable gain amplifier 90 for amplifying the ambient acoustic signal. The amplifier 90 outputs a peripheral acoustic signal to the adjustable gain amplifier 91 whose gain is adjusted using the level control switch in the form of the mixing switch 74 described above. In this embodiment, the mix switch 74, which is preferably a ramp resistor, regulates the level of the amplified ambient acoustic signal provided from the amplifier 91 to the other input of the adder 82.

[0063] Thus, the adder 82 functions to combine the ambient acoustic signal with the audio signal from the mobile telephone 10. By adjusting the mixing switch 74, the user can adjust the relative amplitude or ratio of the ambient acoustic signal band audio signal represented by the output of the adder 82. Thus, the user can adjust the degree to which the ambient sound appears relative to the audio signal.

[0064] In this embodiment, the volume switch 72, which is also preferably a bias resistor, functions to adjust the gain of the amplifier 84. [Thus, the user may adjust the gain of the amplifier 84 to amplify the combined audio and ambient acoustic signals. As a result, the user may adjust the overall volume of the sound played by the headset.

[0065] In a preferred embodiment, the processor 34 may further comprise an optional level detector circuit 92, which functions to detect the level of an audio signal received on the line 80 from the mobile telephone 10. [The level detector circuit 92 is designed to output a gain control signal on the line 94 that is proportional to the level of the audio signal on line 80. The gain adjustment signal on line 94 provides for adjustment of the gain of adjustable gain amplifier 90. As a result, the processor 34 automatically adjusts the level of the ambient acoustic signal in proportion to the change in the level of the received audio signal. In this way, even if the user adjusts the audio output volume of the mobile telephone 10 or other electronic equipment, the mixing ratio of the audio

signal and the surrounding acoustic signal remains substantially constant. Although not shown, the level detector 92 preferably smoothens the response of the amplifier 90 (e.g., to avoid sudden changes in ambient sound levels due to a sudden quiet / noisy period on the audio signal) And a low-pass filter for attenuation.

[0066] Processor 34 includes switch 70 described above that turns on and off power at power supply 100 to other remaining circuits. The power source 100 is preferably built-in, like a battery. Alternatively, the processor 34 is designed to receive power from the electronic equipment itself. The switch 70 may also be included as part of the optional switch 70a. The switch 70a is normally open during operation so that it is effectively an open circuit (switch 70). When the switch 70 is off, the switch 70a is closed. As a result, the audio signal on line 80 bypasses the circuitry within processor 34 and is directly output to earpiece 38 on line 86.

[0067] Moreover, the processor 34 may include a microphone 102 that uses the headset 30 to collect voice of the user during hands-free communication with the mobile telephone 10. [The microphone 102 provides a voice signal to the mobile telephone 10 on the line 104 using conventional techniques.

[0068] FIG. 5 illustrates an alternative embodiment of processor 34. As shown in FIG. In this embodiment, the input 32 is replaced by a low-power wireless interface, such as a Bluetooth transceiver 106. [The Bluetooth transceiver 106 may be programmed to perform conventional communications between a mobile telephone 10 or other electronic equipment equipped similarly to the headset 30. [In such an embodiment, an audio cable 36 is not required.

[0069] Moreover, the embodiments described herein may be modified without departing from the scope of the present invention. For example, while processor 34 is shown as part of a housing that is connected in series between ear piece 38 and input 32, processor 34 may be coupled to a jack And / or in the housing of the earpiece 38 instead. Alternatively, in a Bluetooth embodiment, the processor 34 may be coupled within a small housing designed to fit behind the user's ear, similar to a known Bluetooth headset.

[0070] While the invention has been shown and described with reference to preferred embodiments thereof, it is to be understood that equivalents and modifications may be effected by others skilled in the art upon the reading and understanding of this specification. The present invention includes all such equivalents and modifications and is limited only by the following claims.

KR20080111004A HEADSET WITH AMBIENT SOUND

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

1. A headset for reproducing audio,

An input unit for receiving an audio signal from the electronic equipment;

At least one earpiece having an electro-acoustic speaker for converting electrical signals into audible sound,

At least one microphone for receiving ambient sound and converting ambient sound to ambient sound signals,

A processing circuit formed of an integrated unit of the input section and the at least one earpiece, the processing circuit being configured to combine an audio signal from the electronic equipment with a peripheral acoustic signal from at least one microphone, And at least one earpiece to enable the electro-acoustic speaker to reproduce the ambient sound combined with audio from the audio signal.

2. The headset of claim 1, wherein the input includes an electrical connector configured to be connected directly to the electronic equipment.

3. The headset according to claim 2, wherein the electrical connector includes a connector of an audio plug type.

3. The audio playback headset of claim 2, wherein the processing circuitry is coupled to the electrical connector via a first audio cable and to the at least one earpiece through a second audio cable, .

The headset according to claim 2, wherein the electronic equipment is a mobile telephone.

2. The headset of claim 1, wherein the input includes a low power wireless interface communicating with an interface included in the electronic equipment.

7. The headset of claim 6, wherein the low power wireless interface is a Bluetooth interface.

7. The headset according to claim 6, wherein the electronic equipment is a mobile telephone.

2. The headset of claim 1, wherein the headset comprises a pair of earpieces each including an audio signal and an electro-acoustic speaker for reproducing audio from the ambient sound.

10. The headset according to claim 9, wherein the headset includes a pair of microphones each included in each of the pair of ear pieces.

2. The headset of claim 1, wherein the at least one earpiece is an earbud style earpiece.

12. The headset of claim 11, wherein the earbud style earpiece comprises at least one microphone.

13. The headset of claim 12, wherein the earbud style earpiece is inserted into a user's ear, the microphone being positioned within a cavity formed by an auricle of the ear.

1. A headset for reproducing audio,

An input unit for receiving an audio signal from the electronic equipment;

At least one earpiece having an electro-acoustic speaker for converting electrical signals into audible sound,

At least one microphone for receiving ambient sound and converting ambient sound to ambient sound signals,

Combining an audio signal from the electronic equipment with a peripheral acoustic signal from at least one microphone and providing a combined signal to the at least one earpiece so that the electronic- A processing circuit including a circuit for reproducing sound,

Lt; / RTI &)

Wherein the at least one earpiece includes at least one microphone and wherein the earpiece is configured to position the microphone within a cavity formed by the outer ear of the ear for insertion into a user & apos; s ear.

15. The method of claim 14, wherein the headset comprises a pair of earpieces each comprising a respective electro-acoustic speaker and a corresponding microphone, each of the electro- A headset for audio playback that reproduces ambient sound.

15. The headset of claim 14, wherein the processing circuitry includes a mixing circuit that allows the user to adjust the rate at which the processing circuitry combines the audio signal and the ambient sound signal.

17. The headset of claim 16, wherein the processing circuitry includes a volume adjustment circuit that allows the user to adjust the volume of the combined signal independently of the ratio.

17. The headset of claim 16, wherein the processing circuit comprises a gain control circuit that compensates for the amplitude variation of the input signal such that the ratio is constant.

15. The headset of claim 14, wherein the processing circuit includes a bypass circuit that allows the input signal to pass through at least one earpiece without substantial modification.

15. The headset of claim 14, wherein the processing circuit comprises an integrated unit having the input section and at least one earpiece.

1. A headset for reproducing audio,

An input unit for receiving an audio signal from the electronic equipment;

At least one earpiece having an electro-acoustic speaker for converting electrical signals into audible sound,

At least one microphone for receiving ambient sound and converting ambient sound to ambient sound signals,

Combining the audio signal from the electronic equipment with at least one ambient acoustic signal from the microphone and providing the combined signal to at least one earpiece to enable the electronic- A processing circuit including circuitry for reproducing ambient sound,

Lt; / RTI &)

Wherein the processing circuitry includes a mixing circuit that allows the user to adjust the rate at which the processing circuitry combines the audio signal and the ambient sound signal.

22. The headset of claim 21, wherein the processing circuit comprises a volume adjustment circuit that allows a user to adjust the volume of the combined signal independently of the ratio.

22. The headset of claim 21, wherein the processing circuit comprises a gain control circuit that compensates for amplitude variations of the input signal such that the ratio is constant.

22. The headset of claim 21, wherein the processing circuit comprises a bypass circuit to allow the input signal to pass through at least one earpiece without substantial modification.

22. The headset of claim 21, wherein the at least one earpiece includes at least one microphone, and wherein the earpiece is configured to position the microphone within a cavity formed by the outer ear of the ear.

22. The headset of claim 21, wherein the processing circuit is comprised of an integrated unit having the input and at least one earpiece.

Espacenet

ES2273616A1 Multifunctional headphones for simultaneously listening to audio signal and ambient sound has control circuit that operates activation/deactivation units to automatically balance audio signals

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Multifunctional headphones for simultaneously listening to audio signal and ambient sound has control circuit that operates activation/deactivation units to automatically balance audio signals

Abstract

Ear pieces are comprised of classis loudspeakers connected to a control circuit having an audio signal mixer whose input is connected to at least an audio line of a reproduction device and a microphone to catch environmental sound. An amplification stage reproduces the signals from the loudspeakers simultaneously. The control circuit is programmed to automatically balance the audio signals by operating activation/deactivation units.

ES2273616A1 Multifunctional headphones for simultaneously listening to audio signal and ambient sound has control circuit that operates activation/deactivation units to automatically balance audio signals

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Multifunction headphones.

OBJECT OF THE INVENTION

The present invention relates to a headset, for allowing the listening of audio signals from playback equipment such as audio signals, computers, radios, MP3's, walkman, and the like.

The object of the invention to provide a pair of headphones that allow the user to listen in parallel both the audio signal from the player concerned equipment and the surrounding sound environment where the user is located.

It is also an object of the invention that the device for regulating the volume of input signal again and will, to the point momentarily inhibit either.

The invention is particularly applicable in the field of video calls or video conferencing, can nevertheless applied different devices such as the above.

BACKGROUND OF THE INVENTION

Often the use of headphones when listening to the radio, music, etc., through breeding devices both fixed and portable, such as CD players, MP3, radios, mini-disc, computers, stereos, etc. .

The problem with these devices is that they have a configuration that insulates the user's ear against the outside ambient sound, property may not be desirable in many cases, such as in the case of meetings where many people participate, and that some of them use headphones for video-conferencing, listening to translations of a participant, etc, which forces them to be continually taking off and putting on the aforementioned headphones, so that when the user has positions is not able to follow conversations that are keeping around.

On the other hand, in the field of listening to acoustic signals from portable devices, said user privacy against the sounds in your environment can sometimes be dangerous, not to notice it any alarm or warning sound nature.

Trying to overcome this problem, in certain types of headphones on the left ear cup is independent physically the right earpiece, it is likely that the user will meet only one of the two headphones to be attentive to the proximal ambient sound, but this solution prevents that it can hear the audio signal in stereo player, significantly reducing the quality of it, so it is an unsatisfactory solution.

DESCRIPTION OF THE INVENTION

Headphones proposed by the invention fully and satisfactorily solve the aforementioned problems, allowing performed simultaneously listening to audio signal proviniente a parallel to the audio signal corresponding to the environment around the device player.

To that end and more specifically, it is anticipated that based on the classic structure of a conventional headphones, whatever type they are, the speaker or speakers are associated with an electronic circuit, equipped with a mixer signal circuit to which accessed by a hand signal sound from the player in question through the corresponding line equipment, and secondly the signal proviniente at least one microphone for receiving sounds that are emitted in the vicinity of the device.

According to another feature of the invention, said control circuit means will volume control and on / off of each of the signals.

Furthermore, the control circuit is programmed to perform self-leveling headphone volume of that way circuit is responsible for eliminating external sounds that do not exceed a preset value decibels as the level of activation of the external signal and auto leveling likewise adjusting the volume level of the audio signal proviniente player device, restoring the original level gradually to the end of the input signal external audio proviniente microphone.

Finally include the microphone or microphones may be integrated into the housing in which each speaker, or independently, so that the microphone can be tapped for use in video conferencing, integrated video-calls and the like, so that the control circuit incorporates an input line for signal transmission to the equipment in question.

It is achieved in this way a device that allows proviniente together to hear the sound of an audio player and the proviniente the exterior, comfortable, easy to use and fully automatic.

DESCRIPTION OF DRAWINGS

To complement the description being made and in order to aid a better understanding of the invention's characteristics, according to a preferred practical embodiment thereof, accompanying as an integral part of said description a set of drawings where illustrative and non limitative character, it shows the following:

Figure 1 shows a schematic representation of the various components involved in a multifunction headphones performed in accordance with the subject invention.

Figure 2 shows a perspective view of a possible practical embodiment of the invention.

Figure 3 shows a perspective view of two different embodiments of the invention, in which the device is materialized in a single handset.

PREFERRED EMBODIMENT OF THE INVENTION

In view of the described and especially figure 1 figures, it can be seen in headphones of the invention involved, as is conventional, a pair of speakers (1-1 ') which are integrated in the housing of corresponding auricular , speakers associated with an audio player (2) computer, through an audio cable or line (3) with the mating connector (4).

However, from the above-described conventional structure, the invention centers its characteristics on the fact that between the speakers (1-1 ') and the audio line (3) a control circuit (5) is set to the a mixer audio signal (6) is integrated.

To converge said mixer signals audio line (3) the player (2), and the signal proviniente at least one microphone (7) after amplification (8) of the signal.

Thus, the circuit at the output (9) mixes both signals according to the parameters of internal programming, so that he would listen to user both signals if you should not have a headset and I was listening to said audio signal through one or more speakers conventional.

In addition and as stated above, the control circuit (4) is programmed to automatically perform self-leveling signals, incorporating internal filters that eliminate what could be considered as environmental noise, consisting of lower sounds to a preset minimum level.

The control circuit also has means for adjusting (10) the volume of the audio signal from the player team and cutoff switches (11) and (12) for disabling one or the other incoming signal to the mixer (6) as it deemed appropriate.

The microphone (7) can be integrated into the housing (7-7 ') of the headset, as shown in the embodiments shown in Figure 3, or other area set of headphones, for example to the neck the user, as shown in figure 2, being able to use this microphone to capture is PRG end of the user's voice, in which case the control circuit (5) provided with an input line (14) with its corresponding connector (15) for treatment of PRG end is on the computer (2) concerned.

ES2273616A1 Multifunctional headphones for simultaneously listening to audio signal and ambient sound has control circuit that operates activation/deactivation units to automatically balance audio signals

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

1. Multifunction headphones, which being of the type incorporating at least one speaker (1-1 ') associated with an audio signal output of a player (2) via the corresponding audio line (3) equipment, characterized it includes a control circuit (5) in which a mixer (6) of audio signals, the mixer (6) whose input is associated with said audio line (3) from the equipment player (2) is set as the least one microphone (7) conveniently arranged to capture ambient sound with its corresponding amplification stage (8), audio mixer (6) to the output of said speaker or speakers (1-1 ') are established players both signals .

2. Multifunction headset according to claim 1, wherein the control circuit (5) has cut switches (11-12) are of disabling PRG proviniente nal microphone (7) and team player (2), a regulator volume (10) and an internal software programming self-leveling volume are both PRG nal.

3. Multifunction headphones, according to previous claims, characterized in that the control circuit (5) has a line (14) connecting the auxiliary audio is amplified PRG nal microphone (7) with the player (2) equipment.

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(54) Título: **Auriculares multifunción.**

(57) Resumen:
Auriculares multifunción.
Especialmente concebidos para permitir a su usuario escuchar simultáneamente la señal de audio proveniente de un dispositivo reproductor (2), así como el sonido ambiente que le rodea, los auriculares están constituidos a partir de los clásicos altavoces (1-1') asociados a un circuito de control (5) en el que se establece un mezclador (6) de señales de audio, mezclador (6) cuya entrada está asociada a la línea de audio (3) proveniente del equipo reproductor (2) y a al menos un micrófono (7) convenientemente dispuesto para captar el sonido ambiental, incluyendo la correspondiente etapa de amplificación (8), de manera que a través de los citados altavoces (1-1') se reproducen simultáneamente ambas señales. El circuito de control está programado para autonivelar las señales de audio, contando con medios de activación/inhabilitación (11-12) de las mismas.

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DESCRIPCIÓN

Auriculares multifunción.

Objeto de la invención

La presente invención se refiere a unos auriculares, de los destinados permitir la escucha de señales de audio provenientes de equipos de reproducción de señales de audio tales como, ordenadores, radios, mp3's, "walkman", y similares.

El objeto de la invención es proporcionar unos auriculares que permitan a su usuario escuchar paralelamente tanto la señal de audio proveniente del equipo reproductor de que se trate, así como el sonido ambiente de los alrededores en donde se encuentre el usuario.

Es asimismo objeto de la invención que el dispositivo permita regular el volumen de entrada de una y otra señal a voluntad, hasta el punto de inhibir momentáneamente cualquiera de las dos.

La invención resulta de especial aplicación en el ámbito de las video-llamadas o video-conferencias, pudiendo no obstante aplicarse diferentes dispositivos como los anteriormente citados.

Antecedentes de la invención

Es frecuente el uso de auriculares a la hora de escuchar la radio, música, etc, a través de dispositivos reproductores tanto fijos como portátiles, como pueden ser reproductores de CD's, MP3, radios, minidisc, ordenadores, cadenas de música, etc.

El problema que presentan este tipo de dispositivos es que presentan una configuración que aísla el oído del usuario frente al sonido ambiente exterior, propiedad que puede no ser deseable en muchos casos, como por ejemplo en el caso de reuniones en las que participan múltiples personas, y en las que algunas de ellas emplean auriculares para realizar videoconferencias, escuchar traducciones de algún participante, etc, lo que obliga a estar continuamente quitándose y poniéndose los citados auriculares, de manera que cuando el usuario los tiene puestos no es capaz de seguir las conversaciones que se estén manteniendo a su alrededor.

Por otro lado, y en el ámbito de la escucha de señales acústicas provenientes de dispositivos reproductores portátiles, el citado aislamiento del usuario frente a los sonidos de su entorno puede resultar en ocasiones peligroso, al no advertir éste cualquier alarma o advertencia de naturaleza sonora.

Tratando de obviar este problema, en ciertos tipos de auriculares en los que el auricular izquierdo es independiente físicamente del auricular derecho, es factible que su usuario se adapte únicamente uno de los dos auriculares para poder estar atento al sonido ambiente proximal, pero esta solución impide que el mismo pueda escuchar la señal de audio del reproductor en estéreo, reduciendo sensiblemente la calidad de la misma, por lo que resulta una solución poco satisfactoria.

Descripción de la invención

Los auriculares que la invención propone resuelven de forma plenamente satisfactoria la problemática anteriormente expuesta, permitiendo realizar de forma simultánea la escucha de una señal de audio proviniente de un reproductor paralelamente a la señal de audio correspondiente al ambiente de los alrededores del dispositivo.

Para ello, y de forma más concreta, se ha previsto que partiendo de la estructuración clásica de unos auriculares convencionales, sean del tipo que sean, el altavoz o altavoces estén asociados a un circuito electrónico, dotado de un mezclador de señales, circuito al que accede por un lado la señal de sonido proveniente del equipo reproductor de que se trate a través de la correspondiente línea, y por otro la señal proviniente de al menos un micrófono destinado a recibir los sonidos que se emitan en las proximidades del dispositivo.

De acuerdo con otra de las características de la invención, el citado circuito de control contará con medios de regulación del volumen así como de activación/desactivación de cada una de las señales.

Asimismo, el circuito de control estará programado para realizar la auto-nivelación del volumen de los auriculares, de manera dicho circuito se encarga de eliminar los sonidos externos que no superen un valor de decibelios preestablecido como nivel de activación de la señal externa y auto nivelación, regulando igualmente el nivel del volumen de la señal de audio proviniente del dispositivo reproductor, reestableciendo el nivel original de forma progresiva a término de la señal de entrada de audio externa proviniente del micrófono.

Por último cabe destacar que el micrófono o micrófonos podrán integrarse en la carcasa en la que se integra cada altavoz, o bien de forma independiente, de manera que dicho micrófono puede ser aprovechado para su empleo en video-conferencias, video-llamadas y similares, de manera que el circuito de control incorpore una línea de entrada para transmisión de la señal al equipo de que se trate.

Se consigue de esta forma un dispositivo que permite escuchar conjuntamente el sonido proviniente de un reproductor de audio así el proviniente del exterior, cómodo, sencillo de usar y totalmente automático.

Descripción de los dibujos

Para complementar la descripción que se está realizando y con objeto de ayudar a una mejor comprensión de las características del invento, de acuerdo con un ejemplo preferente de realización práctica del mismo, se acompaña como parte integrante de dicha descripción, un juego de dibujos en donde con carácter ilustrativo y no limitativo, se ha representado lo siguiente:

La figura 1.- Muestra una representación esquemática de los distintos componentes que participan en unos auriculares multifunción realizados de acuerdo con el objeto de la invención.

La figura 2.- Muestra una vista en perspectiva de una posible realización práctica de la invención.

La figura 3.- Muestra una vista en perspectiva dos realizaciones prácticas distintas de la invención, en las que el dispositivo se materializa en un único auricular.

Realización preferente de la invención

A la vista de las figuras reseñadas y en especial de la figura 1, puede observarse como en los auriculares de la invención participan, como es convencional, una pareja de altavoces (1-1'), que se integran en la carcasa del correspondiente auricular, altavoces asociados a un equipo reproductor de audio (2), a través de un cable o línea de audio (3) con el complementario conector (4).

Pues bien, a partir de la estructuración convencional anteriormente descrita, la invención centra sus ca-

Exhibit 1004
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racterísticas en el hecho de que entre los altavoces (1-1') y la línea de audio (3) se establece un circuito de control (5) en el que se integra un mezclador de señales de audio (6).

Al citado mezclador confluyen las señales de la línea de audio (3) del reproductor (2), así como la señal proviniente de al menos un micrófono (7) previa amplificación (8) de la señal.

Así pues, el circuito a la salida (9) mezcla ambas señales según los parámetros de programación interna, de manera tal que el usuario escucharia ambas señales si no dispusiera de unos auriculares y estuviera escuchando dicha señal de audio a través de uno o más altavoces convencionales.

Complementariamente y tal y como se ha dicho con anterioridad, el circuito de control (4) estará programado para realizar de forma automática la auto-nivelación de las señales, incorporando filtros internos que eliminen lo que podría considerarse como ruido ambiental, constituido por sonidos inferiores a un nivel mínimo preestablecido.

El circuito de control cuenta asimismo con medios de regulación (10) del volumen de la señal de audio proveniente del equipo reproductor, así como interruptores de corte (11) y (12) para inhabilitación de una u otra señal entrante al mezclador (6) según se estime conveniente.

El micrófono (7) puede estar integrado en la carcasa (7-7') del auricular, tal como muestran las realizaciones mostradas en la figura 3, o bien establecerse en cualquier otra zona de los auriculares, como por ejemplo a la altura del cuello del usuario, tal como muestra la figura 2, pudiéndose utilizar dicho micrófono para la captura de la señal de la voz del usuario, en cuyo caso el circuito de control (5) dispondrán de una línea de entrada (14) con su correspondiente conector (15) para tratamiento de la señal en el equipo (2) de que se trate.

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REIVINDICACIONES

1. Auriculares multifunción, que siendo del tipo de los que incorporan al menos un altavoz (1-1') asociado a una salida de señal de audio de un equipo reproductor (2) a través de la correspondiente línea de audio (3), se **caracteriza** porque incluye un circuito de control (5) en el que se establece un mezclador (6) de señales de audio, mezclador (6) cuya entrada está asociada a la citada línea de audio (3) proveniente del equipo reproductor (2) y a al menos un micrófono (7) convenientemente dispuesto para captar el sonido ambiental, con su correspondiente etapa de amplificación (8), mezclador de audio (6) a cuya salida se establecen el citado altavoz o altavoces (1-1') reproductores de ambas señales.

2. Auriculares multifunción, según reivindicación 1ª, **caracterizados** porque el circuito de control (5) cuenta con interruptores de corte (11-12) de inhabilitación de la señal proviniente del micrófono (7) y del equipo reproductor (2), un regulador de volumen (10), así como un software interno de programación de la auto-nivelación del volumen de ambas señales.

3. Auriculares multifunción, según reivindicaciones anteriores, **caracterizado** porque el circuito de control (5) cuenta con una línea (14) de audio auxiliar que comunica la señal amplificada del micrófono (7) con el equipo reproductor (2).

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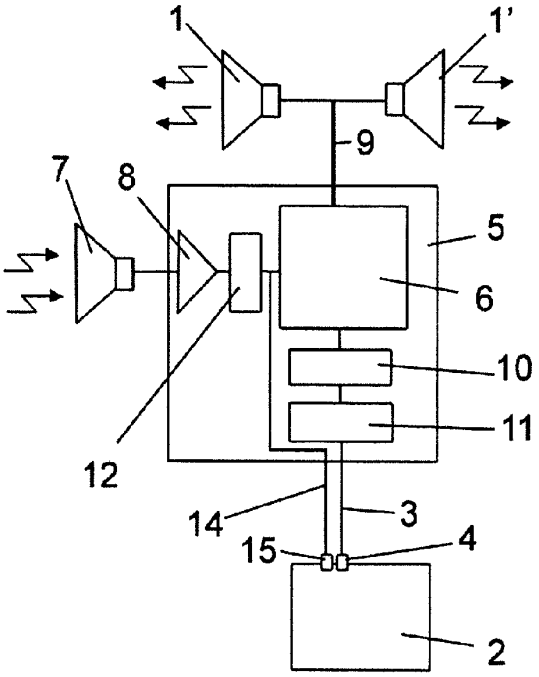


FIG. 1

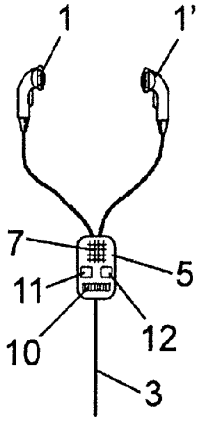


FIG. 2

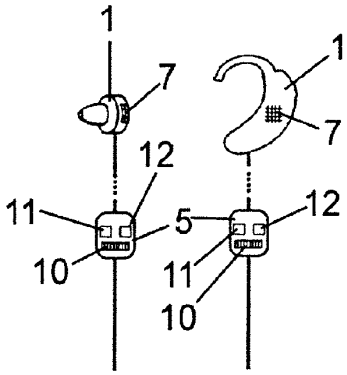


FIG. 3

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DOCUMENTOS RELEVANTES

Categoría	Documentos citados	Reivindicaciones afectadas
X	US 2003035551 A1 (LIGHT et al.) 20.02.2003, resumen; figura 2; párrafos 12,13,16.	1-3
X	US 2006182287 A1 (SCHULEIN et al.) 17.08.2006, párrafos 7-11,14,19,26.	1-3
X	GB 2256996 A (CIRCUITS MAXIMUS CO INC) 23.12.1992, resumen; figura 1; páginas 1,4,5,9.	1-3
X	JP 63221800 A (CANON KK) 14.09.1988, resumen.	1-3

Categoría de los documentos citados

X: de particular relevancia
Y: de particular relevancia combinado con otro/s de la misma categoría
A: refleja el estado de la técnica
O: referido a divulgación no escrita
P: publicado entre la fecha de prioridad y la de presentación de la solicitud
E: documento anterior, pero publicado después de la fecha de presentación de la solicitud

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☒ para todas las reivindicaciones ☐ para las reivindicaciones nº:

Fecha de realización del informe	Examinador	Página
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Espacenet

CN105554610B 耳机环境声音的调节方法和耳机

Applicants: 北京小鸟听听科技有限公司

Inventors: 刘淼,赵剑

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耳机环境声音的调节方法和耳机

Abstract

本发明公开了种耳机环境声音的调节方法和耳机。该方法包括：在耳机外侧设置至少个采集环境声音的外置麦克风，在耳机外壳上或线控器的壳体上设置机械式可移动的调节器。调节器具有电气端口，通过电气端口将调节器连接至耳机的降噪模块和/或监听模块。该方法还包括：实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量；按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。本发明的技术方案实现了耳机对环境声音的灵活调节，使耳机功能更人性化，增强了耳机性能，提高了用户体验。

CN105554610B Method for regulating environmental sound of earphone and earphone

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

Headphone environment sound adjustment method and earphone

Technical field

The present invention relates to the field of earphone technology, and in particular, to a method for adjusting sound of a headphone environment and an earphone.

Background technique

The improvement of social information level enables people to communicate and communicate anytime and anywhere. The wide application of various communication devices (mobile phones, Bluetooth headsets, stereo headphones, etc.) and communication technologies greatly facilitates people's lives and improves their lives. Work efficiency. However, a serious problem brought about by social development is the noise problem. Communication in a noisy environment seriously affects the clarity and intelligibility of communication voice. When the noise is high to a certain extent, not only communication cannot be performed, but also It hurts people's hearing and physical and mental health.

For the noise problem, some earphone products with noise reduction capability are provided in the existing solution. When the noise reduction function of the noise canceling headphone is turned on, the earphone will reduce the noise of the outside sound, and then output to the earphone speaker to let the earphone wearer hear the noise after noise reduction.

However, because the ambient noise is different, the energy level and spectrum distribution of the ambient noise are different at different times and different locations. The existing noise canceling headphones only have two ways of turning on the noise reduction function and turning off the noise reduction function. Directly implementing the noise reduction function does not bring a good user experience to the earphone wearer.

For example, in an ordinary office environment, if the headphone noise reduction function is directly turned on to provide more than 25 dB of noise reduction, the earphone wearer will have a feeling of depression, like a professional acoustic anechoic chamber, and in this environment, there is a certain The environmental noise can not directly turn off the noise reduction function.

Summary of the invention

The invention provides a method for adjusting the ambient sound of the earphone and the earphone, so as to solve the single problem that the existing noise canceling earphone can directly realize the noise reduction function and the noise reduction mode.

In order to achieve the above object, the technical solution of the present invention is achieved as follows:

In one aspect, an embodiment of the present invention provides a method for adjusting an ambient sound of a headset. The method comprises: setting at least one external microphone for collecting ambient sound on the outside of the earphone, setting a mechanical movable adjuster on the earphone casing, or setting a mechanical movable adjustment on the casing of the earphone of the earphone The regulator has an electrical port, and the regulator is connected to the noise reduction module and/or the monitoring module of the earphone through the electrical port, the method includes: monitoring the movement state of the regulator in real time, and knowing when the regulator moves The amount of movement and the type of movement of the regulator; confirm the current adjustment type according to the type of movement, and confirm the current adjustment amount of the ambient sound according to the amount of movement; according to the confirmed adjustment type, control the noise reduction module or the monitoring module to use the confirmed adjustment amount The ambient sound collected by the microphone is adjusted.

Preferably, the method further comprises: dividing the movement state of the regulator into a first movement type corresponding to the noise reduction control and a second movement type corresponding to the monitoring control according to the movement behavior of the regulator; Confirming the current adjustment type according to the movement type, and confirming the current adjustment amount of the ambient sound according to the movement amount, including: when monitoring that the movement of the regulator belongs to the first movement type, confirming that the current adjustment type is noise reduction control, and according to The amount of movement confirms the current amount of noise reduction for the ambient sound; and/or, when it is detected that the movement occurring by the regulator belongs to the second movement type, confirms that the current adjustment type is the monitoring control, and confirms the current environmental sound according to the movement amount. Put a lot.

Preferably, the real-time monitoring of the movement state of the adjuster, when the adjuster moves, knowing the amount of movement and the type of movement of the adjuster include: setting the rotation demarcation point on the rotation trajectory of the adjuster when the adjuster is realized by the rotating component When it is detected that the rotating component rotates to a predetermined side of the rotation demarcation point, it is learned that the movement type of the rotating component is the first movement type, and when the rotation component is detected to rotate to the other side of the rotation demarcation point, it is known that the movement type of the rotating component is The second type of movement is obtained by the amount of rotation of the rotating member.

Preferably, the real-time monitoring of the movement state of the adjuster, when the adjuster moves, knowing the amount of movement and the type of movement of the adjuster include: setting the stroke demarcation point on the push stroke of the adjuster when the adjuster is implemented by the push button When it is detected that the push button slides to the predetermined side of the stroke demarcation point, it is known that the movement type of the push button is the first movement type, and when the push button is detected to slide to the other side of the stroke demarcation point, the movement type of the push button is known as The second type of movement, and the amount of movement is obtained by the sliding distance of the push button.

Preferably, the real-time monitoring of the movement state of the adjuster, when the adjuster moves, knowing the amount of movement and the type of movement of the adjuster include: when the adjuster is implemented by two buttons, when the first button is detected to be pressed, It is known that the movement type of the button is the first movement type. When the second button is detected to be pressed, it is known that the movement type of the button is the second movement type, and the movement amount is obtained by the number of times the button is pressed.

Preferably, when the noise reduction module and/or the monitoring module are implemented by using the MCU, the written value of the MCU control register is modified by using the confirmed adjustment amount, and the adjustment of the ambient sound collected by the microphone by the noise reduction module or the monitoring module is implemented.

Preferably, the confirming the current adjustment amount of the ambient sound according to the movement amount comprises: querying the correspondence between the movement amount and the adjustment amount according to the movement amount, and obtaining the current adjustment amount for the environmental sound.

Preferably, the adjuster is disposed on a housing of the earpiece portion of the headset.

Preferably, a listening mode triggering module is disposed on the earphone casing, or a monitoring triggering module is disposed on the casing of the wire controller of the earphone; the method further includes: monitoring data output by the monitoring mode triggering module in real time, according to the The data output by the listening mode triggering module determines whether the triggering condition of the listening mode is satisfied; when the triggering condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the non-listening mode, the current playing volume and the pair of the earphone are recorded. The adjustment amount of the ambient sound, and controlling the monitoring module to adjust the ambient sound collected by the microphone by using a preset adjustment amount; when the trigger condition of the monitoring mode is changed from being satisfied to not satisfied, according to the recorded playing volume and the pair The amount of adjustment of the ambient sound controls the headset to return to its original working state.

Preferably, the method further includes: when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the noise reduction playing audio mode or simply playing

the audio mode, the playing volume of the earphone is also adjusted to be adjusted in advance. The set playback volume.

Preferably, the monitoring mode triggering module comprises any one of the following: a rotatable panel switch, a touchpad with single or multi-touch function.

Preferably, the listening mode triggering module comprises any one of the following: a capacitive touch sensor, a resistive touch sensor, an infrared distance sensor, a pressure sensor, and a mechanical switch.

In another aspect, an embodiment of the present invention provides an earphone, wherein the earphone is provided with at least one external microphone for collecting ambient sound, the earphone casing is provided with a mechanical movable adjuster, or the wire controller of the earphone a mechanically movable regulator is disposed on the housing; the regulator has an electrical port, the regulator is connected to the noise reduction module and/or the monitoring module of the earphone through an electrical port, the noise reduction module and/or The monitoring module is connected to the main control chip of the earphone, or integrated in the main control chip; the main control chip monitors the movement state of the regulator, and when the regulator moves, the movement amount and the movement type of the regulator are known; the main control chip is based on The movement type confirms the current adjustment type, and confirms the current adjustment amount of the ambient sound according to the movement amount; the main control chip controls the noise reduction module or the monitoring module to use the confirmed adjustment amount to perform the ambient sound collected by the microphone according to the confirmed adjustment type. Adjustment.

Preferably, when the regulator is implemented by the rotating component, a rotation demarcation point is set on the rotation trajectory of the regulator, and when the main control chip detects that the rotating component rotates to a predetermined side of the rotation demarcation point, it is known that the movement type of the rotating component is the first The type of movement, when it is detected that the rotating member rotates to the other side of the rotation demarcation point, it is learned that the movement type of the rotating member is the second movement type, and the amount of movement is obtained by the rotation amount of the rotating member; when the regulator is realized by the push button The stroke demarcation point is set on the push stroke of the adjuster, and the main control chip detects that the push button slides to the predetermined side of the stroke demarcation point, and knows that the movement type of the push button is the first movement type, and the push button is adjusted to the stroke boundary. When the other side of the point slides, it is known that the movement type of the push button is the second movement type, and the movement amount is obtained by the sliding distance of the push button; when the adjuster is realized by two buttons, the first button is detected to be pressed. The master chip knows that the movement type of the button is the first movement type, and when the second button is pressed, it is known that the movement type of the button is the second movement. The number of pressing type, obtained by said movement amount of the key.

Preferably, the adjuster is disposed on a housing of the earpiece portion of the headset.

Preferably, the earphone casing is further provided with a monitoring mode triggering module, or the casing of the earphone of the earphone is provided with a monitoring triggering module; the main control chip is further configured to monitor the monitoring mode triggering module in real time. The output data determines whether the trigger condition of the monitor mode is satisfied according to the data output by the monitor mode trigger module; and, when the trigger condition of the monitor mode is changed from unsatisfied to satisfied, if the headset is currently in the non-listening mode, the record is The current playing volume of the earphone and the adjustment amount of the ambient sound, and controlling the monitoring module to adjust the ambient sound collected by the microphone by using a preset adjustment amount; when the triggering condition of the listening mode is changed from satisfying to not satisfied The earphone is restored to the original working state according to the recorded playing volume and the amount of adjustment to the ambient sound.

Preferably, the main control chip is further configured to: when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, the playing volume of the earphone is also controlled. Adjust to the preset playback volume.

Preferably, the monitoring mode triggering module comprises any one of the following: a rotatable panel switch, a touchpad with single or multi-touch function.

Preferably, the listening mode triggering module comprises any one of the following: a capacitive touch sensor, a resistive touch sensor, an infrared distance sensor, a pressure sensor, and a mechanical switch.

In another aspect, an embodiment of the present invention provides a method for adjusting an ambient sound of a headset, the method comprising: setting at least one external microphone for collecting an ambient sound outside the earphone; and setting a touch with a multi-touch function on the earphone casing a control board, the touch panel is connected to the main control chip of the earphone; the noise reduction module and/or the monitoring module of the earphone is connected to the main control chip of the earphone, or integrated in the main control chip; the touch panel detects the user a movement trajectory of the finger, the main control chip confirms an adjustment type and an adjustment amount of the current ambient sound according to the movement trajectory, the adjustment type of the ambient sound includes a first type corresponding to the noise reduction control, and corresponds to the monitoring The second type of control; the main control chip controls the noise reduction module or the monitoring module to adjust the ambient sound collected by the microphone according to the confirmed adjustment type.

Preferably, when the touchpad detects that the user's palm covers the touchpad, the master chip controls the headset to enter a listening mode.

In another aspect, an embodiment of the present invention provides an earphone, wherein at least one external microphone for collecting ambient sound is disposed outside the earphone; the earphone casing is provided with a touch panel having a multi-touch function, and the touch panel is The main control chip of the earphone is connected; the noise reduction module and/or the monitoring module of the earphone are connected to the main control chip of the earphone, or integrated in the main control chip; the touchpad detects the movement track of the user's finger, and the main control a chip for confirming an adjustment type and an adjustment amount of the current ambient sound according to the movement trajectory, the adjustment type for the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the monitoring control; And, according to the confirmed adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone with the confirmed adjustment amount.

Preferably, when the touchpad detects that the user's palm covers the touchpad, the master chip controls the headset to enter a listening mode.

According to the technical solution of the embodiment of the present invention, a plurality of different gradual movement amounts are obtained by the movement of the regulator, and different movement amounts correspond to different adjustment amounts, so that the noise reduction amount or the monitored environment can be arbitrarily adjusted according to needs. The volume of the sound solves the problem that the existing solution can only turn on or off the noise reduction function alone, realizes the flexible adjustment of the earphone to the ambient sound, makes the earphone function more humanized, enhances the performance of the earphone, and improves the user. Experience.

In another aspect, an embodiment of the present invention provides a method for adjusting an ambient sound of a headset, the method comprising: setting at least one external microphone for collecting an ambient sound outside the earphone, the earphone comprising a noise reduction module and a monitoring module; and the receiving terminal The earphone application client sends an adjustment type and an adjustment amount to the ambient sound; the adjustment type to the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the monitoring control; The type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone by using the adjustment amount.

Preferably, when the earphone receives the command of entering the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the non-listening mode, recording the current playing volume of the earphone and the adjusting amount of the ambient sound, and controlling The monitoring module adjusts the ambient sound collected by the microphone by using a preset adjustment amount; when the earphone receives the command of the earphone application client sent by the terminal to leave the listening mode, according to the recorded playing volume and the ambient sound The adjustment amount controls the earphone to return to the original working state.

Preferably, when the earphone receives the command of entering the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the noise reduction playing audio mode or

simply playing the audio mode, the earphone is also controlled to adjust the playing volume to the advance The set playback volume.

In still another aspect, an embodiment of the present invention provides a terminal, where a headset application client is installed; the headset client provides a first virtual regulator on a interface thereof for adjusting a type and an adjustment amount of an ambient sound by a touch setting. And a second virtual adjuster that can be set to play the volume by touch; the type of adjustment to the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the monitor control.

Preferably, the first virtual regulator and/or the second virtual regulator are any of the following: a virtual knob, a virtual push button, and two virtual buttons.

Preferably, the first virtual regulator or the second virtual adjustment is two virtual keys; when the earphone detects that two virtual keys are simultaneously pressed, the earphone sends a command to enter the listening mode; the earphone detects When the two virtual buttons are simultaneously released, a command to leave the listening mode is sent to the earphone.

According to the technical solution of the embodiment of the present invention, the headset application client of the terminal adjusts the size of the noise reduction or the volume of the monitored ambient sound, thereby solving the problem that the existing solution can only selectively select the noise reduction function to be turned on or off. The flexible adjustment of the ambient sound of the earphone is realized, the function of the earphone is more humanized, the performance of the earphone is enhanced, and the user experience is improved.

Other features and advantages of the present invention will become apparent from the Detailed Description of the

DRAWINGS

The accompanying drawings, which are incorporated in FIG

FIG. 1 is a schematic diagram of a scenario of applying the solution according to an embodiment of the present invention;

2 is a schematic diagram of a method for adjusting ambient sound of a headset according to an embodiment of the present invention;

FIG. 3 is a schematic diagram of an implementation manner of a regulator according to another embodiment of the present invention; FIG.

FIG. 4 is a schematic diagram of another implementation manner of a regulator according to another embodiment of the present invention; FIG.

FIG. 5 is a schematic diagram of still another implementation manner of a regulator according to another embodiment of the present invention; FIG.

FIG. 6 is a schematic diagram of still another implementation manner of a regulator according to another embodiment of the present invention; FIG.

FIG. 7 is a schematic diagram of still another implementation manner of a regulator according to another embodiment of the present invention; FIG.

FIG. 8 is a block diagram of a hybrid active noise canceling earphone system implemented by feedforward and feedback according to another embodiment of the present invention; FIG.

FIG. 9 is a diagram showing experimental results of different noise reduction amounts of the earphone of the present scheme at different noise reduction levels; FIG.

FIG. 10 is a diagram showing experimental results of different monitoring control corresponding EQ curves implemented by the earphone of the present solution; FIG.

FIG. 11 is a schematic diagram of a method for triggering a listening mode according to an embodiment of the present invention; FIG.

FIG. 12 is a schematic diagram of an implementation manner of a listening mode triggering module according to an embodiment of the present invention;

FIG. 13 is a schematic diagram of another implementation manner of a listening mode triggering module according to another embodiment of the present invention.

Detailed ways

Various exemplary embodiments of the present invention will now be described in detail with reference to the drawings. It should be noted that the relative arrangement of the components and steps, numerical expressions and numerical values set forth in the embodiments are not intended to limit the scope of the invention unless otherwise specified.

The following description of the at least one exemplary embodiment is merely illustrative, and is in no way

Techniques, methods and apparatus known to those of ordinary skill in the relevant art may not be discussed in detail, but the techniques, methods and apparatus should be considered as part of the specification, where appropriate.

In all of the examples shown and discussed herein, any specific values are to be construed as illustrative only and not as a limitation. Thus, other examples of the exemplary embodiments may have different values.

It should be noted that similar reference numerals and letters indicate similar items in the following figures, and therefore, once an item is defined in one figure, it is not required to be further discussed in the subsequent figures.

One embodiment of the present invention provides a method for adjusting the ambient sound of a headset. Referring to FIG. 1, a scenario in which the present solution is applied is illustrated. The method includes: setting at least one external microphone for collecting ambient sound outside the earphone, such as MIC1 connected to the noise reduction module and MIC2 connected to the monitoring module. In a variant, the same external microphone can be used to connect the noise reduction module and the monitoring module respectively to achieve noise reduction and monitoring. A mechanically movable regulator is provided on the earphone housing, the regulator having an electrical port that connects the regulator to the noise reduction module and/or the monitoring module of the earphone through the electrical port. The noise reduction module and the monitoring module can be integrated on the main control chip of the earphone, implemented by one chip, or separately set discrete devices in the earphone to implement the noise reduction module and the monitoring module.

Referring to FIG. 2, the method for adjusting the ambient sound of the earphone provided by the embodiment further includes:

S201: The main control chip monitors the movement state of the regulator in real time, and when the regulator moves, the movement amount and the movement type of the regulator are known.

S202: The main control chip confirms the current adjustment type according to the movement type, and confirms the current adjustment amount of the ambient sound according to the movement amount.

S203: The main control chip controls the noise reduction module or the monitoring module to adjust the ambient sound collected by the microphone according to the confirmed adjustment type.

When communicating in a noisy background, the headphone will use a noise reduction module. The noise reduction module uses feed-forward (FF) noise reduction, feedback (FeedBack, FB) noise reduction or feedforward plus feedback hybrid noise reduction. The current noise reduction mainly considers the following two aspects: on the one hand, the acoustic signal processing technology is used at the transmitting end of the communication device to improve the signal-to-noise ratio of the voice signal picked up by the microphone, so that the remote user can hear the speech of the near-end user. On the other hand, it is necessary to improve the

signal-to-noise ratio of the voice of the receiving end at the receiving end of the communication device, so that the near-end can hear the voice signal sent by the remote user.

In other scenarios, in order to avoid talking to the outside world when the headset is not needed, or to monitor the external environment noise, the headset will use the monitor module to amplify the sound in the external environment to ensure the wearer. Can hear the outside world. The existing earphones with monitoring function only have two ways of opening the monitoring function and turning off the monitoring function. This direct realization of the monitoring function can not bring a good user experience to the earphone wearer.

It should be noted that the functions of noise reduction and monitoring are actually the opposite two effects, so the noise reduction module and the monitoring module involved in the present invention may be the same physical entity. For example, the noise reduction module and the monitoring module may be an ANC. (active noise concellation) An active noise reduction chip, or an ANC system circuit.

The above steps S201 to S203 can also be implemented by the noise reduction module or the monitoring module, and the movement of the regulator can be monitored by the noise reduction module or the monitoring module.

This embodiment can be applied to a scenario in which only one of the noise reduction module and the monitoring module exists in the earphone. When the regulator moves, only one type of movement is corresponding, for example, when only the noise reduction module exists, if When the regulator moves, the type of movement of the regulator corresponds only to the type of adjustment of the noise reduction.

According to the technical solution of the embodiment of the present invention, a plurality of different gradual movement amounts are obtained by the movement of the regulator, and different movement amounts correspond to different adjustment amounts, so that the noise reduction amount or the monitored environment can be arbitrarily adjusted according to needs. The volume of the sound solves the problem that the existing solution can only turn on or off the noise reduction function alone, realizes the flexible adjustment of the earphone to the ambient sound, makes the earphone function more humanized, enhances the performance of the earphone, and improves the user. Experience.

On the basis of the embodiment shown in FIG. 2, further, the embodiment of the present invention further includes: dividing the movement state of the regulator into a first movement type corresponding to the noise reduction control according to the movement behavior of the regulator, and corresponding The second type of movement for the listener control. Wherein, the noise reduction module is used to perform noise reduction control, and the monitoring module is used to perform monitoring control.

Depending on the specific implementation of the regulator, the movement behavior of the regulator is also different. For example, when the regulator is a knob, the movement behavior of the regulator may include rotating to the left and rotating to the right.

In this embodiment, the two states of the adjustment are divided into two types of movements, so that the same regulator is used to implement the functions of adjusting noise reduction and adjusting the monitoring. This mainly considers that the function of noise reduction and monitoring is actually the opposite effect. The same structural component is used to realize the adjustment of the two functions, which can simplify the structure of the earphone while ensuring the function of the earphone. This embodiment mainly describes a scenario in which a regulator is set as an example, but is not limited thereto. In this embodiment, two regulators may be provided, one regulator for implementing noise reduction control and one regulator for monitoring control. .

The method further includes: during a monitoring process, the master chip can perform one of the following operations:

When the main control chip detects that the movement of the regulator belongs to the first movement type, confirms that the current adjustment type is noise reduction control, and confirms the current noise reduction amount of the ambient sound according to the movement amount;

When the main control chip detects that the movement of the regulator belongs to the second movement type, confirm that the current adjustment type is the monitoring control, and confirm the current amplification of the ambient sound according to the movement amount.

When two regulators are provided on the earphone, the master chip can perform the above two operations simultaneously for the movement of the two regulators, or perform only one of the above operations.

The main control chip can pre-maintain a correspondence table of the movement amount and the adjustment amount, and query the correspondence between the movement amount and the adjustment amount according to the movement amount, thereby quickly obtaining the current adjustment amount of the environmental sound. In this way, when monitoring, there is no need to calculate the noise reduction amount or the amplification amount in real time, and the current noise reduction amount or amplification amount can be quickly obtained, and the speed of the earphone to adjust the ambient sound can be improved.

Another embodiment of the present invention mainly describes the operation mode of the regulator. Other contents of the earphone environment sound adjustment scheme can be referred to other embodiments of the present invention.

When the adjuster is realized by the knob, a rotation demarcation point is set on the rotation track of the regulator. For example, if the middle point of the rotation track is used as the rotation demarcation point, the main control chip (or the noise reduction module) monitors the predetermined one of the knob to the rotation demarcation point. When the side (such as the left side) rotates, it is known that the movement type of the knob is the first movement type, and the master chip (or the monitoring module) detects that the knob rotates to the other side of the rotation demarcation point (such as the right side), and the knob is known. The movement type is the second movement type, and the amount of movement is obtained by the amount of rotation of the knob. Referring to Figure 3, one implementation of the regulator is shown. The adjuster is a knob, which is coaxially assembled with the earpiece portion of the headset, and a adjuster is provided for each earpiece. The adjuster is located at the outer end of the earpiece housing, and one adjuster is responsible for adjusting the ambient sound of the earphone on the same side. . The knob provided as the adjuster has a certain size and thickness for easy grip by a hand, for example, the adjuster can be a disc member having a certain thickness.

Alternatively, referring to Figure 4, another implementation of the regulator is shown. The adjuster is a knob that protrudes to a certain height outside the earpiece housing for easy grip by a human hand, and the knob is a cylindrical member.

Alternatively, referring to Fig. 5, another implementation of the adjuster is illustrated, the adjuster being a rotatable panel structure disposed on the outer casing of the earpiece earpiece.

When the adjuster is realized by the push button, the stroke demarcation point is set on the push stroke of the adjuster, and when the push button is slid to the predetermined side (such as the upper side) of the stroke demarcation point, it is known that the movement type of the push button is the first movement type. When it is detected that the push button slides to the other side of the stroke demarcation point (the following side), it is known that the movement type of the push button is the second movement type, and the movement amount is obtained by the sliding distance of the push button. Referring to Fig. 6, there is shown a schematic view of the adjuster being a push button, the push button being mounted in the sliding groove, and the push button sliding up and down along the sliding groove.

When the regulator is implemented by two buttons, when the first button is pressed, it is known that the movement type of the button is the first movement type, and when the second button is pressed, it is known that the movement type of the button is the second movement type. And the amount of movement is obtained by the number of presses of the button. Referring to Figure 7, a schematic diagram of the regulator being a button is shown, including a first button and a second button that can be pushed up and down.

In one scene, if the adjuster is a knob, you can set the knob to be in the middle position, and the ambient sound will not be amplified or reduced. If you turn to the left, the noise reduction will be achieved. The more the left rotation, the more noise reduction. , the rotation to the leftmost position is the largest amount of noise reduction; the setting knob is rotated to the right to monitor the ambient sound amplification, the more the rightward rotation, the greater the monitoring sound, the rotation to the rightmost position, the monitor sound amplification is maximized, the same push button And the button can achieve the same function.

In addition, regardless of the manner of rotating components (such as knobs, rotating disks, rotatable panel structures, etc.), pushing or pressing, or in addition to the above-mentioned mechanical structure keys, it is also possible to use a proximity-sensitive touch key. To achieve the rotation adjustment, up and down sliding adjustment and up and down button adjustment function.

The above mainly describes the regulator as an example of a headset. For a headset or an in-ear earphone, the regulator can also be set on the line controller. For the specific setting method, refer to the regulator of the headset.

Another embodiment of the present invention mainly describes an adjustment scenario of the noise reduction control. Other contents of the earphone environment sound adjustment scheme may be referred to other embodiments of the present invention.

As shown in FIG. 8, a block diagram of a hybrid active noise canceling earphone system when the noise reduction module is implemented by feedforward and feedback is shown. The maximum noise reduction and noise reduction bands of the system are superimposed on the noise reduction of feedback (FB active noise reduction control ANC) and feedforward (FF ANC). The basic noise reduction performance is shown in Figure 8. The noise reduction module is at low frequency 1000Hz. With active noise reduction, the maximum active noise reduction can reach about 25db. With the passive sound insulation of the earmuffs or earplugs, the noise canceling headphones can provide almost 20db noise reduction effect in the entire frequency band.

The noise reduction/amplification can be achieved by adjusting the circuit or by modifying the register in software to achieve different levels of noise reduction/monitoring. The user can manually adjust the noise reduction/monitoring effect as needed.

For example, when the MCU (Micro Control Unit) is used to implement the noise reduction module and/or the monitoring module, if the main control chip is an MCU, the MCU control MCU register write value is modified by the confirmed adjustment amount to change the drop. Noise or amplification. The noise reduction module or the monitoring module will amplify and compensate the signal picked up by the microphone (such as FF/FB noise reduction MIC1 or monitor MIC2) placed in the noise of the external environment through a certain EQ (equalizer), and then output to the earphone speaker. The headset wearer hears. Through EQ processing, the effect of weakening the low frequency to raise the middle and high frequency can be achieved, so that the sound output by the earphone is more in line with the human ear hearing effect.

The beneficial effects of the present solution will be more clearly explained below in conjunction with the experimental results.

Referring to Fig. 9, there is shown an experimental result diagram of different noise reduction amounts realized by the earphone to which the present scheme is applied.

FIG. 9 is a schematic diagram showing the different noise reduction levels of the earphone of the present solution at different noise reduction levels, which is an implementation manner of the solution, wherein the noise reduction amount and the noise reduction frequency band required at different levels can be performed as needed. Different adjustments to achieve a better user experience. Among them, the total level of the noise reduction level can also be designed according to actual needs. In the manner shown in FIG. 9, the earphone of the present solution can implement ten different noise reduction levels according to different movement amounts.

Referring to FIG. 10, an experimental result diagram of different monitor control corresponding EQ (equalizer) curves implemented by the earphone of the present scheme is shown. This is an implementation manner of the solution. In the mode shown in FIG. 10, the earphone of the solution can implement ten different monitoring levels according to different movement amounts. Among them, the total level of the monitoring level, the magnification corresponding to each level, and the EQ curve can be designed according to different headphones or different needs to achieve a better user experience.

According to the technical solution of the embodiment of the present invention, the movement of the regulator may result in a plurality of different gradations (each gradual change may be equal or unequal), and the different movement amounts correspond to different adjustment amounts, thereby being able to Arbitrarily adjust the amount of noise reduction or the volume of the monitored ambient sound, which solves the problem that the existing solution can only turn on or off the noise reduction function alone, and realizes the flexible adjustment of the earphone to the ambient sound, so that the earphone function More user-friendly, enhanced headphone performance and improved user experience.

Yet another embodiment of the present invention provides an earphone. The outside of the earphone is provided with at least one external microphone for collecting ambient sound, the earphone casing is provided with a mechanical movable adjuster, the adjuster has an electrical port, the adjuster is connected to the noise reduction module of the earphone through the electrical port and/or the monitor Module.

The noise reduction module and/or the monitoring module are connected to the main control chip of the earphone, or integrated in the main control chip; the main control chip monitors the movement state of the regulator, and when the regulator moves, the movement amount and movement of the regulator are known. Type; the main control chip confirms the current adjustment type according to the movement type, and confirms the current adjustment amount of the ambient sound according to the movement amount; the main control chip controls the noise reduction module or the monitoring module to use the confirmed adjustment amount to the microphone according to the confirmed adjustment type The collected ambient sound is adjusted.

When the adjuster is realized by the knob, a rotation demarcation point is set on the rotation track of the adjuster, and when the main control chip detects that the knob rotates to a predetermined side of the rotation demarcation point, it is learned that the movement type of the knob is the first movement type, and the knob is monitored. When rotating to the other side of the rotation demarcation point, it is learned that the movement type of the knob is the second movement type, and the amount of movement is obtained by the rotation amount of the knob;

When the regulator is implemented by the push button, the stroke demarcation point is set on the push stroke of the adjuster, and when the main control chip detects that the push button slides to the predetermined side of the stroke demarcation point, it is known that the movement type of the push button is the first movement type. When it is detected that the push button slides to the other side of the stroke demarcation point, it is learned that the movement type of the push button is the second movement type, and the movement amount is obtained by the sliding distance of the push button;

When the regulator is implemented by two buttons, when the first button is pressed, the master chip knows that the type of movement of the button is the first type of movement, and when the second button is pressed, it is known that the type of movement of the button is The second type of movement, and the amount of movement is obtained by the number of presses of the button.

When the noise reduction module or the monitoring module is separated from the active chip, the operation performed by the main control chip in the above-mentioned earphone may also be implemented by the noise reduction module or the monitoring module.

Further, the adjuster is disposed on the housing of the earpiece portion of the headset, and the knob-implemented adjuster can be coaxially assembled or assembled in an off-axis manner with the earpiece portion.

In many applications, users want to quickly switch from noise reduction and listening mode to listening to ambient sound mode. After a while, they need to quickly return to the previous noise reduction mode and listening state, such as wearing The person is listening to music with the noise reduction function. Someone needs to communicate with the wearer for a short time. If you adjust the noise reduction mode to the monitor mode by rotating the adjustment method, it may take a little time. After the communication is finished, adjust it to The previous noise reduction level also needs to consume a certain time, so the present invention also provides a technical solution capable of realizing the requirement of such a fast short-time switching to the listening mode:

Referring to FIG. 11 , an implementation manner of a trigger listening mode according to an embodiment of the present invention is provided. The monitoring mode triggering module is set on the earphone shell, and the following steps are further included:

S301: Real-time monitoring data output by the monitoring mode triggering module, and determining, according to the data output by the monitoring mode triggering module, whether the triggering condition of the monitoring mode is satisfied;

S302: When the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the non-listening mode, recording the current playing volume of the earphone and the adjusting amount of the ambient sound, and controlling the monitoring module to adopt the preset The adjusted amount adjusts the ambient sound collected by the microphone; when the trigger condition of the listening mode

changes from satisfied to not satisfied, the earphone is restored to the original working state according to the recorded playing volume and the adjusting amount of the ambient sound. .

In step S302, when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the listening mode, the current listening state may be maintained without processing.

For headphones with both noise reduction module and monitoring module, there are five working modes: simple playback audio mode, simple noise reduction mode, noise reduction playback audio mode, simple listening mode, and monitor playback audio mode. . The simple noise reduction mode is suitable for users to keep quiet in a noisy environment. In this mode, the headphones will not play music and other content, but only the ambient sound is denoised, that is, the playback volume is zero. The simple play audio mode refers to a mode in which the earphone only plays music and the like, and does not make any adjustment to the external environment sound (corresponding to the knob mentioned above in the middle position), that is, the adjustment amount to the external environment sound is zero.

The method further includes: when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, it is also required to control the playing volume adjustment of the earphone to preset Play volume. This situation is more suitable for the user to be playing noise reduction music or simply playing music, and still want to continue playing music after triggering into the monitoring mode, in addition to controlling the monitoring module to adjust the ambient sound according to a preset amount of amplification, The music is also continuously played according to the preset playback volume, and the preset playback volume is usually a lower volume so that the user can clearly hear the ambient sound.

The monitoring mode triggering module includes any one of the following: a capacitive touch sensing switch, a resistive touch sensing switch, a proximity sensing switch, a pressure sensing switch, and a mechanical switch. The proximity sensor switch is, for example, an infrared proximity sensor switch. Mechanical switches are, for example, push button mechanical switches, knob mechanical switches, push button mechanical switches. The monitor mode trigger module can also be a rotatable panel switch or a single metal touch switch. Among them, the single-metal touch switch, the capacitive touch sensor switch, the resistive touch sensor switch, and the proximity sensor switch can realize the monitor mode trigger module on the palm cover of the earphone casing, and can enter from the non-listening mode. In the monitor mode, the palm can automatically return to the original working state after leaving the earphone casing.

Correspondingly, the present invention also provides an earphone with a monitoring mode triggering module, wherein the earphone housing is further provided with a monitoring mode triggering module; the main control chip is further configured to monitor the monitoring mode triggering module in real time. The output data determines whether the trigger condition of the monitor mode is satisfied according to the data output by the monitor mode trigger module; and, when the trigger condition of the monitor mode is changed from unsatisfied to satisfied, if the headset is currently in the non-listening mode, the record is The current playing volume of the earphone and the adjustment amount of the ambient sound, and controlling the monitoring module to adjust the ambient sound collected by the microphone by using a preset adjustment amount; when the triggering condition of the listening mode is changed from satisfying to not satisfied The earphone is restored to the original working state according to the recorded playing volume and the amount of adjustment to the ambient sound.

The master chip is further configured to continue to maintain the current listening state if the headset is currently in the listening mode when the trigger condition of the listening mode is changed from unsatisfied to satisfied.

The main control chip is further configured to: when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, the playing volume adjustment of the earphone is also controlled. Go to the preset playback volume.

The monitoring mode triggering module includes any one of the following: a capacitive touch sensor, a resistive touch sensor, an infrared distance sensor, a pressure sensor, and a mechanical switch. The monitor mode trigger module can be, for example, a single metal chip touch detection capacitor.

The monitoring mode triggering module is a touchpad with multi-touch function, and when the user's palm covers most of the area of the touchpad or the entire touchpad, the listening mode is triggered.

The invention provides a monitoring mode triggering module arranged on the earphone casing, and the monitoring mode triggering module can directly switch from the noise reduction listening mode to the listening mode, and when the triggering condition disappears, the automatic noise reduction level can be automatically and quickly returned. And playing the listening state, the invention can not only meet the user's need for progressive control and adjustment of the external environment sound, but also provides a quick function of quickly entering the monitoring and returning to the noise reduction playing state, providing a more user-friendly user experience and use. Feel, and a more complete way to control your headphones.

Referring to FIG. 12, an implementation manner of a monitor mode trigger module according to an embodiment of the present invention is provided. The monitor mode trigger module is a single metal chip touch detection capacitor disposed on a housing of an earpiece of the headset. The adjuster is disposed on the housing of the other earpiece portion of the headset.

The action of the user's palm touching or leaving the earphone casing may cause the capacitance value of the single metal chip type touch detection capacitor to change, real-time monitoring the capacitance value of the single metal chip type touch detection capacitance output, and judging the capacitance value of the single metal chip type touch detection capacitance output. Whether the threshold is greater than or equal to the threshold, if the threshold is greater than or equal to the threshold, it is determined that the trigger condition of the listening mode is satisfied, otherwise, the trigger condition for determining the listening mode is not satisfied.

When the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the non-listening mode, the current playing volume of the earphone and the adjustment amount of the ambient sound are recorded, and the monitoring module adopts a preset adjustment amount to the microphone. The collected ambient sound is adjusted; when the trigger condition of the listening mode is changed from satisfied to unsatisfied, the earphone is restored to the original working state according to the recorded playing volume and the adjustment amount to the ambient sound.

The invention provides that when the palm covers the monitor mode triggering module on the earphone casing, the mode can be directly switched from the noise reduction listening mode to the listening mode, thereby quickly entering the listening state, and the palm can automatically return quickly after leaving the earphone casing. The noise reduction level and the playing sound state, the invention not only satisfies the user's need for progressive control and adjustment of the external environment sound, but also provides a quick function of quickly entering the monitoring and returning the noise reduction playing state, and provides a more humanized function. User experience and usage experience, as well as improved headset control.

Referring to FIG. 13 , another implementation manner of the monitor mode trigger module according to another embodiment of the present invention is: the monitor trigger module is a mechanical knob switch disposed on the outer casing of the earpiece portion of the earphone, and the earpiece housing is protruded. The outside is a certain height so that it can be held by human hands.

12 and FIG. 13 mainly illustrate a monitor trigger module by using a headset as an example. For an in-ear earphone or a headset, the monitor trigger module can also be disposed on the housing of the online controller.

The invention also provides a method for adjusting the ambient sound of the earphone, wherein at least one external microphone for collecting ambient sound is disposed outside the earphone; a touch panel with multi-touch function is provided on the earphone casing, and the touch panel and the earphone are provided. The main control chip is connected; the noise reduction module and/or the monitoring module of the earphone are connected to the main control chip of the earphone, or integrated in the main control chip;

The touchpad detects the movement trajectory of the user's finger, and the main control chip confirms the current adjustment type and adjustment amount of the ambient sound according to the movement trajectory, and the adjustment type of the environmental sound includes the first type corresponding to the noise reduction control, and corresponds to the monitoring control. Second type;

The main control chip controls the noise reduction module or the monitoring module to adjust the ambient sound collected by the microphone according to the confirmed adjustment type.

For example, when it is detected that the user's finger moves clockwise, the main control chip confirms that it should be the first type of noise reduction control, and confirms the magnitude of the noise reduction amount according to the length or angle of the movement trajectory. Conversely, for example, when it is detected that

the user's finger moves counterclockwise, the master chip confirms that it should be the second type of the monitor control, and confirms the amount of amplification of the ambient sound collected by the microphone according to the length or angle of the movement trajectory.

Preferably, when the touchpad detects that the user's palm covers the touchpad, the master chip controls the headset to enter the listening mode.

The invention also provides an earphone, wherein the outer earphone is provided with at least one external microphone for collecting ambient sound; the earphone casing is provided with a touch panel having a multi-touch function, and the touch panel is connected with the main control chip of the earphone; The noise reduction module and/or the monitoring module of the earphone are connected to the main control chip of the earphone, or integrated in the main control chip;

The touchpad detects a movement trajectory of the user's finger, and the main control chip is configured to confirm the current adjustment type and the adjustment amount of the ambient sound according to the movement trajectory, and the adjustment type for the environmental sound includes a first type corresponding to the noise reduction control, and corresponding The second type of the monitoring control; and, according to the confirmed adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone with the confirmed adjustment amount.

Preferably, when the touchpad detects that the user's palm covers the touchpad, the main control chip controls the earphone to enter the listening mode. Otherwise, when the touchpad detects that the user's palm covers the touchpad and leaves, the main control chip controls the earphone to leave the listening mode. , return to the previous working state.

In this way, a touchpad with multi-touch function can be used to gradually adjust the amount of noise reduction of the earphone or the volume of the ambient sound that is monitored, as well as quickly enter the listening mode and return to the noise reduction playing state. The quick function provides a more user-friendly experience and usage experience, as well as a more complete way of controlling the headphones.

The above mentioned method for adjusting the ambient sound of the earphone, the regulator, the method for triggering the listening mode, and the monitoring trigger module can all be implemented by the headset application client of the terminal:

The present invention provides a method for adjusting the ambient sound of a headphone, wherein at least one external microphone for collecting ambient sound is disposed outside the earphone, and the earphone includes a noise reduction module and/or a monitoring module;

Receiving, by the earphone application client of the terminal, an adjustment type and an adjustment amount of the ambient sound; the adjustment type of the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the monitoring control; The adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone by using the adjustment amount; and the playing volume sent by the earphone application client of the receiving terminal, and sets the power amplifier of the earphone according to the playing volume. .

Further, when the headset receives the command of entering the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the non-listening mode, recording the current playing volume of the earphone and the adjusting amount of the ambient sound, and controlling The monitoring module adjusts the ambient sound collected by the microphone by using a preset adjustment amount; when the earphone receives the command of the earphone application client sent by the terminal to leave the listening mode, according to the recorded playing volume and the ambient sound The adjustment amount controls the earphone to return to the original working state.

When the earphone receives the command of entering the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, the earphone is also controlled to adjust the playing volume to a preset value. Play volume.

The invention also provides a terminal, which is provided with a headset application client; the headset client provides a first virtual regulator on the interface for adjusting the type and adjustment amount of the ambient sound by the touch setting, and the touchable A second virtual adjuster that sets a play volume; the type of

adjustment to the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the snoop control.

The first virtual regulator and/or the second virtual regulator are any of the following: a virtual knob, a virtual push button, and two virtual buttons.

The first virtual regulator or the second virtual adjustment is two virtual keys; when the earphone detects that two virtual keys are simultaneously pressed, the earphone sends a command to enter the listening mode; the earphone detects two virtual keys When the button is released at the same time, a command to leave the listening mode is sent to the earphone.

It should be noted that the mechanical movable regulator and the monitoring trigger module mentioned in the present invention can coexist and coordinate with the above-mentioned earphone application client installed in the terminal, and those skilled in the art on the basis of the present invention This can be set up and will not be described here.

For the specific working mode of the device in the embodiment of the present invention, reference may be made to the method embodiment of the present invention, and details are not described herein again.

From the above, the application has at least the following advantages:

1、The use of mechanical components to achieve the regulator, to meet the noise reduction and monitoring of both functions have manual adjustment level of user needs, and can use the same mechanical components to achieve the regulator, to achieve a powerful headset with a simple structure;

2、The mechanical structure of the regulator can be implemented by means of a knob, a push button, two independent buttons or any other structure that can realize the function, and the implementation is flexible;

3、Using the knob, push button or up and down button, the user can adjust the amount of noise reduction or monitor the amount of ambient sound according to the need, avoiding the existing single can only select the noise reduction function on or off, and the monitoring function is on or off. Bad use, providing a more user-friendly experience and experience.

4、It provides fast access to the monitor and return to the noise reduction playback state, which is well adapted to the user's temporary and sudden monitoring needs, thus providing a better user experience and better control of the headset.

5、A regulator with a mechanically adjustable structure and a virtual regulator provided by the terminal are provided, both of which can achieve the object of the present invention, and can also be used in combination.

Those skilled in the art should understand that the above unit or module can be implemented in various manners. For example, the screen creation module, the environment model parsing loading module, the matching module, and the 3D video processing module can be implemented by an instruction configuration processor. Each of the above modules may be independent units or may be implemented together.

The invention can be a system, method and/or computer program product. The computer program product can comprise a computer readable storage medium having computer readable program instructions embodied thereon for causing a processor to implement various aspects of the present invention. Computer program instructions for performing the operations of the present invention may be assembly instructions, instruction set architecture (ISA) instructions, machine instructions, machine related instructions, microcode, firmware instructions, state setting data, or in one or more programming languages. Source code or object code written in any combination, including object oriented programming languages such as Smalltalk, C++, etc., as well as conventional procedural programming languages such as the "C" language or similar programming languages. The computer readable program instructions can execute entirely on the user's computer, partly on the user's computer, as a stand-alone software package, partly on the user's computer, partly on the remote computer, or entirely on the remote computer or server. carried out.

Aspects of the present invention are described herein with reference to flowchart illustrations and/or block diagrams of methods, apparatus, and computer program products according to embodiments of the invention. It

will be understood that each block of the flowchart illustrations and/or block diagrams, and combinations of blocks in the flowcharts and/or block diagrams can be implemented by computer readable program instructions.

The flowchart and block diagrams in the Figures illustrate the architecture, functionality and operation of possible implementations of systems, methods and computer program products according to various embodiments of the invention. In this regard, each block in the flowchart or block diagram can represent a module, a program segment, or a portion of an instruction that includes one or more Executable instructions. In some alternative implementations, the functions noted in the blocks may also occur in a different order than those illustrated in the drawings. For example, two consecutive blocks may be executed substantially in parallel, and they may sometimes be executed in the reverse order, depending upon the functionality involved. It is also noted that each block of the block diagrams and/or flowcharts, and combinations of blocks in the block diagrams and/or flowcharts, can be implemented in a dedicated hardware-based system that performs the specified function or action. Or it can be implemented by a combination of dedicated hardware and computer instructions. It is well known to those skilled in the art that implementation by hardware, implementation by software, and implementation by a combination of software and hardware are all equivalent.

While the invention has been described in detail with reference to the preferred embodiments of the present invention, it is understood that It will be appreciated by those skilled in the art that the above embodiments may be modified without departing from the scope and spirit of the invention. The scope of the invention is defined by the appended claims.

CN105554610B Method for regulating environmental sound of earphone and earphone

Data originating from sources other than the EPO may not be accurate, complete, or up to date.

The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

1.A method for adjusting ambient sound of a headphone, characterized in that at least one external microphone for collecting ambient sound is disposed outside the earphone, a mechanical movable adjuster is disposed on the earphone casing, or a shell of the wire controller of the earphone a mechanically movable adjuster is provided on the body;

The regulator has an electrical port, and the regulator is connected to the noise reduction module and/or the monitoring module of the earphone through the electrical port, the method comprising:

Real-time monitoring of the movement state of the regulator, when the regulator moves, knowing the amount of movement and the type of movement of the regulator;

Confirm the current adjustment type according to the movement type, and confirm the current adjustment amount of the ambient sound according to the movement amount;

According to the confirmed adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone by using the confirmed adjustment amount;

Wherein, the confirming the current adjustment amount of the ambient sound according to the amount of movement includes:

According to the correspondence between the movement amount and the adjustment amount, the current adjustment amount of the ambient sound is obtained.

2.The method according to claim 1, wherein the method further comprises: dividing the movement state of the regulator into a first movement type corresponding to the noise reduction control according to the movement behavior of the regulator, and corresponding to the monitoring The second type of movement controlled;

Then, the current adjustment type is confirmed according to the movement type, and the current adjustment amount of the ambient sound is confirmed according to the movement amount, including:

When it is detected that the movement of the regulator belongs to the first movement type, confirm that the current adjustment type is noise reduction control, and confirm the current noise reduction amount of the ambient sound according to the movement amount; and/or,

When it is detected that the movement of the regulator belongs to the second movement type, it is confirmed that the current adjustment type is the monitor control, and the current amplification amount of the ambient sound is confirmed according to the movement amount.

3.The method according to claim 2, wherein the real-time monitoring of the movement state of the regulator, when the regulator moves, knowing the amount of movement and the type of movement of the regulator include:

When the regulator is realized by the rotating member, a rotation demarcation point is set on the rotation trajectory of the regulator, and when the rotation member is rotated to a predetermined side of the rotation demarcation point, it is learned that the movement type of the rotation member is the first movement type, and the rotation is monitored. When the member is rotated to the other side of the rotation demarcation point, it is learned that the movement type of the rotation member is the second movement type, and the amount of movement is obtained from the amount of rotation of the rotation member.

4.The method according to claim 2, wherein the real-time monitoring of the movement state of the regulator, when the regulator moves, knowing the amount of movement and the type of movement of the regulator include:

When the adjuster is realized by the push button, the stroke demarcation point is set on the push stroke of the adjuster, and when the push button is slid to the predetermined side of the stroke demarcation point, it is known that the movement type of the push button is the first movement type, and the push is monitored. When the button slides to the other side of the stroke demarcation point, it is known that the movement type of the push button is the second movement type, and the movement amount is obtained by the sliding distance of the push button.

5.The method according to claim 2, wherein the real-time monitoring of the movement state of the regulator, when the regulator moves, knowing the amount of movement and the type of movement of the regulator include:

When the regulator is implemented by two buttons, when the first button is pressed, it is known that the movement type of the button is the first movement type, and when the second button is pressed, it is known that the movement type of the button is the second movement type. And the amount of movement is obtained by the number of presses of the button.

6.The method of claim 2 wherein:

When the noise reduction module and/or the monitoring module are implemented by using the MCU, the written value of the MCU control register is modified by using the confirmed adjustment amount, so that the noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone.

7.The method of claim 2 wherein:

The adjuster is disposed on a housing of the earpiece portion of the headset.

8.The method according to any one of claims 1 to 7, wherein a listening mode triggering module is disposed on the earphone casing, or a monitoring triggering module is disposed on a casing of the wire controller of the earphone; the method further comprises :

The data output by the monitoring mode triggering module is monitored in real time, and the triggering condition of the monitoring mode is determined according to the data output by the monitoring mode triggering module;

When the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the non-listening mode, the current playing volume of the earphone and the adjustment amount of the ambient sound are recorded, and the monitoring module is controlled to adopt a preset The adjustment amount adjusts the ambient sound collected by the microphone; when the trigger condition of the listening mode changes from satisfied to not satisfied, the earphone is restored to the original working state according to the recorded playing volume and the adjustment amount of the ambient sound.

9.The method according to claim 8, wherein the method further comprises: if the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, The playback volume of the earphone is controlled to be adjusted to a preset playback volume.

10.The method according to claim 8, wherein the monitor mode triggering module comprises any one of the following: a rotatable panel switch, a touchpad having a single or multi-touch function.

11.The method according to claim 8, wherein the monitor mode triggering module comprises any one of the following: a capacitive touch sensor, a resistive touch sensor, an infrared distance sensor, a pressure sensor, and a mechanical switch.

12.An earphone, wherein the outer side of the earphone is provided with at least one external microphone for collecting ambient sound, the earphone casing is provided with a mechanical movable adjuster, or the outer casing of the earphone is provided with a casing Mechanically movable regulator;

The regulator has an electrical port, the regulator is connected to the noise reduction module and/or the monitoring module of the earphone through an electrical port; the noise reduction module and/or the monitoring module is connected to the main control chip of the earphone, or is integrated in In the main control chip;

The main control chip monitors the movement state of the regulator, and when the regulator moves, the movement amount and the movement type of the regulator are known;

The main control chip confirms the current adjustment type according to the movement type, and confirms the current adjustment amount of the environmental sound according to the movement amount;

The main control chip controls the noise reduction module or the monitoring module to adjust the ambient sound collected by the microphone according to the confirmed adjustment type;

Wherein, the confirming the current adjustment amount of the ambient sound according to the amount of movement includes:

According to the correspondence between the movement amount and the adjustment amount, the current adjustment amount of the ambient sound is obtained.

13.The earphone according to claim 12, wherein

When the regulator is realized by the rotating component, a rotation demarcation point is set on the rotation track of the regulator, and when the main control chip detects that the rotating component rotates to a predetermined side of the rotation demarcation point, it is known that the movement type of the rotating component is the first movement type. When it is detected that the rotating component rotates to the other side of the rotational demarcation point, it is learned that the moving type of the rotating component is the second moving type, and the moving amount is obtained by the rotating amount of the rotating component;

When the regulator is implemented by the push button, the stroke demarcation point is set on the push stroke of the adjuster, and when the main control chip detects that the push button slides to the predetermined side of the stroke demarcation point, it is known that the movement type of the push button is the first movement type. When it is detected that the push button slides to the other side of the stroke demarcation point, it is learned that the movement type of the push button is the second movement type, and the movement amount is obtained by the sliding distance of the push button;

When the regulator is implemented by two buttons, when the first button is pressed, the master chip knows that the type of movement of the button is the first type of movement, and when the second button is pressed, it is known that the type of movement of the button is The second type of movement, and the amount of movement is obtained by the number of presses of the button.

14.The earphone according to claim 12, wherein

The adjuster is disposed on a housing of the earpiece portion of the headset.

15.The earphone according to any one of claims 12 to 14, wherein the earphone casing is further provided with a monitoring mode triggering module, or the casing of the earphone of the earphone is provided with a monitoring triggering module;

The main control chip is further configured to monitor data output by the monitoring mode triggering module in real time, determine whether the triggering condition of the monitoring mode is satisfied according to the data output by the monitoring mode triggering module; and, when the triggering condition of the monitoring mode is not When the satisfaction becomes satisfied, if the earphone is currently in the non-listening mode, the current playing volume of the earphone and the adjustment amount of the ambient sound are recorded, and the monitoring module is controlled to adopt an preset adjustment amount to the environment collected by the microphone. The sound is adjusted; when the trigger condition of the listening mode is changed from being satisfied to not satisfied, the earphone is returned to the original working state according to the recorded playing volume and the amount of adjustment to the ambient sound.

16.The earphone according to claim 15, wherein the main control chip is further configured to: when the trigger condition of the listening mode is changed from unsatisfied to satisfied, if the earphone is currently in a noise reduction playing audio mode or simply playing audio The mode also controls the playback volume of the headset to be adjusted to a preset playback volume.

17.The earphone according to claim 15, wherein the monitoring mode triggering module comprises any one of the following: a rotatable panel switch, a touchpad having a single or multi-touch function.

18.The earphone according to claim 15, wherein the listening mode triggering module comprises any one of the following: a capacitive touch sensor, a resistive touch sensor, an infrared distance sensor, a pressure sensor, and a mechanical switch.

19.A method for adjusting an ambient sound of a headphone, wherein at least one external microphone for collecting ambient sound is disposed outside the earphone; and a touch panel having a multi-touch function is disposed on the earphone casing, the touch pad and the earphone The main control chip is connected; the noise reduction module and/or the monitoring module of the earphone are connected to the main control chip of the earphone, or integrated in the main control chip;

The touch panel detects a movement trajectory of a user's finger, and the main control chip confirms an adjustment type and an adjustment amount of the current ambient sound according to the movement trajectory, and the adjustment type for the ambient sound includes a corresponding to the noise reduction control. a type, and a second type corresponding to the snoop control;

The main control chip controls the noise reduction module or the monitoring module to adjust the ambient sound collected by the microphone according to the confirmed adjustment type.

20.The method according to claim 19, wherein the master chip controls the earphone to enter a listening mode when the touchpad detects that the user's palm covers the touchpad.

21.The earphone is characterized in that: at least one external microphone for collecting ambient sound is disposed outside the earphone; the earphone casing is provided with a touch panel having a multi-touch function, and the touch panel is connected with the main control chip of the earphone The noise reduction module and/or the monitoring module of the earphone are connected to the main control chip of the earphone, or integrated in the main control chip;

The touch panel detects a movement trajectory of a user's finger, and the main control chip is configured to confirm an adjustment type and an adjustment amount of the current ambient sound according to the movement trajectory, where the adjustment type of the ambient sound includes corresponding to noise reduction. The first type of control, and the second type corresponding to the monitoring control; and, according to the confirmed adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone with the confirmed adjustment amount.

22.The earphone according to claim 21, wherein when the touch panel detects that the user's palm covers the touch panel, the main control chip controls the earphone to enter a listening mode.

23.A method for adjusting ambient sound of a headset, characterized in that at least one external microphone for collecting ambient sound is disposed outside the earphone, the earphone comprising a noise reduction module and a monitoring module;

Receiving, by the earphone application client of the terminal, an adjustment type and an adjustment amount of the ambient sound; the adjustment type of the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the monitoring control;

According to the adjustment type, the control noise reduction module or the monitoring module adjusts the ambient sound collected by the microphone by using the adjustment amount.

24.The method according to claim 23, wherein when the headset receives the command to enter the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the non-listening mode, the current playing volume of the earphone is recorded. And adjusting the amount of the ambient sound, and controlling the monitoring module to adjust the ambient sound collected by the microphone by using a preset adjustment amount; when the earphone receives the command of the headset application client sent by the terminal to leave the listening mode The earphone is restored to the original working state according to the recorded playing volume and the amount of adjustment to the ambient sound.

25.The method according to claim 24, wherein when the headset receives the command to enter the listening mode sent by the earphone application client of the terminal, if the earphone is currently in the noise reduction playing audio mode or simply playing the audio mode, The playback volume of the earphone is controlled to be adjusted to a preset playback volume.

26.A terminal, characterized in that a headset application client is installed; the headset client provides a first virtual regulator on the interface for adjusting the type and adjustment amount of the ambient sound by the touch setting, and the touchable setting a second virtual adjuster that plays a volume; the type of adjustment to the ambient sound includes a first type corresponding to the noise reduction control, and a second type corresponding to the sounding control.

27.The terminal according to claim 26, wherein the first virtual adjuster and/or the second virtual adjuster are any of the following: a virtual knob, a virtual push button, and two virtual buttons.

28.The terminal according to claim 26, wherein the first virtual regulator or the second virtual adjustment is two virtual keys; and the earphone detects that two virtual keys are simultaneously pressed, and sends the entry to the earphone. The command of the listening mode; when the earphone detects that two virtual keys are simultaneously released, the earphone sends a command to leave the listening mode.

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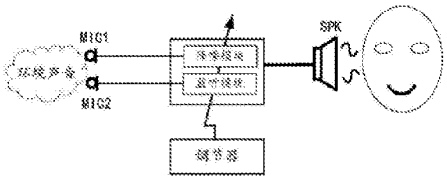
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(54)发明名称

耳机环境声音的调节方法和耳机

(57)摘要

本发明公开了一种耳机环境声音的调节方法和耳机。该方法包括:在耳机外侧设置至少一个采集环境声音的外置麦克风,在耳机外壳上或线控器的壳体上设置机械式可移动的调节器。调节器具有电气端口,通过电气端口将调节器连接至耳机的降噪模块和/或监听模块。该方法还包括:实时监测调节器的移动状态,当调节器发生移动时,获知调节器的移动量和移动类型;根据移动类型确认当前的调节类型,并根据移动量确认当前对环境声音的调节量;按照确认的调节类型,控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。本发明的技术方案实现了耳机对环境声音的灵活调节,使耳机功能更人性化,增强了耳机性能,提高了用户体验。



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1.一种耳机环境声音的调节方法，其特征在于，在耳机外侧设置至少一个采集环境声音的外置麦克风，在耳机外壳上设置机械式可移动的调节器，或者，在耳机的线控器的壳体上设置机械式可移动的调节器；

所述调节器具有电气端口，通过电气端口将调节器连接至耳机的降噪模块和/或监听模块，所述方法包括：

实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；

根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量；

按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节；

其中，所述根据移动量确认当前对环境声音的调节量包括：

根据移动量查询移动量和调节量的对应关系，得到当前对环境声音的调节量。

2.根据权利要求1所述的方法，其特征在于，所述方法还包括：根据调节器的移动行为，将调节器的移动状态划分为包括对应于降噪控制的第一移动类型，以及对应于监听控制的第二移动类型；

则所述根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量包括：

当监测到调节器发生的移动属于第一移动类型时，确认当前的调节类型为降噪控制，并根据移动量确认当前对环境声音的降噪量；和/或，

当监测到调节器发生的移动属于第二移动类型时，确认当前的调节类型为监听控制，并根据移动量确认当前对环境声音的放大量。

3.根据权利要求2所述的方法，其特征在于，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：

通过旋转部件实现调节器时，在调节器的旋转轨迹上设置旋转分界点，监测到旋转部件向旋转分界点的预定一侧旋转时，获知旋转部件的移动类型为第一移动类型，监测到旋转部件向旋转分界点的另一侧旋转时，获知旋转部件的移动类型为第二移动类型，并由旋转部件的旋转量得到所述移动量。

4.根据权利要求2所述的方法，其特征在于，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：

通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，监测到推钮向行程分界点的预定一侧滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离得到所述移动量。

5.根据权利要求2所述的方法，其特征在于，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：

通过两个按键实现调节器时，监测到第一按键被按动时，获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到所述移动量。

6.根据权利要求2所述的方法，其特征在于，

采用MCU实现所述降噪模块和/或监听模块时，采用确认的调节量修改MCU控制寄存器

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的写入值，实现降噪模块或监听模块对麦克风采集到的环境声音的调节。

7.根据权利要求2所述的方法，其特征在于，

所述调节器设置在头戴式耳机的听筒部位的壳体上。

8.根据权利要求1-7任一项所述的方法，其特征在于，在耳机外壳上设置监听模式触发模块，或者，在耳机的线控器的壳体上设置监听触发模块；所述方法还包括：

实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；

当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

9.根据权利要求8所述的方法，其特征在于，所述方法还包括：当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

10.根据权利要求8所述的方法，其特征在于，所述监听模式触发模块包括下列任一：可旋转面板式开关、具有单点或多点触控功能的触控板。

11.根据权利要求8所述的方法，其特征在于，所述监听模式触发模块包括下列任一：电容式触摸传感器、电阻式触摸传感器、红外线距离传感器、压力传感器、机械式开关。

12.一种耳机，其特征在于，所述耳机外侧设置有至少一个采集环境声音的外置麦克风，耳机外壳上设置有机械式可移动的调节器，或者，耳机的线控器的壳体上设置有机械式可移动的调节器；

所述调节器具有电气端口，所述调节器通过电气端口连接至耳机的降噪模块和/或监听模块；所述降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；

主控芯片监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；

主控芯片根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量；

主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节；

其中，所述根据移动量确认当前对环境声音的调节量包括：

根据移动量查询移动量和调节量的对应关系，得到当前对环境声音的调节量。

13.根据权利要求12所述的耳机，其特征在于，

通过旋转部件实现调节器时，在调节器的旋转轨迹上设置旋转分界点，主控芯片监测到旋转部件向旋转分界点的预定一侧旋转时，获知旋转部件的移动类型为第一移动类型，监测到旋转部件向旋转分界点的另一侧旋转时，获知旋转部件的移动类型为第二移动类型，并由旋转部件的旋转量得到所述移动量；

通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，主控芯片监测到推钮向行程分界点的预定一侧滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离

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得到所述移动量；

通过两个按键实现调节器时，监测到第一按键被按动时，主控芯片获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到所述移动量。

14.根据权利要求12所述的耳机，其特征在于，

所述调节器设置在头戴式耳机的听筒部位的壳体上。

15.根据权利要求12-14任一项所述的耳机，其特征在于，所述耳机外壳上还设置有监听模式触发模块，或者，耳机的线控器的壳体上设置有监听触发模块；

所述主控芯片，还用于实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；以及，当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

16.根据权利要求15所述的耳机，其特征在于，所述主控芯片，还用于当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

17.根据权利要求15所述的耳机，其特征在于，所述监听模式触发模块包括下列任一：可旋转面板式开关、具有单点或多点触控功能的触控板。

18.根据权利要求15所述的耳机，其特征在于，所述监听模式触发模块包括下列任一：电容式触摸传感器、电阻式触摸传感器、红外线距离传感器、压力传感器、机械式开关。

19.一种耳机环境声音的调节方法，其特征在于，在耳机外侧设置至少一个采集环境声音的外置麦克风；在耳机外壳上设置具有多点触控功能的触控板，所述触控板与耳机的主控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；

所述触控板检测用户手指的移动轨迹，所述主控芯片根据所述移动轨迹确认当前对环境声音的调节类型和调节量，所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；

所述主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

20.根据权利要求19所述的方法，其特征在于，所述触控板检测到用户手掌覆盖所述触控板时，所述主控芯片控制所述耳机进入监听模式。

21.一种耳机，其特征在于，耳机外侧设置有至少一个采集环境声音的外置麦克风；耳机外壳上设置有具有多点触控功能的触控板，所述触控板与耳机的主控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；

所述触控板检测用户手指的移动轨迹，所述主控芯片，用于根据所述移动轨迹确认当前对环境声音的调节类型和调节量，所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；以及，按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

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22.根据权利要求21所述的耳机，其特征在于，所述触控板检测到用户手掌覆盖所述触控板时，所述主控芯片控制所述耳机进入监听模式。

23.一种耳机环境声音的调节方法，其特征在于，在耳机外侧设置至少一个采集环境声音的外置麦克风，所述耳机包括降噪模块和监听模块；

接收终端的耳机应用客户端发送的对环境声音的调节类型和调节量；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；

按照所述调节类型，控制降噪模块或监听模块采用所述调节量对麦克风采集到的环境声音进行调节。

24.根据权利要求23所述的方法，其特征在于，所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；所述耳机接收到终端的耳机应用客户端发送的离开监听模式的命令时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

25.根据权利要求24所述的方法，其特征在于，所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

26.一种终端，其特征在于，安装有耳机应用客户端；所述耳机客户端在其界面上提供可供触摸设置对环境声音的调节类型和调节量的第一虚拟调节器，以及可供触摸设置播放音量的第二虚拟调节器；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型。

27.根据权利要求26所述的终端，其特征在于，所述第一虚拟调节器和/或第二虚拟调节器为下列任一：虚拟旋钮、虚拟推钮、两个虚拟按键。

28.根据权利要求26所述的终端，其特征在于，所述第一虚拟调节器或第二虚拟调节为两个虚拟按键；所述耳机检测到两个虚拟按键同时被按住时，向耳机发送进入监听模式的命令；所述耳机检测到两个虚拟按键同时被放开时，向耳机发送离开监听模式的命令。

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耳机环境声音的调节方法和耳机

技术领域

[0001] 本发明涉及耳机技术领域，特别涉及一种耳机环境声音的调节方法和耳机。

背景技术

[0002] 社会信息化程度的提高使得人们能随时随地的进行通信和交流，各种各样的通信设备(手机、蓝牙耳机、立体声耳机等)和通信技术的广泛应用极大地方便了人们的生活，提高了工作效率。然而社会发展带来的一个比较严重的问题是噪声问题，在噪声环境中进行通信，严重影响到通信语音的清晰度和可懂度，当噪声高到一定程度时，不但通信无法进行，而且会伤害到人的听力和身心健康。

[0003] 针对噪声问题，现有方案中提供了一些具有降噪能力的耳机产品。当开启降噪耳机的降噪功能时，耳机会对外界声音进行降噪，然后输出给耳机扬声器让耳机佩戴者听到降噪后的声音。

[0004] 然而，因为外界环境噪声是变化的，不同时间不同地点环境噪声的能量级别和频谱分布是不一样的，现有的降噪耳机仅仅具有开启降噪功能和关闭降噪功能两种方式，这种直接实现降噪功能的有无，并不能给耳机佩戴者带来好的用户体验。

[0005] 例如，在普通办公室环境，如果直接打开耳机降噪功能提供25dB以上的降噪量，耳机佩戴者会有一种压抑的、犹如在专业声学消声室的感觉，而这种环境下，又存在一定的环境噪声，也不能直接把降噪功能关掉。

发明内容

[0006] 本发明提供的一种耳机环境声音的调节方法和耳机，以解决现有的降噪耳机仅能直接实现降噪功能的有无、降噪方式单一问题。

[0007] 为达到上述目的，本发明的技术方案是这样实现的：

[0008] 一方面，本发明实施例提供了一种耳机环境声音的调节方法。该方法包括：在耳机外侧设置至少一个采集环境声音的外置麦克风，在耳机外壳上设置机械式可移动的调节器，或者，在耳机的线控器的壳体上设置机械式可移动的调节器；所述调节器具有电气端口，通过电气端口将调节器连接至耳机的降噪模块和/或监听模块，所述方法包括：实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量；按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0009] 优选地，所述方法还包括：根据调节器的移动行为，将调节器的移动状态划分为包括对应于降噪控制的第一移动类型，以及对应于监听控制的第二移动类型；则所述根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量包括：当监测到调节器发生的移动属于第一移动类型时，确认当前的调节类型为降噪控制，并根据移动量确认当前对环境声音的降噪量；和/或，当监测到调节器发生的移动属于第二移动类型时，确认当前的调节类型为监听控制，并根据移动量确认当前对环境声音的放大量。

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[0010] 优选地，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：通过旋转部件实现调节器时，在调节器的旋转轨迹上设置旋转分界点，监测到旋转部件向旋转分界点的预定一侧旋转时，获知旋转部件的移动类型为第一移动类型，监测到旋转部件向旋转分界点的另一侧旋转时，获知旋转部件的移动类型为第二移动类型，并由旋转部件的旋转量得到所述移动量。

[0011] 优选地，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，监测到推钮向行程分界点的预定一侧滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离得到所述移动量。

[0012] 优选地，所述实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型包括：通过两个按键实现调节器时，监测到第一按键被按动时，获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到所述移动量。

[0013] 优选地，采用MCU实现所述降噪模块和/或监听模块时，采用确认的调节量修改MCU控制寄存器的写入值，实现降噪模块或监听模块对麦克风采集到的环境声音的调节。

[0014] 优选地，所述根据移动量确认当前对环境声音的调节量包括：根据移动量查询移动量和调节量的对应关系，得到当前对环境声音的调节量。

[0015] 优选地，所述调节器设置在头戴式耳机的听筒部位的壳体上。

[0016] 优选地，在耳机外壳上设置监听模式触发模块，或者，在耳机的线控器的壳体上设置监听触发模块；所述方法还包括：实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0017] 优选地，所述方法还包括：当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

[0018] 优选地，所述监听模式触发模块包括下列任一：可旋转面板式开关、具有单点或多点触控功能的触控板。

[0019] 优选地，所述监听模式触发模块包括下列任一：电容式触摸传感器、电阻式触摸传感器、红外线距离传感器、压力传感器、机械式开关。

[0020] 又一方面，本发明实施例提供了一种耳机，所述耳机外侧设置有至少一个采集环境声音的外置麦克风，耳机外壳上设置有机械式可移动的调节器，或者，耳机的线控器的壳体上设置有机械式可移动的调节器；所述调节器具有电气端口，所述调节器通过电气端口连接至耳机的降噪模块和/或监听模块，所述降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；主控芯片监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；主控芯片根据移动类型确认当前的调节类型，并根据移动量

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确认当前对环境声音的调节量；主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0021] 优选地，通过旋转部件实现调节器时，在调节器的旋转轨迹上设置旋转分界点，主控芯片监测到旋转部件向旋转分界点的预定一侧旋转时，获知旋转部件的移动类型为第一移动类型，监测到旋转部件向旋转分界点的另一侧旋转时，获知旋转部件的移动类型为第二移动类型，并由旋转部件的旋转量得到所述移动量；通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，主控芯片监测到推钮向行程分界点的预定一侧滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离得到所述移动量；通过两个按键实现调节器时，监测到第一按键被按动时，主控芯片获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到所述移动量。

[0022] 优选地，所述调节器设置在头戴式耳机的听筒部位的壳体上。

[0023] 优选地，所述耳机外壳上还设置有监听模式触发模块，或者，耳机的线控器的壳体上设置有监听触发模块；所述主控芯片，还用于实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；以及，当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0024] 优选地，所述主控芯片，还用于当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

[0025] 优选地，所述监听模式触发模块包括下列任一：可旋转面板式开关、具有单点或多点触控功能的触控板。

[0026] 优选地，所述监听模式触发模块包括下列任一：电容式触摸传感器、电阻式触摸传感器、红外线距离传感器、压力传感器、机械式开关。

[0027] 又一方面，本发明实施例提供了一种耳机环境声音的调节方法，该方法包括：在耳机外侧设置至少一个采集环境声音的外置麦克风；在耳机外壳上设置具有多点触控功能的触控板，所述触控板与耳机的主控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；所述触控板检测用户手指的移动轨迹，所述主控芯片根据所述移动轨迹确认当前对环境声音的调节类型和调节量，所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；所述主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0028] 优选地，所述触控板检测到用户手掌覆盖所述触控板时，所述主控芯片控制所述耳机进入监听模式。

[0029] 又一方面，本发明实施例提供了一种耳机，耳机外侧设置有至少一个采集环境声音的外置麦克风；耳机外壳上设置有具有多点触控功能的触控板，所述触控板与耳机的主

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控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；所述触控板检测用户手指的移动轨迹，所述主控芯片，用于根据所述移动轨迹确认当前对环境声音的调节类型和调节量，所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；以及，按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0030] 优选地，所述触控板检测到用户手掌覆盖所述触控板时，所述主控芯片控制所述耳机进入监听模式。

[0031] 本发明实施例的技术方案，通过调节器的移动会得到多个不同渐变的移动量，不同的移动量对应于不同的调节量，从而能够根据需要任意调节降噪量的大小或监听到的环境声音的音量大小，解决了现有方案只能单一选择降噪功能开启或者关闭带来的问题，实现了耳机对环境声音的灵活调节，使耳机功能更人性化，增强了耳机性能，提高了用户体验。

[0032] 又一方面，本发明实施例提供了一种耳机环境声音的调节方法，该方法包括：在耳机外侧设置至少一个采集环境声音的外置麦克风，所述耳机包括降噪模块和监听模块；接收终端的耳机应用客户端发送的对环境声音的调节类型和调节量；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；按照所述调节类型，控制降噪模块或监听模块采用所述调节量对麦克风采集到的环境声音进行调节。

[0033] 优选地，所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；所述耳机接收到终端的耳机应用客户端发送的离开监听模式的命令时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0034] 优选地，所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

[0035] 又一方面，本发明实施例提供了一种终端，安装有耳机应用客户端；所述耳机客户端在其界面上提供可供触摸设置对环境声音的调节类型和调节量的第一虚拟调节器，以及可供触摸设置播放音量的第二虚拟调节器；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型。

[0036] 优选地，所述第一虚拟调节器和/或第二虚拟调节器为下列任一：虚拟旋钮、虚拟推钮、两个虚拟按键。

[0037] 优选地，所述第一虚拟调节器或第二虚拟调节为两个虚拟按键；所述耳机检测到两个虚拟按键同时被按住时，向耳机发送进入监听模式的命令；所述耳机检测到两个虚拟按键同时被放开时，向耳机发送离开监听模式的命令。

[0038] 本发明实施例的技术方案，通过终端的耳机应用客户端调节降噪量的大小或监听到的环境声音的音量大小，解决了现有方案只能单一选择降噪功能开启或者关闭带来的问题，实现了耳机对环境声音的灵活调节，使耳机功能更人性化，增强了耳机性能，提高了用户体验。

[0039] 通过以下参照附图对本发明的示例性实施例的详细描述，本发明的其它特征及其

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优点将会变得清楚。

附图说明

[0040] 被结合在说明书中并构成说明书的一部分的附图示出了本发明的实施例，并且连同其说明一起用于解释本发明的原理。

[0041] 图1为本发明一个实施例提供的应用本方案的一个场景示意图；

[0042] 图2为本发明一个实施例提供的耳机环境声音的调节方法示意图；

[0043] 图3为本发明又一个实施例提供的调节器的一种实现方式示意图；

[0044] 图4为本发明又一个实施例提供的调节器的另一种实现方式示意图；

[0045] 图5为本发明又一个实施例提供的调节器的又一种实现方式示意图；

[0046] 图6为本发明又一个实施例提供的调节器的又一种实现方式示意图；

[0047] 图7为本发明又一个实施例提供的调节器的又一种实现方式示意图；

[0048] 图8为本发明又一个实施例提供的采用前馈加反馈实现的混合主动降噪耳机系统框图；

[0049] 图9示出了本方案的耳机在不同降噪级别不同降噪量的实验结果图；

[0050] 图10示出了本方案的耳机实现的不同监听控制对应EQ曲线的实验结果图；

[0051] 图11为本发明一个实施例提供的触发监听模式的一种方法示意图；

[0052] 图12为本发明一个实施例提供的监听模式触发模块的一种实现方式示意图；

[0053] 图13为本发明又一个实施例提供的监听模式触发模块的另一种实现方式示意图。

具体实施方式

[0054] 现在将参照附图来详细描述本发明的各种示例性实施例。应注意到：除非另外具体说明，否则在这些实施例中阐述的部件和步骤的相对布置、数字表达式和数值不限制本发明的范围。

[0055] 以下对至少一个示例性实施例的描述实际上仅仅是说明性的，决不作为对本发明及其应用或使用的任何限制。

[0056] 对于相关领域普通技术人员已知的技术、方法和设备可能不作详细讨论，但在适当情况下，所述技术、方法和设备应当被视为说明书的一部分。

[0057] 在这里示出和讨论的所有例子中，任何具体值应被解释为仅仅是示例性的，而不是作为限制。因此，示例性实施例的其它例子可以具有不同的值。

[0058] 应注意到：相似的标号和字母在下面的附图中表示类似项，因此，一旦某一项在一个附图中被定义，则在随后的附图中不需要对其进行进一步讨论。

[0059] 本发明一个实施例提供了一种耳机环境声音的调节方法。参见图1，示出了应用本方案的一个场景，该方法包括：在耳机外侧设置至少一个采集环境声音的外置麦克风，如连接至降噪模块的MIC1和连接至监听模块的MIC2。在一种变形的方式中，可利用同一个外置麦克风分别与降噪模块和监听模块连接，实现降噪和监听。在耳机外壳上设置机械式可移动的调节器，调节器具有电气端口，通过电气端口将调节器连接至耳机的降噪模块和/或监听模块。降噪模块和监听模块可以集成在耳机的主控芯片上，由一个芯片实现，或者在耳机内分别设置分立的器件实现降噪模块和监听模块。

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[0060] 参见图2，本实施例提供的耳机环境声音的调节方法还包括：

[0061] S201：主控芯片实时监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型。

[0062] S202：主控芯片根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量。

[0063] S203：主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0064] 在噪声背景下进行通信时，耳机会采用降噪模块。降噪模块采用前馈式(Feed-Forward，FF)降噪、反馈式(FeedBack，FB)降噪或者是前馈加反馈的混合降噪。目前的降噪主要考虑如下两个方面：一方面是在通信设备送话端采用声学信号处理技术提高传声器拾取的语音信号的信噪比，使得远端用户能够听清近端用户的讲话。另一方面需要在通信设备受话端提高受话端语音的信噪比，使得近端能够听清远端用户送过来的语音信号。

[0065] 另外一些场景中，为了在有需要的时候不拿下耳机也可以方便地和外界进行通话或者是对外界环境噪声进行监听，耳机会采用监听模块，对外界环境中的声音进行放大，保证佩戴者能够听到外界声音。现有具有监听功能的耳机，也仅仅具有开启监听功能和关闭监听功能两种方式，这种直接实现监听功能的有无，不能给耳机佩戴者带来好的用户体验。

[0066] 需要说明的是，降噪和监听的功能其实是相反的两个作用效果，所以本发明中涉及的降噪模块和监听模块可以为同一物理实体，例如，降噪模块和监听模块可以为一个ANC(active noise concellation)有源降噪芯片，或者一个ANC系统电路。

[0067] 上述步骤S201至S203也可以由降噪模块或监听模块实现，可以由降噪模块或监听模块对调节器的移动进行监测。

[0068] 本实施例可以应用在耳机中仅存在降噪模块和监听模块中的一种的场景，这时调节器移动时，仅对应一种移动类型，例如，当仅存在降噪模块时，若监测到调节器移动，则调节器的移动类型仅对应于降噪的调节类型。

[0069] 本发明实施例的技术方案，通过调节器的移动会得到多个不同渐变的移动量，不同的移动量对应于不同的调节量，从而能够根据需要任意调节降噪量的大小或监听到的环境声音的音量大小，解决了现有方案只能单一选择降噪功能开启或者关闭带来的问题，实现了耳机对环境声音的灵活调节，使耳机功能更人性化，增强了耳机性能，提高了用户体验。

[0070] 在图2所示实施例的基础上，进一步的，本发明实施例还包括：根据调节器的移动行为，将调节器的移动状态划分为包括对应于降噪控制的第一移动类型，以及对应于监听控制的第二移动类型。其中，利用降噪模块执行降噪控制，利用监听模块执行监听控制。

[0071] 根据调节器的具体实现方式不同，调节器的移动行为也不同，如调节器为旋钮时，调节器的移动行为可以包括向左旋转、向右旋转。

[0072] 本实施例通过将调节器的移动状态划分为两种移动类型，以利用同一调节器实现调节降噪和调节监听两种功能。这主要考虑到降噪和监听的功能其实是相反的两个作用效果，采用同一个结构件的方式来实现两种功能的调节，能够在保证耳机功能的同时，简化耳机的结构。本实施例主要以设置一个调节器为例的场景进行说明，但并不局限于此，本实施例也可以设置两个调节器，一个调节器用于实现降噪控制，一个调节器用于实现监听控制。

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[0073] 则上述方法还包括：在一次监测过程中，主控芯片可以执行下述的一种操作：

[0074] 当主控芯片监测到调节器发生的移动属于第一移动类型时，确认当前的调节类型为降噪控制，并根据移动量确认当前对环境声音的降噪量；

[0075] 当主控芯片监测到调节器发生的移动属于第二移动类型时，确认当前的调节类型为监听控制，并根据移动量确认当前对环境声音的放大量

[0076] 当耳机上设置有两个调节器时，主控芯片可以针对两个调节器的移动，同时执行上述两种操作，或者仅执行上述一种操作。

[0077] 主控芯片可以预先维护一个移动量和调节量的对应关系表，根据移动量查询移动量和调节量的对应关系，即可快速得到当前对环境声音的调节量。这种方式，在进行监测时，不需要实时对降噪量或放大量进行计算，能够快速得到当前所使用的降噪量或放大量，提高耳机对环境声调节的速度。

[0078] 本发明又一实施例主要对调节器的工作方式进行说明，耳机环境声音调节方案的其他内容可以参见本发明的其他实施例。

[0079] 通过旋钮实现调节器时，在调节器的旋转轨迹上设置旋转分界点，如将旋转轨迹的中间点作为旋转分界点，主控芯片(或降噪模块)监测到旋钮向旋转分界点的预定一侧(如左侧)旋转时，获知旋钮的移动类型为第一移动类型，主控芯片(或监听模块)监测到旋钮向旋转分界点的另一侧(如右侧)旋转时，获知旋钮的移动类型为第二移动类型，并由旋钮的旋转量得到所述移动量。参见图3，示出了调节器的一种实现方式。调节器为旋钮，与头戴式耳机的听筒部分同轴装配，为每一个听筒设置一个调节器，调节器位于听筒部分壳体的外端，一个调节器负责同侧的听筒的环境声音的调节。设置作为调节器的旋钮具有一定的大小和厚度，便于人手握持，例如调节器可以为具有一定厚度的圆盘部件。

[0080] 或者，参见图4，示出了调节器的另一种实现方式。调节器为旋钮，该旋钮突出听筒壳体外侧一定高度，以便于人手握持，旋钮为柱体部件。

[0081] 或者，参见图5，示出了调节器的另一种实现方式，调节器为可旋转面板结构，设置在耳机听筒的外壳上。

[0082] 通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，监测到推钮向行程分界点的预定一侧(如上方)滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧(如下方)滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离得到移动量。参见图6，示出了调节器为推钮的示意图，推钮安装在滑动槽中，推钮沿着滑动槽上下滑动。

[0083] 通过两个按键实现调节器时，监测到第一按键被按动时，获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到所述移动量。参见图7，示出了调节器为按键的示意图，包括可以上下按动的第一按键和第二按键。

[0084] 一个场景下，如果调节器是旋钮的方式，则可以设置旋钮处在中间位置时对外界环境声音不放大也不降低，往左旋转则实现降噪，往左旋转的越多降噪量越大，旋转到最左边的位置降噪量最大；设置旋钮往右旋转为监听到的环境声放大，越往右旋转监听声越大，旋转到最右边的位置则监听声放大达到最大，同样推钮和按键可以实现同样的功能。

[0085] 另外，无论对于旋转部件(例如旋钮、旋转盘、可旋转面板结构等)、推扭还是按键

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的方式，除了上述的采用机械结构键的方式实现外，也可以采用近距离感应的触摸键的方式来实现旋转调节，上下滑动调节和上下按键调节的功能。

[0086] 上述主要以头戴式耳机为例对调节器进行了说明，对于头戴式耳机或入耳式耳机，调节器还可以设置在线控器上，具体设置方式可以参见头戴式耳机的调节器。

[0087] 本发明又一实施例主要对降噪控制的调节场景进行说明，耳机环境声音调节方案的其他内容可以参见本发明的其他实施例。

[0088] 如图8所示，示出了降噪模块采用前馈加反馈实现时的混合主动降噪耳机系统框图。其系统最大降噪量和降噪频带为反馈(FB主动降噪控制ANC)和前馈(FF ANC)两方面降噪量的叠加，基本降噪性能参见图8，降噪模块在低频1000Hz都有主动降噪量，最大主动降噪量可达到25db左右，配合耳罩或者耳塞的被动隔声，降噪耳机在整个频带内都能提供差不多20db的降噪效果。

[0089] 降噪量/放大量可以通过调整电路或者是软件修改寄存器的形式实现不同的降噪级别/监听级别，用户可以根据需要手动进行降噪效果/监听效果的调整。

[0090] 例如，采用MCU(Micro Control Unit，微控制单元)实现降噪模块和/或监听模块时，如主控芯片为MCU时，利用确认的调节量修改MCU控制MCU寄存器的写入值，来改变降噪量或放大量。降噪模块或监听模块会将置于外界环境噪声中的麦克风(如FF/FB降噪MIC1或监听MIC2)拾取到的信号经过一定的EQ(均衡器)放大和补偿，然后输出给耳机扬声器让耳机佩戴者听到。通过EQ处理可以实现弱化低频提升中高频的作用，使耳机输出的声音更符合人耳听觉效果。

[0091] 下面结合实验结果图，更加清楚的说明本方案的有益效果。

[0092] 参见图9，示出了应用本方案的耳机实现的不同降噪量的实验结果图。

[0093] 图9示出了一种本方案的耳机在不同降噪级别不同降噪量的示意图，这为本方案的一种实现方式，其中在不同级别需要的降噪量和降噪频带可以根据需要进行不同的调整，来达到更好的用户体验。其中，降噪级别的总级别量也可以根据实际需要进行设计。如图9所示的方式下，本方案的耳机能够根据不同的移动量实现十种不同的降噪级别。

[0094] 参见图10，示出了应用本方案的耳机实现的不同监听控制对应EQ(均衡器)曲线的实验结果图。这为本方案的一种实现方式，如图10所示的方式下，本方案的耳机能够根据不同的移动量实现十种不同的监听级别。其中，监听级别的总级别量、每个级别量对应的放大倍数、EQ曲线都可以根据不同耳机或不同需要进行设计以达到更好的用户体验。

[0095] 本发明实施例的技术方案，通过调节器的移动会得到多个不同渐变(各次渐变可等量或不等量)的移动量，不同的移动量对应于不同的调节量，从而能够根据需要任意调节降噪量的大小或监听到的环境声音的音量大小，解决了现有方案只能单一选择降噪功能开启或者关闭带来的问题，实现了耳机对环境声音的灵活调节，使耳机功能更人性化，增强了耳机性能，提高了用户体验。

[0096] 本发明又一个实施例提供了一种耳机。该耳机外侧设置有至少一个采集环境声音的外置麦克风，耳机外壳上设置有机械式可移动的调节器，调节器具有电气端口，调节器通过电气端口连接至耳机的降噪模块和/或监听模块。

[0097] 降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；主控芯片监测调节器的移动状态，当调节器发生移动时，获知调节器的移动量和移动类型；主控

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芯片根据移动类型确认当前的调节类型，并根据移动量确认当前对环境声音的调节量；主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0098] 通过旋钮实现调节器时，在调节器的旋转轨迹上设置旋转分界点，主控芯片监测到旋钮向旋转分界点的预定一侧旋转时，获知旋钮的移动类型为第一移动类型，监测到旋钮向旋转分界点的另一侧旋转时，获知旋钮的移动类型为第二移动类型，并由旋钮的旋转量得到移动量；

[0099] 通过推钮实现调节器时，在调节器的推动行程上设置行程分界点，主控芯片监测到推钮向行程分界点的预定一侧滑动时，获知推钮的移动类型为第一移动类型，监测到推钮向行程分界点的另一侧滑动时，获知推钮的移动类型为第二移动类型，并由推钮的滑动距离得到移动量；

[0100] 通过两个按键实现调节器时，监测到第一按键被按动时，主控芯片获知按键的移动类型为第一移动类型，监测到第二按键被按动时，获知按键的移动类型为第二移动类型，并由按键的按动次数得到移动量。

[0101] 在降噪模块或监听模块与主动芯片分立时，上述耳机中主控芯片执行的操作，也可以由降噪模块或监听模块实现。

[0102] 进一步的，调节器设置在头戴式耳机的听筒部位的壳体上，旋钮实现的调节器可以和听筒部位同轴装配或者异轴装配。

[0103] 在很多应用场合下，用户想要快速的从降噪和听音模式下直接切换到监听环境声的模式，在一段时间后又需要很快回到之前的降噪模式和听音状态，例如佩戴者正在开着降噪功能听音乐，旁边有人需要和佩戴者短时沟通，这时如果通过旋转调节的方式从降噪模式调整到监听模式，可能会稍微耗时一些，沟通完之后再调整到之前的降噪等级也需要消耗一定的时间，所以本发明还提供了一种能够实现这种快速短时切换到监听模式的需求的技术方案：

[0104] 参考图11为本发明一个实施例提供的触发监听模式的一种实现方式，在耳机外壳上设置监听模式触发模块，还包括以下步骤：

[0105] S301：实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；

[0106] S302：当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0107] 在步骤S302中，当监听模式的触发条件由不满足变为满足时，如果耳机当前正处于监听模式，则可以不做处理，继续保持当前的监听状态。

[0108] 对于同时具有降噪模块和监听模块的耳机来说，可以具有以下五种工作模式：单纯的播放音频模式、单纯的降噪模式、降噪播放音频模式、单纯的监听模式、以及监听播放音频模式。单纯的降噪模式，适合于用户在较为嘈杂的环境中保持安静使用，在这种模式下，耳机不会播放音乐等内容，只是对环境声音进行降噪，即播放音量为零。单纯的播放音

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频模式，是指耳机只播放音乐等内容，对外界环境声音不做任何调节(对应于前文提到的旋钮处在中间位置)的模式，即对外界环境声音的调节量为零。

[0109] 所述方法还包括：当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还需要控制所述耳机的播放音量调整到预先设定的播放音量。这种情况较为适用于，用户原本正在降噪播放音乐或者单纯的播放音乐，在触发进入监控模式后仍然想继续播放音乐，则除了控制监听模块按照预先设定的放大量对环境声音进行调节，还按照预先设定的播放音量继续播放音乐，预先设定的播放音量通常是一个较低的音量，以便于用户能够清楚听到环境声音。

[0110] 其中，所述监听模式触发模块包括下列任一：电容式触摸传感开关、电阻式触摸传感开关、近距离传感开关、压力传感开关、机械式开关。近距离传感开关例如为红外近距离传感开关。机械式开关例如为按钮式机械开关、旋钮式机械开关、推钮式机械开关。监听模式触发模块还可以为可旋转面板式开关或单金属片式触摸开关。其中，单金属片式触摸开关、电容式触摸传感开关、电阻式触摸传感开关、以及近距离传感开关都能够实现手掌覆盖耳机外壳上的监听模式触发模块就可以从非监听模式进入到监听模式、手掌离开耳机外壳后就可以自动快速返回原工作状态的功能。

[0111] 与之对应的，本发明还提出了一种具有监听模式触发模块的耳机，所述耳机外壳上还设置有监听模式触发模块；所述主控芯片，还用于实时监测所述监听模式触发模块输出的数据，根据所述监听模式触发模块输出的数据判断监听模式的触发条件是否满足；以及，当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0112] 所述主控芯片，还用于当监听模式的触发条件由不满足变为满足时，如果耳机当前正处于监听模式，则继续保持当前的监听状态。

[0113] 其中，所述主控芯片，还用于当监听模式的触发条件由不满足变为满足时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

[0114] 其中，所述监听模式触发模块包括下列任一：电容式触摸传感器、电阻式触摸传感器、红外线距离传感器、压力传感器、机械式开关。所述监听模式触发模块可以例如为单金属片式触摸检测电容。

[0115] 其中，所述监听模式触发模块为具有多点触控功能的触控板，当用户手掌覆盖触控板的大部分面积或者整个触控板时，即会触发监听模式。

[0116] 本发明提供了设置在耳机外壳上的监听模式触发模块，利用监听模式触发模块可以直接从降噪听音模式直接切换到监听模式，当触发条件消失时，还可以自动快速返回之前的降噪级别和播放听音状态，本发明既能满足用户对外界环境声进行渐进控制和调节的需求，又提供了快速进入监听和返回降噪播放状态的快捷功能，提供了更人性化的用户体验和使用感受，以及更完善的耳机控制方式。

[0117] 参考图12所示为本发明一个实施例提供的监听模式触发模块的一种实现方式：监听模式触发模块为单金属片式触摸检测电容，设置于头戴式耳机的一个听筒部位的壳体

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上，调节器设置在头戴式耳机的另一个听筒部位的壳体上。

[0118] 用户手掌接触或离开耳机外壳的动作会导致单金属片式触摸检测电容的容值发生变化，实时监测单金属片式触摸检测电容输出的容值，判断单金属片式触摸检测电容输出的容值是否大于等于阈值，如果大于等于阈值就判断监听模式的触发条件为满足，否则判断监听模式的触发条件为不满足。

[0119] 当监听模式的触发条件由不满足变为满足时，如果耳机当前处于非监听模式，则记录耳机当前的播放音量和对环境声音的调节量，并且控制监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；当监听模式的触发条件由满足变为不满足时，根据记录的播放音量和对环境声音的调节量控制耳机恢复到原来的工作状态。

[0120] 本发明提供了手掌覆盖耳机外壳上的监听模式触发模块时，就可以直接从降噪听音模式直接切换到监听模式，从而快速地进入监听状态，当手掌离开耳机外壳后就可以自动快速返回之前的降噪级别和播放听音状态，本发明既能满足用户对外界环境声进行渐进控制和调节的需求，又提供了快速进入监听和返回降噪播放状态的快捷功能，提供了更人性化的用户体验和使用感受，以及更完善的耳机控制方式。

[0121] 参考图13所示为本发明又一个实施例提供的监听模式触发模块的另一种实现方式：监听触发模块为一个机械旋钮式开关，设置于耳机的听筒部分的外壳上，并且突出听筒壳体外侧一定高度，以便于人手握持。

[0122] 图12和图13主要以头戴式耳机为例对监听触发模块进行了说明，对于入耳式耳机或头戴式耳机，监听触发模块还可以设置在线控器的壳体上。

[0123] 本发明还提出了一种耳机环境声音的调节方法，在耳机外侧设置至少一个采集环境声音的外置麦克风；在耳机外壳上设置具有多点触控功能的触控板，触控板与耳机的主控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；

[0124] 触控板检测用户手指的移动轨迹，主控芯片根据移动轨迹确认当前对环境声音的调节类型和调节量，对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；

[0125] 主控芯片按照确认的调节类型，控制降噪模块或监听模块采用确认的调节量对麦克风采集到的环境声音进行调节。

[0126] 例如，当检测到用户手指顺时针移动时，主控芯片确认应当为降噪控制的第一类型，根据移动轨迹的长度或角度确认降噪量的大小。反之，例如，当检测到用户手指逆时针移动时，主控芯片确认应当为监听控制的第二类型，根据移动轨迹的长度或角度确认对麦克风采集到的环境声音的放大量。

[0127] 优选地，触控板检测到用户手掌覆盖触控板时，主控芯片控制耳机进入监听模式。

[0128] 本发明还提出了一种耳机，耳机外侧设置有至少一个采集环境声音的外置麦克风；耳机外壳上设置有具有多点触控功能的触控板，触控板与耳机的主控芯片连接；耳机的降噪模块和/或监听模块连接至耳机的主控芯片，或者，集成在主控芯片中；

[0129] 触控板检测用户手指的移动轨迹，主控芯片，用于根据移动轨迹确认当前对环境声音的调节类型和调节量，对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；以及，按照确认的调节类型，控制降噪模块或监听模块采用确

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认的调节量对麦克风采集到的环境声音进行调节。

[0130] 优选地，触控板检测到用户手掌覆盖触控板时，主控芯片控制耳机进入监听模式，反之，触控板检测到用户手掌覆盖触控板后离开时，主控芯片控制耳机离开监听模式，回到之前的工作状态。

[0131] 采用这种方式，就可以利用一个具有多点触控功能的触控板实现逐渐调节耳机降噪量的大小或监听到的环境声音的音量大小，以及快速进入监听模式和返回降噪播放状态的快捷功能，提供了更人性化的用户体验和使用感受，以及更完善的耳机控制方式。

[0132] 上述提到的耳机环境声音的调节方法、调节器、触发监听模式的方法、以及监听触发模块都可以通过终端的耳机应用客户端实现：

[0133] 本发明提供了一种耳机环境声音的调节方法，在耳机外侧设置至少一个采集环境声音的外置麦克风，所述耳机包括降噪模块和/或监听模块；

[0134] 接收终端的耳机应用客户端发送的对环境声音的调节类型和调节量；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型；按照所述调节类型，控制降噪模块或监听模块采用所述调节量对麦克风采集到的环境声音进行调节；以及接收终端的耳机应用客户端发送的播放音量，根据所述播放音量对耳机的功放进行设置。

[0135] 进一步的，所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于非监听模式，则记录所述耳机当前的播放音量和对环境声音的调节量，并且控制所述监听模块采用预先设定的调节量对麦克风采集到的环境声音进行调节；所述耳机接收到终端的耳机应用客户端发送的离开监听模式的命令时，根据记录的播放音量和对环境声音的调节量控制所述耳机恢复到原来的工作状态。

[0136] 所述耳机接收到终端的耳机应用客户端发送的进入监听模式的命令时，如果耳机当前处于降噪播放音频模式或者单纯的播放音频模式，还控制所述耳机的播放音量调整到预先设定的播放音量。

[0137] 本发明还提供了一种终端，安装有耳机应用客户端；所述耳机客户端在其界面上提供可供触摸设置对环境声音的调节类型和调节量的第一虚拟调节器，以及可供触摸设置播放音量的第二虚拟调节器；所述对环境声音的调节类型包括对应于降噪控制的第一类型，以及对应于监听控制的第二类型。

[0138] 所述第一虚拟调节器和/或第二虚拟调节器为下列任一：虚拟旋钮、虚拟推钮、两个虚拟按键。

[0139] 所述第一虚拟调节器或第二虚拟调节为两个虚拟按键；所述耳机检测到两个虚拟按键同时被按住时，向耳机发送进入监听模式的命令；所述耳机检测到两个虚拟按键同时被放开时，向耳机发送离开监听模式的命令。

[0140] 需要说明的是，本发明中提到的机械式可移动的调节器和监听触发模块，可以和上述安装于终端的耳机应用客户端同时存在、协调工作，在本发明的基础上本领域技术人员可以对此做出设置，在此不再赘述。

[0141] 本发明耳机实施例中各器件的具体工作方式可以参见本发明的方法实施例，在此不再赘述。

[0142] 由上，本申请至少具有如下优点：

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[0143] 1、采用机械构件实现调节器，满足降噪和监听两个功能都有手动调节等级的用户需求，并可以采用同一个机械构件实现调节器，以简单的结构实现功能强大的耳机；

[0144] 2、调节器的机械结构的实现方式可以采用旋钮的方式、推钮的方式，两个独立的按键的方式或者是其他任意可以实现该功能的结构方式，实现方式灵活；

[0145] 3、采用旋钮、推钮或者上下按键等方式，使用户自行根据需要调节降噪量的大小或监听环境声音量大小，避免了现有单一只能选择降噪功能开或者关，监听功能开或者关的不好的使用方式，提供更人性化的用户体验和使用感受。

[0146] 4、提供了快速进入监听和返回降噪播放状态的快捷功能，很好地适应了用户临时性、突发性的监听需求，从而提供更好的用户体验和更完善的耳机控制方式。

[0147] 5、提供了机械式可调结构的调节器和终端提供的虚拟调节器，这两种方式都能够实现本发明的目的，也可以结合起来使用。

[0148] 本领域技术人员应当明白，可以通过各种方式来实现上述单元或者模块，例如，可以通过指令配置处理器来实现屏幕创建模块、环境模型解析加载模块，匹配模块和3D影视处理模块。上述各个模块可以是互相独立的单元，也可以是合并在一起实现，

[0149] 本发明可以是系统、方法和/或计算机程序产品。计算机程序产品可以包括计算机可读存储介质，其上载有用于使处理器实现本发明的各个方面的计算机可读程序指令。用于执行本发明操作的计算机程序指令可以是汇编指令、指令集架构(ISA)指令、机器指令、机器相关指令、微代码、固件指令、状态设置数据、或者以一种或多种编程语言的任意组合编写的源代码或目标代码，所述编程语言包括面向对象的编程语言—诸如Smalltalk、C++等，以及常规的过程式编程语言—诸如“C”语言或类似的编程语言。计算机可读程序指令可以完全地在用户计算机上执行、部分地在用户计算机上执行、作为一个独立的软件包执行、部分在用户计算机上部分在远程计算机上执行、或者完全在远程计算机或服务器上执行。

[0150] 这里参照根据本发明实施例的方法、装置(系统)和计算机程序产品的流程图和/或框图描述了本发明的各个方面。应当理解，流程图和/或框图的每个方框以及流程图和/或框图中各方框的组合，都可以由计算机可读程序指令实现。

[0151] 附图中的流程图和框图显示了根据本发明的多个实施例的系统、方法和计算机程序产品的可能实现的体系架构、功能和操作。在这点上，流程图或框图中的每个方框可以代表一个模块、程序段或指令的一部分，所述模块、程序段或指令的一部分包含一个或多个用于实现规定的逻辑功能的可执行指令。在有些作为替换的实现中，方框中所标注的功能也可以以不同于附图中所标注的顺序发生。例如，两个连续的方框实际上可以基本并行地执行，它们有时也可以按相反的顺序执行，这依所涉及的功能而定。也要注意的是，框图和/或流程图中的每个方框、以及框图和/或流程图中的方框的组合，可以用执行规定的功能或动作的专用的基于硬件的系统来实现，或者可以用专用硬件与计算机指令的组合来实现。对于本领域技术人员来说公知的是，通过硬件方式实现、通过软件方式实现以及通过软件和硬件结合的方式实现都是等价的。

[0152] 虽然已经通过例子对本发明的一些特定实施例进行了详细说明，但是本领域的技术人员应该理解，以上例子仅是为了进行说明，而不是为了限制本发明的范围。本领域的技术人员应该理解，可在不脱离本发明的范围和精神的情况下，对以上实施例进行修改。本发明的范围由所附权利要求来限定。

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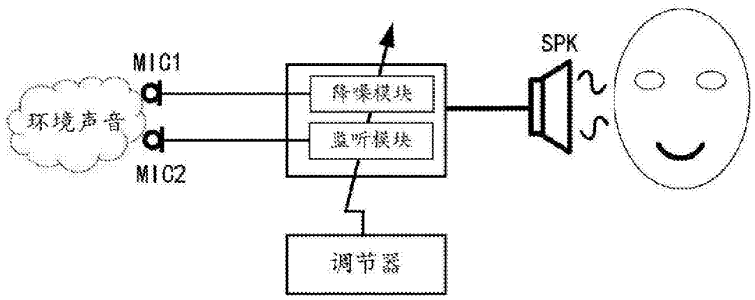


图1

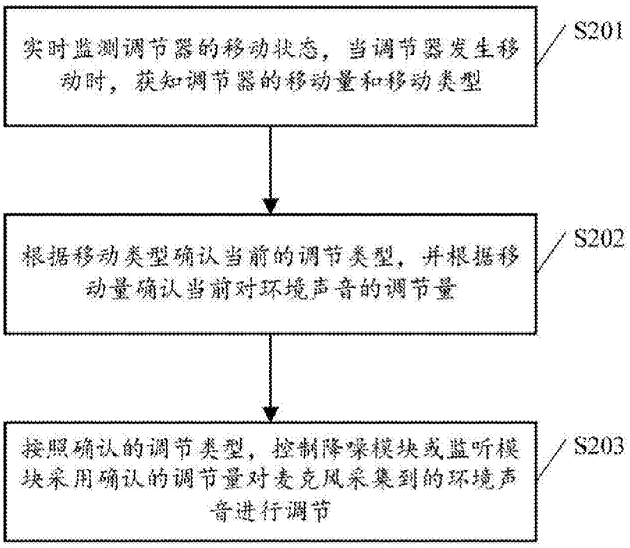


图2

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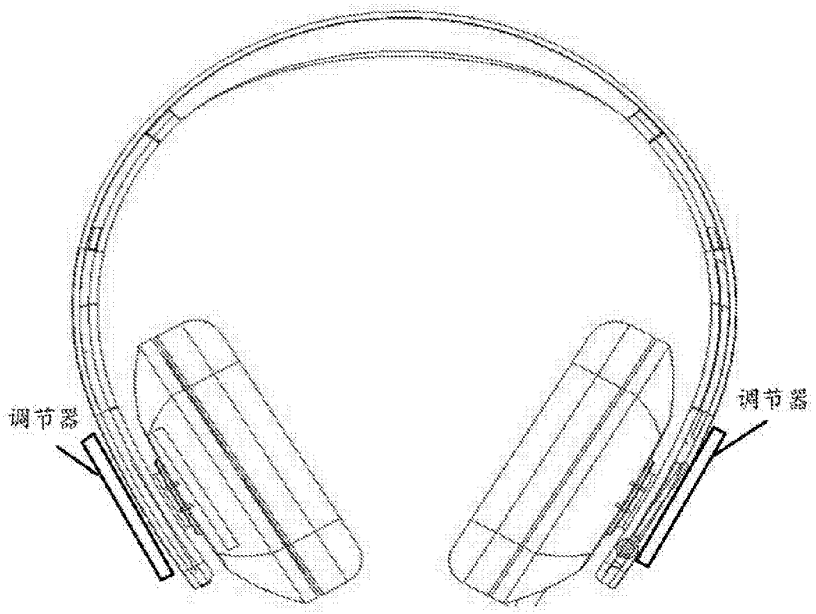


图3

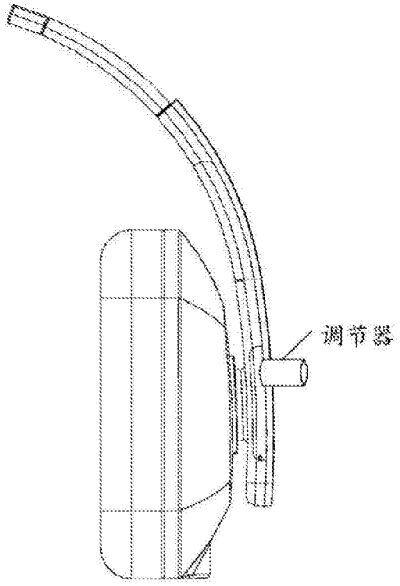


图4

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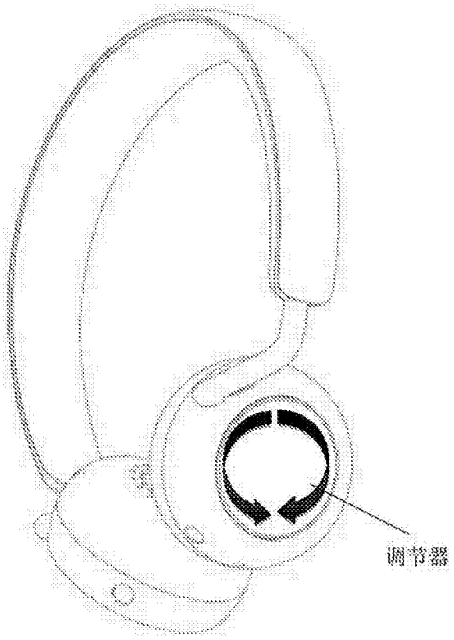


图5

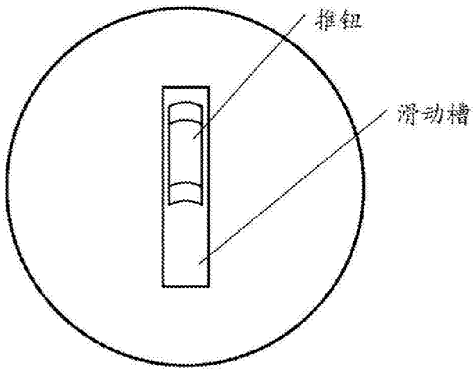


图6

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Page 0380 of 1251

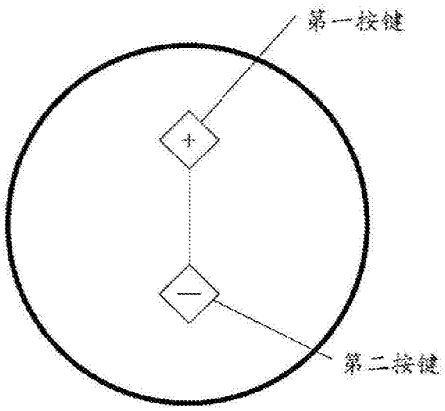


图7

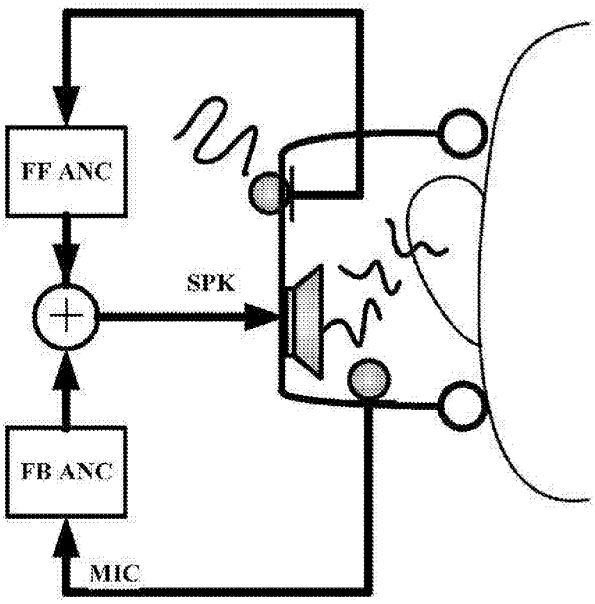


图8

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Page 0381 of 1251

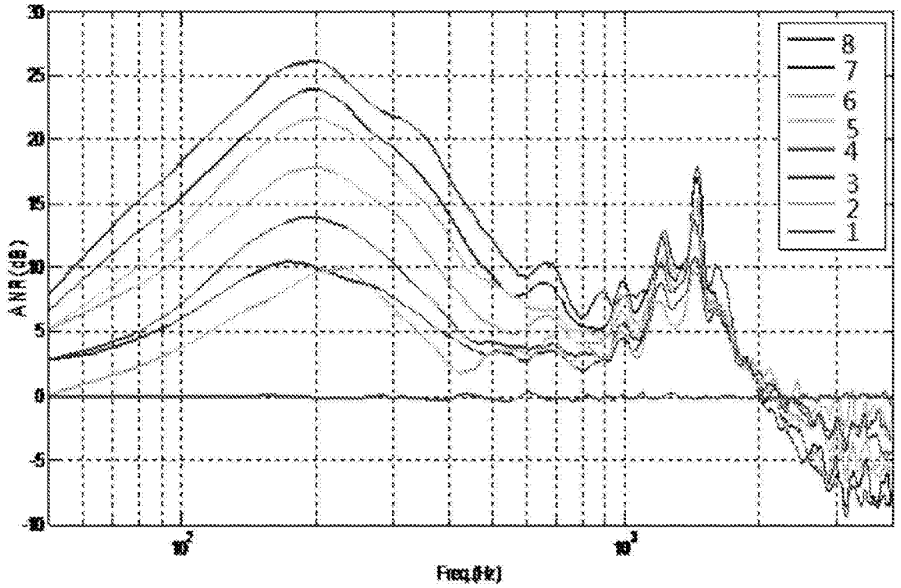


图9

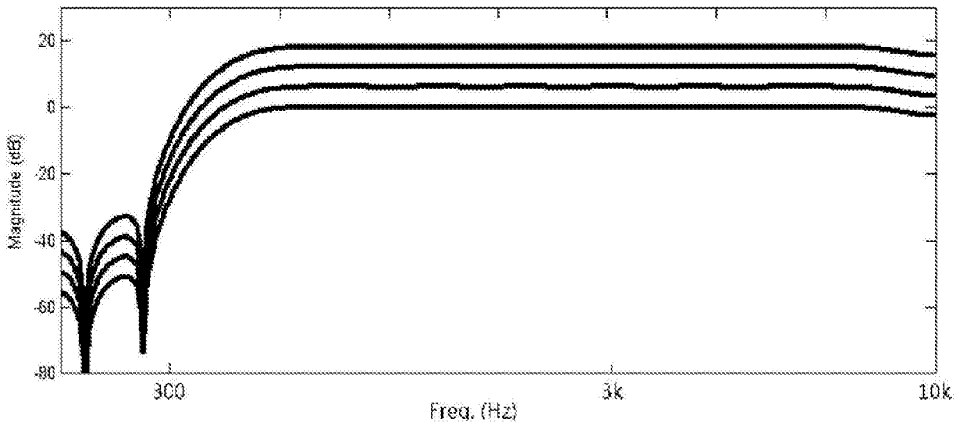


图10

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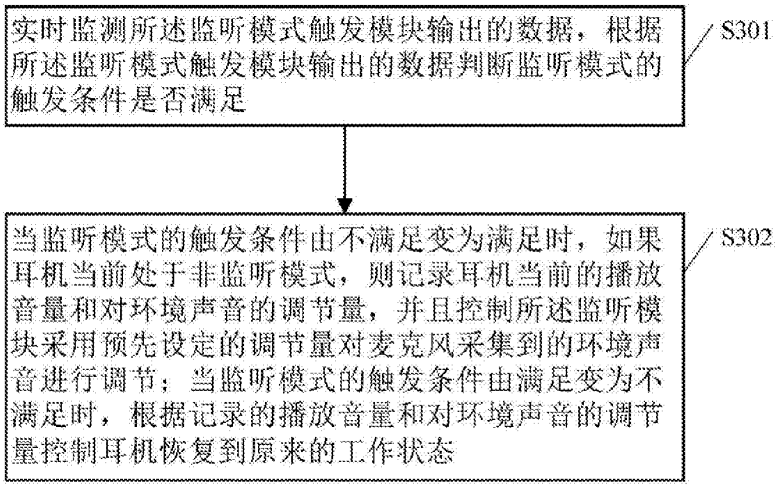


图11



图12

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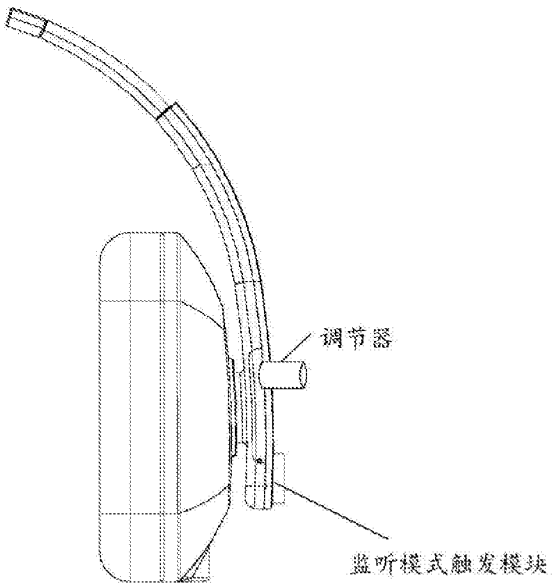


图13

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Page 0384 of 1251

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APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/215,804	**04/07/2023 07:35:57 PM ET**	**11472-39-5_CONT2-16247186**

Title of Invention

METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	-
CONFIRMATION #	9372	FILED BY	Yoshi Hassan
PATENT CENTER #	61889322	FILING DATE	03/29/2021
CUSTOMER #	150139	FIRST NAMED INVENTOR	Steven Wayne Goldstein
CORRESPONDENCE ADDRESS	-	AUTHORIZED BY	Peter Chiabotti

Documents

TOTAL DOCUMENTS: 30

DOCUMENT	PAGES	DESCRIPTION	SIZE (KB)
EP2963647B1.pdf	22	Foreign Reference	858 KB
ES2273616A1.pdf	12	Foreign Reference	1041 KB
KR20080111004A.pdf	23	Foreign Reference	2237 KB
NPL_2_Oshana.pdf	18	Non Patent Literature	1256 KB
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NPL_3_Mulgrew.pdf		14	Non Patent Literature	1992 KB
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CN105554610B.pdf		46	Foreign Reference	6090 KB
11472-39-5_Amendment_with_RCE.pdf		6	-	33 KB
11472-39-5_Amendment_with_RCE-AMSB.pdf	(1-1)	1	Amendment Submitted/Entered with Filing of Continued Prosecution Application (CPA)/Request for Continued Examination(RCE)	17 KB
11472-39-5_Amendment_with_RCE-CLM.pdf	(2-5)	4	Claims	23 KB
11472-39-5_Amendment_with_RCE-REM.pdf	(6-6)	1	Applicant Arguments/Remarks Made in an Amendment	16 KB
[3] [5] [6] MASTER-UPDATED.pdf		4	Information Disclosure Statement (IDS) Form (SB08)	1011 KB
[9] MASTER-BIOMETRIC-IDS.pdf		5	Information Disclosure Statement (IDS) Form (SB08)	1234 KB
11472-39-5_RCE.pdf		3	Request for Continued Examination (RCE)	1628 KB
[8] MASTER_COURT-IDS-UPDATED.pdf		7	Information Disclosure Statement (IDS) Form (SB08)	1011 KB

[3] MASTER-FOREIGN-NONPATENT-IDS-UPDATED.pdf	4	Information Disclosure Statement (IDS) Form (SB08)	1010 KB
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[1] 11472-39-5_IDS.pdf	6	Information Disclosure Statement (IDS) Form (SB08)	1235 KB
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CN105637892A.pdf	21	Foreign Reference	2837 KB
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EP1640972A1.pdf	15	Foreign Reference	637 KB

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Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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	2	10970375		2021-04-06	Devu Manikantan	
	3	10142332		2015-01-05	Sourabh Ravindran	
	4	8851372		2014-10-07	Tiger T G Zhou	
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	Examiner Name	Lunsee Lao
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	1	20090085873		2009-04-02	Betts; William L.	
	2	20100316033		2010-12-16	Atwal; Peter	
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	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	Lunsee Lao
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Name/Print	Peter A. Chiabotti	Registration Number	54603

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(54) System and method for separation of a users voice from ambient sound

(57) The invention relates to a system for separation of a user's voice from ambient sound, comprising a device (10) to be wom at the user's ear or at least partly in the user's ear canal (12) comprising a first microphone (M1) oriented outwardly towards the environment and a second microphone (M2) oriented inwardly towards the user's ear canal, and an audio signal processing unit (24, 26) for processing audio signals from the first and second microphone by a blind source separation algorithm adapted to separate the user's voice from ambient sound.

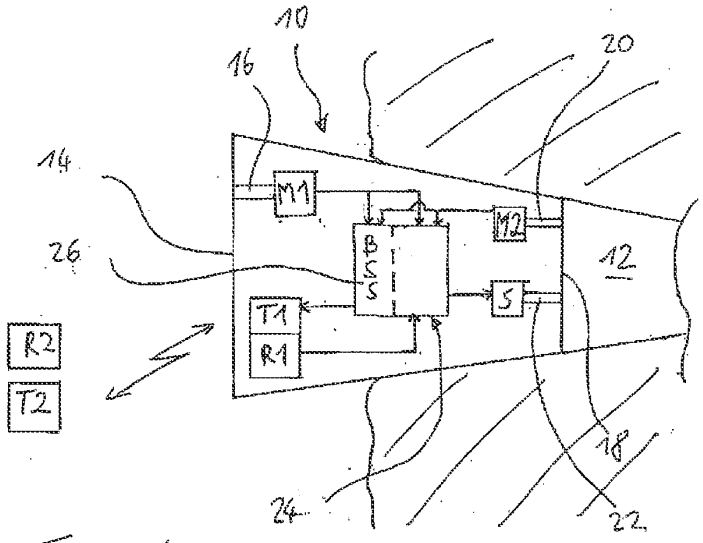


Fig. 1

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Description

[0001] The present invention relates to a system and a method for separation of a user's voice from ambient sound by using at least one device to be worn at the user's ear or at least partly in the user's ear canal.

[0002] For communication purposes, in particular for wireless electronic communication between or from persons exposed to a noisy environment, such as workers in industrial plants, policemen, soldiers, firemen, etc., it is desirable to have a sound pick-up system which is capable at least to some extent to separate the user's voice from ambient noise, or generally ambient sound, in order to improve the intelligibility of the person's speech to the listener, who may be one of the other persons exposed to the noisy environment or who may be a remote person.

[0003] A common approach to achieve such separation of a person's voice is the use of a boom microphone, i.e. a microphone which is placed close to the mouth, carried by a headset, helmet or any other device worn by the person. Such microphone selectively emphasizes the near field around the mouth.

[0004] Other approaches are vibration pick-up devices which are in direct contact with the throat, picking up the vibrations of the vocal chord, or which are in direct contact with the meatus wall or the outer ear canal, picking up the vibrations of the head tissue (i.e. "bone conduction" microphones) or which are in direct contact with the cheek-bone.

[0005] Devices of these types are either fairly sensitive to acoustic noise masking the speech or certain speech sounds are poorly transmitted, especially the high frequency consonant sounds necessary for good intelligibility. Furthermore, for industrial applications boom microphones have the drawbacks that they limit the freedom of movement of the user and that, when combined with a hearing protection device, they will affect the stability and hence the attenuation of the hearing protection device. Bone-conduction microphones have the drawbacks that they have a very limited audio bandwidth which limits the intelligibility of the speech and that they often have to be pressed fairly hard which causes discomfort to the user.

[0006] US 6,661,901 B1 relates to an active hearing protection system comprising an earplug with an outer microphone for picking up ambient sound and an inner microphone which is sealed with respect to ambient sound but is open towards the inner part of the user's ear canal. In an operation mode in which separation of the user's voice from ambient noise is desired only the inner microphone is activated while the outer microphone is not, with the signal from the inner microphone being processed by an electronics unit integrated within the earplug in order to make the user's voice highly natural and intelligible, either for the user himself or his external communication partners.

[0007] Another approach is based on a so-called "blind source separation" (BSS) algorithm in order to separate a person's voice from background noise by corresponding audio signal processing. In this respect, US 2003/0055535 A1 relates to the use of a BSS algorithm for separating the voice of an operator of a vehicle wheel alignment system with a voice audio interface from background noise by using a microphone array in order to avoid the necessity to use a headset. US 2005/0060142 A1 relates in a more general manner to the use of two spaced-apart microphones operated with a BSS algorithm for voice separation from background noise in audio applications.

[0008] It is an object of the invention to provide for a system and a method for separation of a person's voice from ambient sound wherein good intelligibility of speech is achieved while nevertheless discomfort to the person is to be avoided as far as possible.

[0009] According to the invention this object is achieved by a system as defined in claims 1 and 25, and by corresponding methods as defined in claims 28 and 29, respectively.

[0010] The invention is beneficial in that, by using a first microphone and a second microphone, wherein according to the solution of claims 1 and 28 the first microphone is oriented outwardly towards the environment and the second microphone oriented inwardly towards the user's ear canal and according to the solution of claims 25 and 29 the first microphone is located at the right ear and the second microphone is located at the left ear, and by processing the audio signals from the first and second microphone by a blind source separation algorithm, good separation of the user's voice from ambient sound with resulting high intelligibility of the user's speech is achieved without the need for additional restrictions regarding the location of the microphones, so that in particular the need for a boom microphone or a bone-conduction microphone can be avoided. In particular, by orienting the first microphone towards the environment and orienting the second microphone towards the user's ear canal or by locating the first microphone at the right ear and the second microphone at the left ear these two microphones pick up sufficiently differently mixed signals of the ambient sound and the user's voice so that a BSS algorithm will work efficiently.

[0011] Generally, blind source separation (also referred to as "independent component analysis" (ICA)) is a technique for separating mixed source signals (components) which are presumably statistically independent from each other. In its simplified form, blind source separation applies an "un-mixing" matrix of weights to the mixed signals, for example, multiplying the matrix with the mixed signals, to produce separated signals. The weights are assigned initial values, and then adjusted to maximize joint entropy of the signals in order to minimize information redundancy. This weight-adjusting and entropy-increasing process is repeated until the information redundancy of the signals is reduced to a minimum. Because this technique does not require information on the

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source of each signal, it is referred to as "blind source separation". An introduction to blind source separation is found, for example, in US 2005/0060142 A1.

[0012] In the most simple cage, BSS is applied to two different mixtures of two (acoustic) sources, wherein the two different mixtures are obtained by using two spaced apart microphones. Mixing of the two sources can be represented by a matrix A, with the BSS algorithm corresponding mathematically to finding the inverse matrix of A without knowing anything about the matrix nor about the sources, except that they are statistically independent. In the case of a person's voice mixed with background noise the latter assumption usually is valid. The mixtures of the two sources could be different with respect to amplitude and/or phase of the two sources. In other words, by picking up sound signals with two differently oriented microphones the signal of each of these microphones will correspond to a mixture which is different with regard to the difference in amplitude and/or phase of the two acoustic sources (i.e. user's voice on the one hand and ambient noise on the other hand). By orienting one of the microphones outwardly towards the environment and the other microphone inwardly to the ear canal, a particularly large difference between the two mixtures can be obtained in a simple and particularly comfortable manner, i.e. no boom microphones or bone-cunduction microphones which would cause discomfort to the user need to be used.

[0013] According to one embodiment, the two microphones are part of a hearing protection device. With such a configuration, the microphones can be arranged such that the ambient sound reaching the inwardly oriented microphone is attenuated by the hearing protection device relative to ambient sound reaching the outwardly oriented microphone. Although such hearing protection device could be an earmuff, according to a preferred embodiment the hearing protection device is an earplug comprising a shell which is to be inserted at least partially into the user's ear canal. According to one embodiment, the shell is a customized hard shell having an elasticity from shore D85 to D65 and having an outer shape according to the measured inner shape of the user's outer ear and ear canal. According to an alternative embodiment, the shell is a generic soft shell capable of adapting to the shape of the user's outer ear and ear canal.

[0014] Preferably, the inwardly oriented microphone is located at or is open to the inner part of the shell which is to be inserted into the ear canal, with the inwardly oriented microphone preferably being located at the inner end of the shell or within a channel of the shell open to the inner end of the shell.

[0015] Preferably, the outwardly oriented microphone is located at or is open to the outer part of the shell which is not to be inserted into the ear canal, with the outwardly oriented microphone preferably being located at the outer end of the shell or within a channel of the shell open to the outer end of the shell.

[0016] If the hearing protection device is an earmuff, the first microphone is located at or is open to the outer side of the earmuff and the second microphone is located at or is open to the inner side of the earmuff.

[0017] In all embodiments, the device preferably comprises a speaker adapted to provide an external audio signal to the user's ear. Thereby bidirectional communication with the user is achieved. Preferably the speaker is located at or is open to a portion of the device which is to be worn within the user's ear canal, whereby the speaker is brought acoustically close to the user's ear drum so that good intelligibility of the sound provided by the speaker is achieved even in noisy environments.

[0018] Usually the device will be binaural, i.e. it will comprise one unit for the right ear and another unit for the left ear. The speaker and the microphones may be integrated in the same unit, i.e. in at least one of the units, or the speaker may be part of the unit for one ear and the microphones may be part of the unit for the other ear.

[0019] According to one embodiment, the device may be adapted to be worn completely within the user's ear canal, whereby the device can be more or less completely hidden from the views of other persons. According to other embodiments, the device may be any other kind of a wired or wireless headset.

[0020] Preferably the audio signal processing unit is integrated within the device. In an alternative embodiment, the audio signal processing unit may be adapted to be worn behind the user's ear or somewhere at the user's body. In this case, although the device becomes more handy, more space, and hence also more power, is available for the audio signal processing unit, which may reduce the costs an/or may improve the performance of the blind source separation. In such case, the audio signal processing unit may be connected to the microphone either by wires, which is the most simple solution, or via a wireless link, for example, a radio frequency link such as Bluetooth link, an inductive link or an infrared link, which solution would result in enhanced wearing comfort for the user.

[0021] As a further alternative, the audio signal processing unit could be designed to be located remote from the user and is connected to the microphones via a radio frequency link such as Bluetooth link. In this case, there would be even less restrictions regarding the size and power consumption of the audio signal processing unit, which might result in reduced costs and/or increased performance of the blind source separation.

[0022] In all embodiments, the system preferably comprises a radio frequency transmitter for transmitting the processed audio signal output of the audio signal processing unit to a remote radio frequency receiver in order to provide the user's voice to another person. According to one embodiment, the radio frequency transmitter is integrated within the audio signal processing unit, i.e. the radio frequency transmitter may be either integrated within the device or it may be adapted to be worn behind the use's ear or at the user's body or it could be even located remote from the user. However, in an

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alternative embodiment, the radio frequency transmitter could be remote from the audio signal processing unit, in which case the processed audio signal output of the audio signal processing unit would be provided to a radio frequency transmitter by wires or via a radio frequency link, an inductive link or an infrared link. In this case the location of the audio signal processing unit and the radio frequency transmitter can be optimized independently form each other.

[0023] In all embodiments, the blind source separation algorithm preferably works in the frequency domain, i.e. the algorithm is simultaneously carried out in different frequency bands/bins and the outcomes of these bands/bins are combined in an appropriate way.

[0024] In all embodiments, the blind source separation algorithm preferably works with the assumption that the sources are statistically independent, i.e. that the user's voice is independent of the ambient sound.

[0025] In the following, examples of the invention will be explained in detail by reference to the attached drawings.

Fig. 1 shows a schematic view of a first embodiment of a hearing protection earplug system according to the invention inserted into a person's ear canal with a block diagram of the most relevant electronic components;

Fig. 2 shows a schematic view of a modified embodiment of a hearing protection earplug according to the invention;

Figs. 3 and 4 show further modified embodiments of a system according to the invention;

Fig. 5 shows a modification of the embodiment of Fig. 1; and

Fig. 6 shows a modification of the embodiment of Fig. 5.

[0026] The system of Fig. 1 comprises a hearing protection earplug 10 which may have a hard shell with an elasticity from shore D85 to shore D65, which is customized, i.e. it has an outer shape according to the individual measured inner shape of the user's outer ear and ear canal. The hard shell may be manufactured by layer-by-layer laser sintering of a powder material, for example, polyamide powder, or by laser stereo-lithography or photo-polymerization. An overview regarding such additive layer-by-layer build-up processes for manufacturing customized shells of hearing devices can be found, for example, in US 2003/0133583 A1 or US 6,533,062 B1. The inner shape of the person's ear canal and outer ear can be measured, for example, by taking an impression which then undergoes laser scanning or by direct laser scanning of the ear.

[0027] Preferably the hard shell is designed such that it provides for an acoustic attenuation, averaged over the audible frequency range, of at least 10 dB when inserted into the user's ear canal.

[0028] Rather than having a customized hard shell, earplug 10 may have a generic soft shell which adapts to the shape of the users outer ear and ear canal due to its elasticity.

[0029] In Fig. 1 the earplug 10 is shown to be inserted with its inner end portion into a user's ear canal 12. The earplug 10 comprises an outer microphone M1, which is located at the outer part of the earplug 10 and which is open to the outer end 14 of the earplug 10 via a sound channel 16, and an inner microphone M2, which is located at the inner part of the earplug 10 and which is open to the inner end 18 of the earplug 10 via a sound channel 20. The earplug 10 further comprises a speaker S which is likewise located at the inner part of the earplug 10 and is open to the inner end 18 via a sound channel 22. The earplug 10 also includes a digital audio signal processing unit 24 comprising a digital signal processor as well as a radio frequency transmitter unit T1 and a radio frequency receiver unit R1. The audio signal processing unit 24 functionally includes a unit 26 for applying a blind source separation algorithm to the audio signals provided by the microphones M1 and M2. It is evident to the skilled person that the earplug 10 includes additional electrical components, such as amplifiers and analogue-to-digital converters for the audio signals provided by the microphones M1 and M2, a digital-to-analogue converter and an amplifier for the audio signal to be transformed into sound by the speaker S, program and data memory components for the audio signal processing unit 24, a battery for power supply, a demodulator, a modulator, an antenna, etc., which are not shown in Fig. 1 for the sake of simplicity.

[0030] The transmitter T1 of the earplug 10 is adapted to transmit audio signals from the earplug 10 to a remote receiver R2 via a radio link, while the receiver R1 of the earplug 10 is adapted to receive audio signals from a remote transmitter T2. The audio signals received by the receiver R1 are demodulated and then undergo signal processing in the audio signal processing unit 24 as input to the speaker S in order to provide remote audio signals to the user. Such remote audio signals could be the speech of another person picked up by a microphone whose output is sent to the remote transmitter T2 by wires or via, for example, a mobile telephone or mobile radio device.

[0031] The audio signals provided by the microphones M1 and M2 are passed as input to the blind source separation unit 26, in which a processed audio signal output for the transmitter T1 is produced, with the processed audio signal output consisting completely or at least essentially of the user's voice which has been separated from the ambient sound by action of the blind source separation algorithm carried out in the BSS unit 26. Such BSS signal processing utilizes the fact that the sound

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mixtures picked up by the microphone M1 which is oriented towards the environment and M2 which is oriented towards the ear canal 12, respectively, consist - due to the different orientation of the microphones M1 and M2 - of essentially different mixtures of the ambient sound and the user's voice, which are different regarding amplitude ratio of these two signal contributions or sources (i.e. ambient sound on the one hand and user's voice on the other hand) and regarding phase difference of these two signal contributions of the mixture.

[0032] The output signal of the BSS unit 26 is transmitted via the transmitter T1 to the remote receiver R2 which usually will be connected to a remote speaker for presenting the user's voice to another person. The remote speaker and the remote microphone connected to the remote transmitter T2 could be part of an earpiece or an earplug worn by another person, which may be similar or identical to the earplug 10. Thereby in-the-ear hearing protection devices with integrated communication function can be achieved. Such a system could be used by any persons who need to communicate in a noisy environment, such as workers, soldiers, firemen, etc. However, the remote receiver R2 also might serve for communication via a mobile telephone or a mobile radio device.

[0033] In general, the remote receiver R2 / transmitter T2 could be a part of an interface of a standard wireless communication device, such as a mobile telephone device or a mobile radio device Preferably, the wireless link between the transmitter T1 / receiver R1 and the remote receiver R2 / remote transmitter T2 is a Bluetooth link. The remote receiver R2 / transmitter T2 then could be a part of the Bluetooth interface of a standard wireless communication device, such as a mobile telephone device or a mobile radio device.

[0034] According to Fig. 1, the audio signal produced by the microphones M1 and M2, which is used as input to the BSS unit 26, in addition could undergo other audio signal processing within the audio signal processing unit 24, for example, in order to provide the ambient sound picked up by the microphone M1 to the speaker S for enabling perception of desired ambient sounds during times when the noise level is low so that the hearing protection function provided by the earplug 10 is not needed, or to provide for an in-situ measurement of the actual acoustic attenuation provided by the earplug 10 when inserted into the user's ear canal 12 by comparing the sound levels measured by the microphones M1 and M2.

[0035] In Fig. 2 a modified embodiment is shown, wherein the transmitter T1 and the receiver R1 have been removed from the earplug 10 and now are located in an external unit 28 which is connected by wires 30 to the earplug 10 and which is to be worn at an appropriate location at the user's body, for example behind the user's ear or somewhere within or below the user's clothing, for example, within a helmet. The external unit 28 may include further electrical components removed from the earplug 10, for example, the power supply battery.

[0036] In Fig. 3 a further modified embodiment is shown, wherein not only the transmitter T1 and the receiver R1 (and the power supply battery) but in addition also the audio signal processing unit 24 including the BSS unit 26 has been removed from the earplug 10 and is connected via corresponding wires to the microphones M1 and M2 and the speaker S, respectively.

[0037] In Fig. 4 a still further modified embodiment is shown, wherein the wire connection 30 between the earplug 10 and the external unit 28 has been replaced by a bidirectional wireless link 32, which is established by a transmitter T3 and a receiver R3 located in the earplug 10 and a corresponding receiver R4 and transmitter T4 located in the external unit 28. The wireless link 32 could be a radio frequency link, an inductive link or an infrared link. If it is an inductive link, the external unit 28 would have to be located relatively close to the earplug 10, for example at a distance of not more than 20 cm. With an RF link, the distance could be larger; in this case it would be even be possible to locate the external unit 28 at a position remote from the user's body, for example, within a distance of a few meters. Preferably the wireless link 32 of the embodiment of Fig. 4 is a Bluetooth link.

[0038] Although in the embodiments shown is Figs. 1 to 4 the speaker S is shown as being integrated in the same earplug 10 as the microphones M1 and M2, in other embodiments the speaker S may be provided at an earplug other than the earplug at which the microphones M1 and M2 are provided.

[0039] Fig. 5 shows a modified version of the embodiment of Fig. 1, wherein the inwardly oriented second microphone M2 is not part of the earplug 10 — as shown in Fig. 1 — but rather is provided at the earplug 11 for the other ear. The second microphone M2 is located at the inner part of the earplug 11 and is open to the inner end 18 of the earplug 11 via a sound channel 20. In addition, the earplug 11 further comprises a speaker S which is likewise located at the inner part of the earplug 11 and is open to the inner end 18 via a sound channel 22. The microphone M2 and the speaker S of the earplug 11 are connected to the audio signal processing unit 24 located in the earplug 10 via a wire connection 40 which be integrated in a mechanical connection (not shown) between the earplugs 10 and 11 which may be provided for preventing loss of the earplugs 10, 11.

[0040] According to a modification of the embodiment of Fig. 5, the outwardly oriented first microphone M1 may be provided at the earplug 11, while the inwardly oriented second microphone M2 may be provided at the earplug 10 as in Fig. 1.

[0041] An alternative modification of the embodiment of Fig. 5 is shown in Fig. 6, wherein both the first microphone M1 of the earplug 10 and the second microphone M2 of the earplug 11 are outwardly oriented. In this case the differences in the sound mixtures picked up by the microphone M1 of the earplug 10 and the microphone M2 of the earplug 11 are due to the different location of

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the microphone M1 at one ear and the microphone M2 at the other ear (for example, a disturbing noise source usually will be located closer to one of the two ears and hence will contribute more strongly and with different phase to the signal picked up by the microphone located at that ear than to that located at the other ear).

[0042] According to a modification of the embodiment of Fig. 6 both microphones M1 and M2 could be inwardly oriented rather than outwardly oriented.

[0043] While Figs. 5 and 6 and the above-discussed variants thereof are modifications of the embodiment of Fig. 1, according to which both the audio signal processing unit 24 and the transmitter T1 / receiver R1 are part of the earplug 10, it is to be understood that the embodiments of Figs. 2 to 4 may be analogously modified, i.e. by providing the second microphone M2 at the earplug 11 rather than at the earplug 10.

[0044] While the examples discussed so far relate to an application of the communication system to hearing protection earplugs, the invention also may be applied to earmuffs serving as a hearing protection device or generally to any kind of headset, i.e. also to hearing devices which do not provide for a hearing protection function. For example, the headset may consist of a device with a customized or a generic shell which is designed such that it can be worn completely in the ear canal 12 (i.e. as a "CIC device") and which serves exclusively communication purposes, for example, for security persons, policemen, firemen, etc., i.e. for persons who are exposed to a noisy environment, with the noise level, however, being below a threshold value which would require the use of hearing protection devices. Such CIC device generally could have the construction of the earplug of Fig. 1, however, without considerable acoustic attenuation provided due to the material and the design of the shell, and would be reduced in size so that it can be worn completely within the ear canal 12.

[0045] In general, the device need not be designed as an earplug but it may rather be any kind of headset.

Claims

1. A system for separation of a user's voice from ambient sound, comprising a device (10) to be worn at the user's ear or at least partly in the user's ear canal (12) comprising a first microphone (M1) oriented outwardly towards the environment and a second microphone (M2) oriented inwardly towards the user's ear canal, and an audio signal processing unit (24, 26) for processing audio signals from the first and second microphone by a blind source separation algorithm adapted to separate the user's voice from ambient sound.

2. The system of claim 1, wherein the device (10) is designed as a hearing protection device.

3. The system of claim 2, wherein the hearing protection device (10) is designed to provide for an acoustic attenuation, averaged over the audible frequency range, of at least 10 dB.

4. The system of claim 3, wherein the microphones (M1, M2) are arranged such that ambient sound reaching the second microphone (M2) is attenuated by the hearing protection device (10) relative to ambient sound reaching the first microphone (M1).

5. The system of one of claims 2 to 4, wherein the hearing protection device is an earplug (10) comprising a shell which is to be inserted at least partially into the user's ear canal (12).

6. The system of claim 5, wherein the shell is a hard shell having an elasticity from Shore D 85 to Shore D 65 and having an outer shape according to the measured inner shape of the user's outer ear and ear canal (12).

7. The system of one of claims 5 and 6, wherein the second microphone (M2) is located at or is open to the inner part of the shell which is to be inserted into the ear canal (12).

8. The system of claim 7, wherein the second microphone (M2) is located at the inner end of the shell or within a channel (20) of the shell open to the inner end of the shell.

9. The system of one of claims 5 to 8, wherein the first microphone (M1) is located at or is open to the outer part of the shell which is not to be inserted into the ear canal (12).

10. The system of claim 9, wherein the second microphone (M2) is located at the outer end of the shell or within a channel (16) of the shell open to the outer end of the shell.

11. The system of one of claims 1 to 4, wherein the hearing protection device is an earmuff and wherein the first microphone is located at or is open to the outer side of the earmuff and the second microphone is located at or is open to the inner side of the earmuff.

12. The system of one of the preceding claims, wherein the device comprises a speaker (S) adapted to provide an external audio signal to the user's ear.

13. The system of claim 12, wherein the speaker (S) is located at or within a channel open to that end of the device which is to be worn within the user's ear canal.

14. The system of one of the preceding claims, wherein the device is adapted to be worn completely within

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the user's ear canal.

15. The system of one of the preceding claims, wherein the audio signal processing unit (24, 26) is integrated within the device (10).

16. The system of one of claims 1 to 14, wherein the audio signal processing unit (24, 26) is adapted to be worn behind the user' s ear or at the user's body.

17. The system of claim 16, wherein the audio signal processing unit (24, 26) is connected to the microphones (M1, M2) by wires (30) or via a radio frequency link (32) such as a Bluetooth link, an inductive link or an infrared link.

18. The system of one of claims 1 to 14, wherein the audio signal processing unit (24, 26) is designed to be located remote from the user and is connected to the microphones (M1, M2) via a radio frequency link (32) such as a Bluetooth link.

19. The system of one of the preceding claims, wherein the system comprises a radio frequency transmitter (T1) for transmitting the processed audio signal output of the audio signal processing unit (24, 26) to a remote radio frequency receiver (R2) in order to provide the user's voice to another person.

20. The system of claim 19, wherein the remote radio frequency receiver (R2) is part of a Bluetooth interface of a mobile telephone or a mobile radio system.

21. The system of claim 19 or 20, wherein the radio frequency transmitter (T1) is integrated with the audio signal processing unit (24, 26).

22. The system of claim 19 or 20, wherein the radio frequency transmitter (T1) is remote from the audio signal processing unit (24, 26) and wherein the processed audio signal output of the audio signal processing unit is provided to the radio frequency transmitter by wires (30) or via an inductive link.

23. The system of claim 12, wherein the system comprises a radio frequency receiver (R1) for receiving the external audio signal from a remote radio frequency transmitter (T2).

24. The system of claim 23, wherein the remote radio frequency transmitter (T2) is part of a Bluetooth interface of a mobile telephone or a mobile radio system.

25. A system for separation of a user's voice from ambient sound, comprising: a first device (10) to be worn at the user's right ear or at least partly in the user's right ear canal (12), a second device (11) to be worn at the user's left ear or at least partly in the user's left ear canal (12), the first device comprising a first microphone (M1) and the second device comprising a second microphone (M2), and an audio signal processing unit (24, 26) for processing audio signals from the first and second microphone by a blind source separation algorithm adapted to separate the user's voice from ambient sound.

26. The system of claim 25, wherein one of the first microphone (M1) and second microphone (M2) is oriented outwardly towards the environment and the other one of the first microphone and second microphone is oriented inwardly towards the user's ear canal (12).

27. The system of one of claims 25 and 26, wherein the audio signal processing unit (24, 26) is located in one of the first device (10) and the second device (11) or is adapted to be worn behind the user's ear or at the user's body or is designed to be located remote from the user and is connected to the microphones via a radio frequency link (32) such as a Bluetooth link.

28. A method for separation of a user's voice from ambient sound, comprising:

providing the user with a device (10) to be worn at a user's ear or at least partly in the user's ear canal (12) and comprising a first microphone (M1) oriented towards the environment and a second microphone (M2) oriented towards the user's ear canal (12),
picking up sound by the first microphone to create a first audio signal and by the second microphone to create a second audio signal, and
processing the first and second audio signals by a blind source separation algorithm in order to produce a processed audio signal wherein the user's voice is separated from ambient sound.

29. A method for separation of a user's voice from ambient sound, comprising:

providing the user with a first device (10) to be worn at the user's right ear or at least partly in the user's right ear canal (12) and a second device (11) to be worn at the user's left ear or at least partly in the user's left ear canal (12), the first device comprising a first microphone (M1) and the second device comprising a second microphone (M2),
picking up sound by the first microphone to create a first audio signal and by the second microphone to create a second audio signal, and
processing the first and second audio signals by a blind source separation algorithm in order to

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produce a processed audio signal wherein the user's voice is separated from ambient sound.

30. The method of one of claims 28 and 29, wherein the blind source separation algorithm works in the frequency domain.

31. The method of one of claims 28 to 30, wherein the processed audio signal is provided to another person via a wireless link such as a Bluetooth link.

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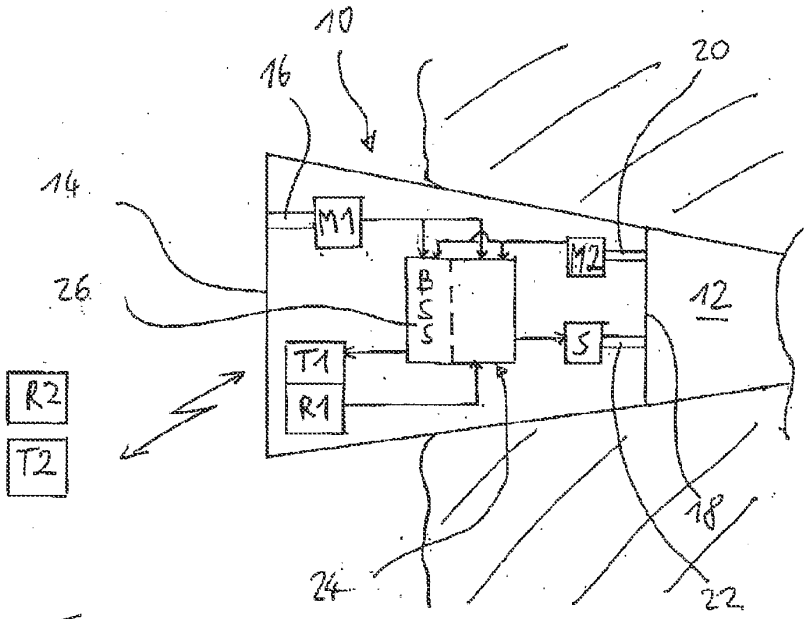


Fig. 1

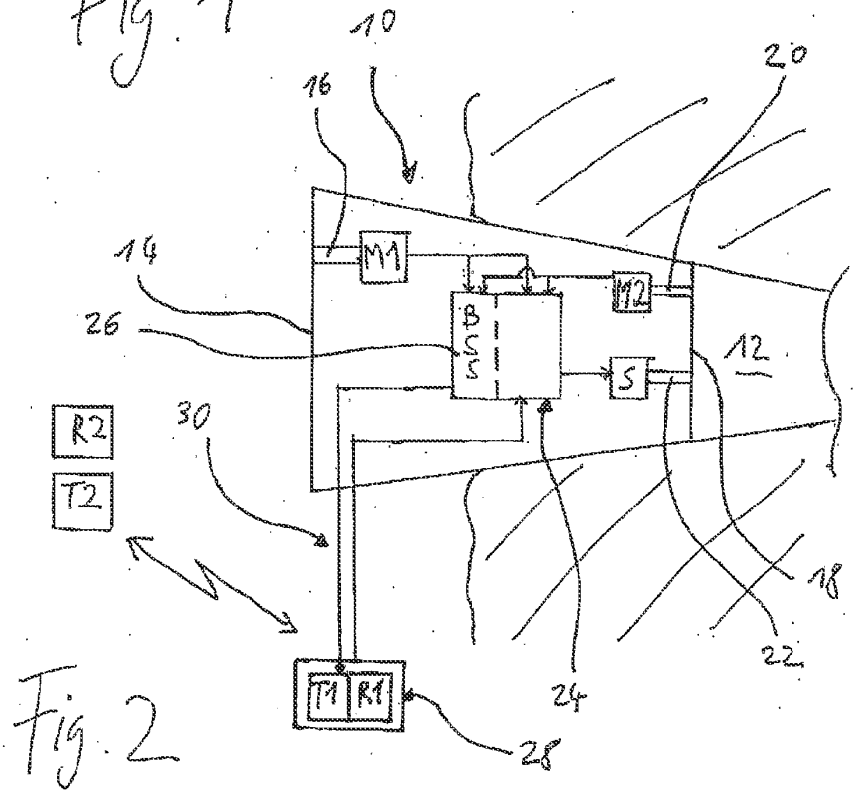


Fig. 2

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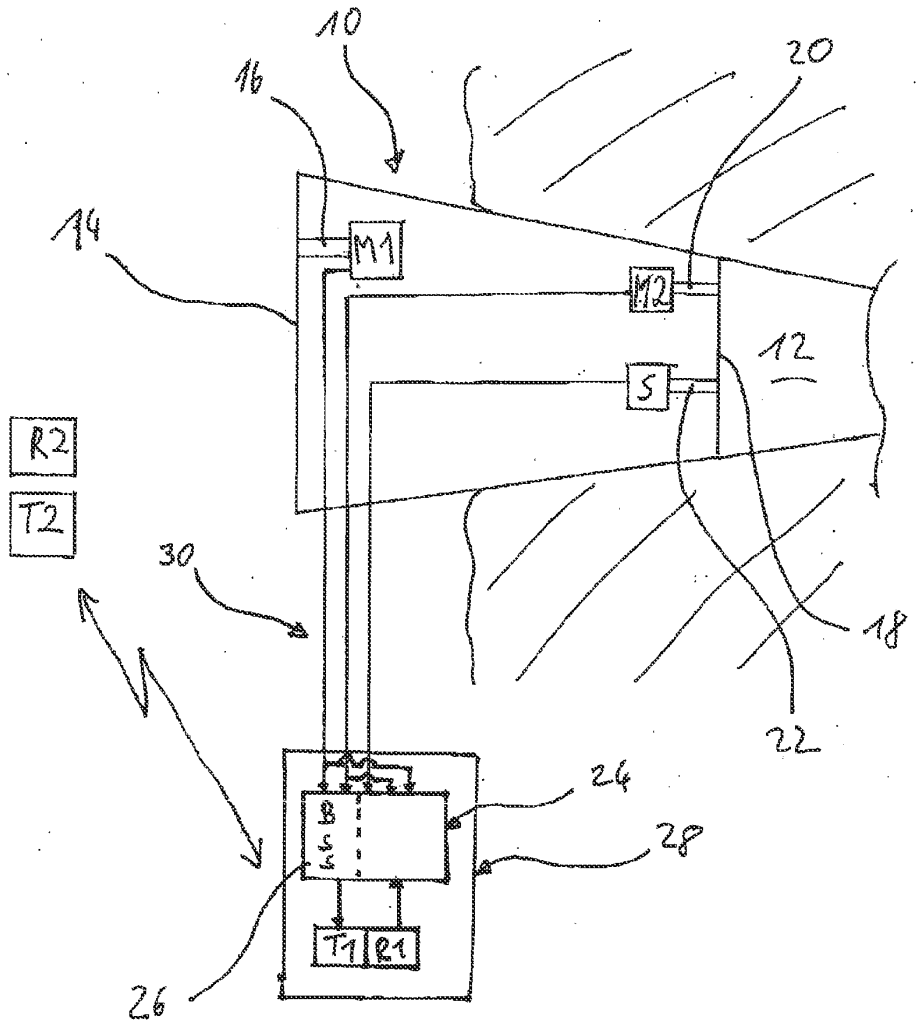


Fig. 3

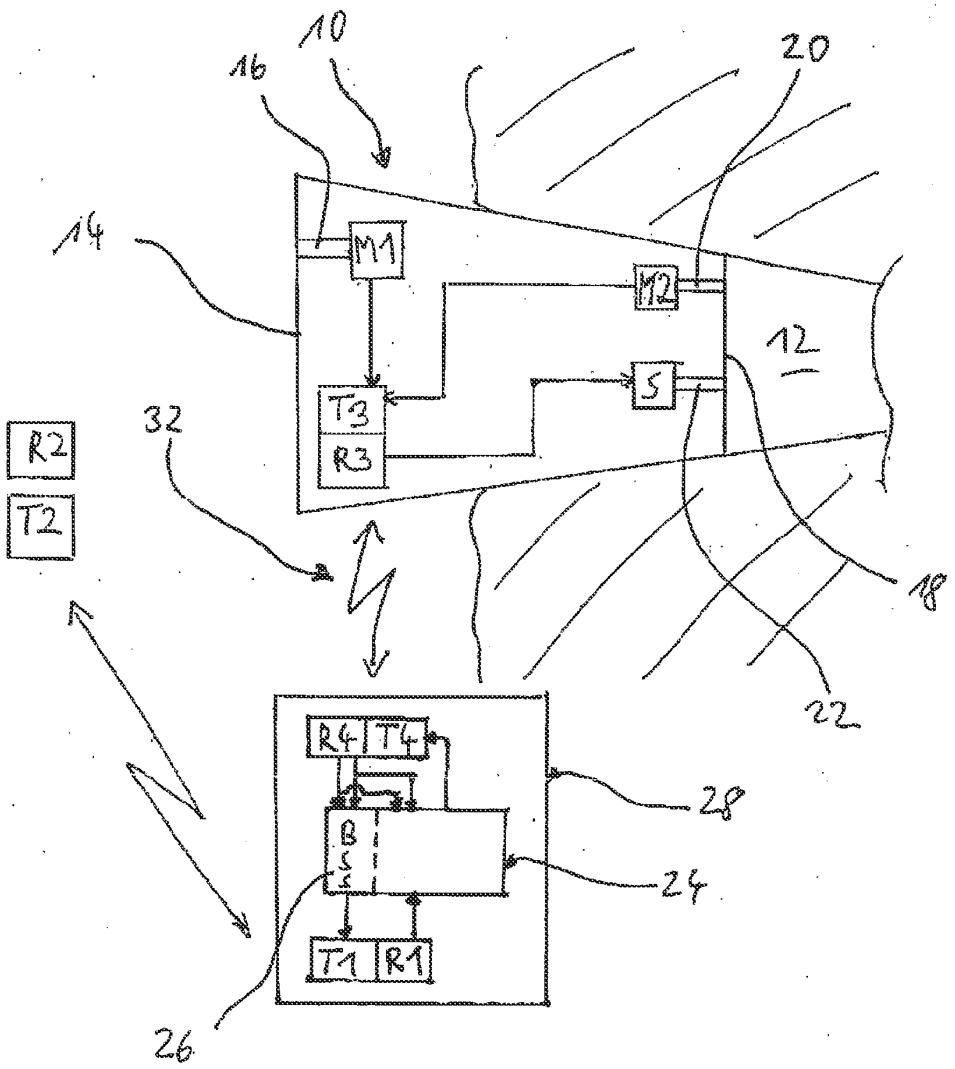


Fig. 4

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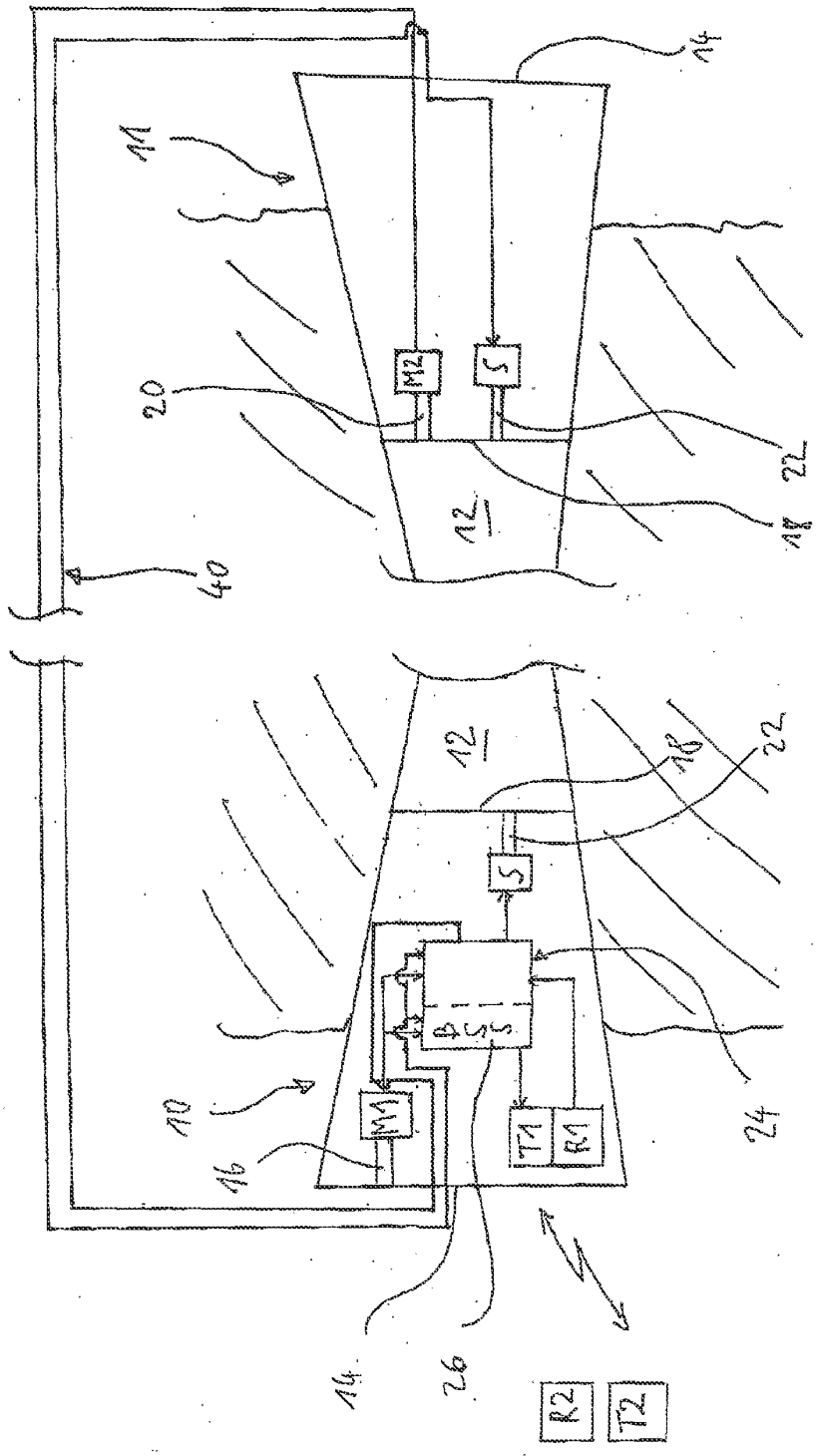


Fig. 5

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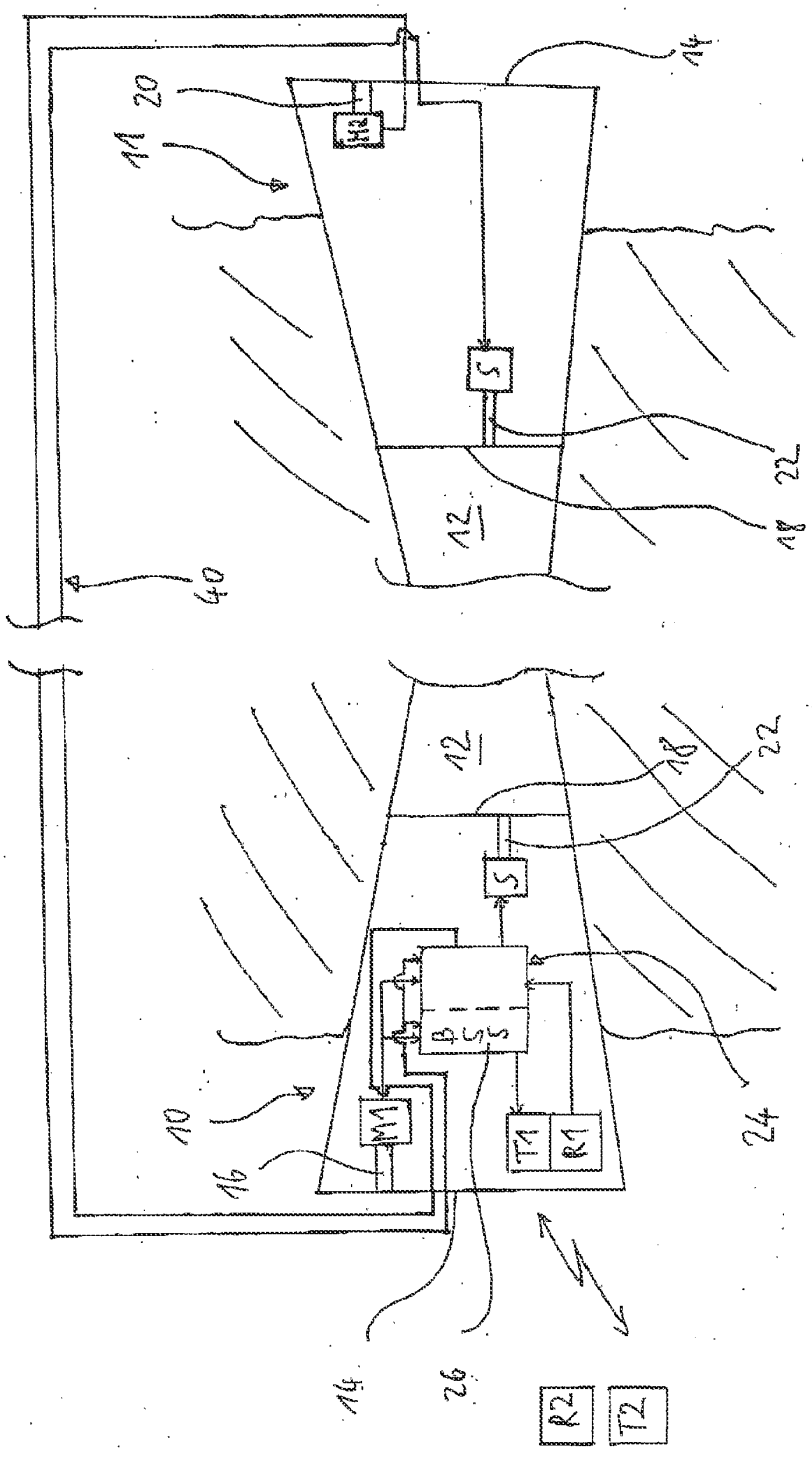


Fig. 6

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European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 05 02 8366

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
X Y	EP 1 509 065 A (BERNAFON AG) 23 February 2005 (2005-02-23) * page 2, paragraph 1 - page 3, paragraph 14 * * page 5, paragraphs 15,16 * * page 6, paragraphs 23,27 * * claim 1 *	25,29,30 26,27,31	G10L21/02 H04R25/00 H04R3/00 H04R1/10
Y,D	US 6 661 901 B1 (SVEAN JARLE ET AL) 9 December 2003 (2003-12-09) * column 1, line 5 - column 2, line 33 * * column 4, line 21 - line 32 * * column 5, line 26 - column 7, line 60 * * column 10, line 52 - column 11, line 40 * * claims 10,11 *	1-24, 26-28,31	
Y	US 2005/060142 A1 (VISSER ERIK ET AL) 17 March 2005 (2005-03-17) * page 1, paragraph 5-7 * * page 3, paragraph 19 - page 5, paragraph 37 * * page 8, paragraph 61 *	1-24,28	
A	SANG MIN LEE ET AL: "New idea of hearing aid algorithm to enhance speech discrimination in a noisy environment and its experimental results" SAN FRANCISCO, CA, USA 1-5 SEPT. 2004, PISCATAWAY, NJ, USA,IEEE, US, vol. 3, 1 September 2004 (2004-09-01), page 978Vol2, XP010774935 ISBN: 0-7803-8439-3 * the whole document *	1-31	

TECHNICAL FIELDS SEARCHED (IPC)
G10L
H04R

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
Munich	3 February 2006	Coda, R

CATEGORY OF CITED DOCUMENTS

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

PTO/SB/08a (02-18)
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First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	Lunsee Lao
Attorney Docket Number	11472-39-5_CONT2-16247186

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