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

Application Number	
Filing Date	
First Named Inventor	John Usher
Art Unit	
Examiner Name	Unknown
Attorney Docket Number	CONT-16838277

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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
	1	7801318		2010-09-21	Barthel	
	2	8774433		2014-07-08	Goldstein	
	3	7801318		2010-09-21	Barthel, Roland	
	4	8401200		2009-11-19	Tiscareno	
	5	5138722		2010-09-21	Urella	
	6	5764781		1998-06-09	Ding	
	7	6321070		2001-11-20	Clark	
	8	9293128		2016-03-22	Goldstein	

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	Application Number	
	Filing Date	
	First Named Inventor	John Usher
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	Examiner Name	Unknown
	Attorney Docket Number	CONT-16838277

	9	4020297		1977-04-26	Brodie S Dan	
	10	5142587		1992-08-25	Kobayashi	
	11	5319163		1994-06-07	Scott	
	12	5712453		1998-01-27	Bungardt	
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	18	8971561		2015-03-03	Howes et al.	
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	First Named Inventor	John Usher
	Art Unit	
	Examiner Name	Unknown
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	2	20140247952		2014-09-04	Goldstein	
	3	20130177179		2013-07-11	Ambrose	
	4	20180176681		2018-06-21	Asada	

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	1	107071123	CN		2017-08-18			

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	Application Number	
	Filing Date	
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	Art Unit	
	Examiner Name	Unknown
	Attorney Docket Number	CONT-16838277

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	Filing Date	
	First Named Inventor	John Usher
	Art Unit	
	Examiner Name	Unknown
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2021-02-23
Name/Print	John P. Keady	Registration Number	56389

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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.

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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM

As the below named inventor, I hereby declare that:

This declaration is directed to:

- [■] The attached application, or
- [ ] United States application or PCT international application number _____ filed on _____.

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

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

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

WARNING:

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LEGAL NAME OF INVENTOR

Inventor: John Usher Date (Optional): 13-Aug-2015

Signature: J. Usher

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

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Document Description: Oath or declaration filed

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DECLARATION FOR UTILITY OR DESIGN PATENT APPLICATION (37 CFR 1.63)

☐ Declaration Submitted With Initial Filing OR ☒ Declaration Submitted After Initial Filing (surcharge (37 CFR 1.16(f)) required)

Attorney Docket Number	PRS-160US2
First Named Inventor	John Usher
COMPLETE IF KNOWN	
Application Number	14/827,332
Filing Date	08-17-2015
Art Unit	2654
Examiner Name	KIM, PAUL

ACOUSTIC SEALING ANALYSIS SYSTEM

(Title of the Invention)

As a below named inventor, I hereby declare that:

This declaration is directed to:

☐ The attached application,

OR

☒ United States Application Number or PCT International application number 14/827,332

filed on 08-17-2015.

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

I believe I am the original inventor or an original joint inventor of a claimed invention in the application.

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

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DECLARATION — Utility or Design Patent Application

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LEGAL NAME OF SOLE OR FIRST INVENTOR:

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

John P Keady

Inventor's Signature	Date (Optional)
/John P. Keady/	

Residence: City	State	Country
Fairfax Station	VA	USA

Mailing Address

8138 Crestridge Rd.

City	State	Zip	Country
Fairfax Station	VA	22039	USA

☐ Additional inventors are being named on the _____ Supplemental sheet(s) PTO/AIA/10 attached hereto

ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of U.S. Patent Application Ser. No. 16/838,277, filed April 2, 2020, which is a continuation of U.S. Patent Application Ser. No. 16/414,136, filed May 16, 2019, which is a continuation of U.S. Patent Application Ser. No. 15/700,511, filed September 11, 2017, which is a continuation of U.S. Patent Application Ser. No. 14/827,332, filed August 17, 2015, now U.S. Patent No. 9,781,530, which is a continuation of U.S. Patent Application Ser. No. 14/054,015, filed October 15, 2013, now U.S. Patent No. 9,113,267, which is a Divisional Application of U.S. Application Ser. No. 12/555,864, filed September 9, 2009, now U.S. Patent No. 8,600,067 and claims the benefit of U.S. Provisional Patent Application No. 61/098,250 filed September 19, 2008. The disclosure of all the aforementioned references are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to testing the seal of an orifice-inserted device, and more particularly, though not exclusively, to a device and method for determining if an earpiece is sealed correctly in an ear canal.

BACKGROUND OF THE INVENTION

[0003] It can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

[0004] Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient sound to provide adequate performance to the user. Additionally, user conditions may change substantially

during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

[0005] Broadly stated, embodiments are directed to a device and method to determine if an earpiece is sealing within the design specification of the device.

[0006] In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume. The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

[0007] At least one exemplary embodiment is directed to a method of detecting sealing integrity of an earpiece comprising the steps of: providing a test signal; generating an acoustic signal corresponding to the test signal incident on an ear canal side of a sealing section; converting the acoustic signal incident on a first side of the sealing section to an electrical signal; and cross- correlating the test signal to the electrical signal where the earpiece is sealed correctly when a cross-correlation between the test signal and the electrical signal is above a threshold.

[0008] At least one exemplary embodiment is directed to a method of adjusting attenuation of an earpiece comprising the steps of: relating cross-correlation of a test signal and a measured acoustic signal in an ear canal of a user to an attenuation level of a sealing section of the earpiece; comparing the attenuation level of the sealing section of the earpiece to a minimum attenuation value; and adjusting a pressure of the sealing section to meet the minimum attenuation value.

[0009] At least one exemplary embodiment is directed to a device comprising: a sealing section configured to seal a user's orifice, where the sealing section is configured to produce an acoustic seal between a first side of the sealing section and a second side of the sealing section; a transducer configured to generate a first acoustic signal incident on the first side of the sealing section; and a first microphone configured to measure a second acoustic signal incident on the second side of the sealing section, where the second acoustic signal includes at least a portion of the first acoustic signal that has passed from the first side to the second side of the sealing section where the first acoustic signal is compared to the second acoustic signal to determine if

the sealing section is sealed.

[0010] Further areas of applicability of exemplary embodiments of the present invention will become apparent from the detailed description provided hereinafter. It should be understood that the detailed description and specific examples, while indicating exemplary embodiments of the invention, are intended for purposes of illustration only and are not intended to limit the scope of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0012] FIG. 1 is a diagram of an earpiece inserted in an ear canal in accordance with an exemplary embodiment;

[0013] FIG. 2 is a block diagram of optional components of an earpiece in accordance with an exemplary embodiment;

[0014] FIG. 3 is a flowchart of a method for checking an ear seal in accordance with an exemplary embodiment;

[0015] FIG. 4 is a flowchart to determine acoustic seal integrity of an earpiece in accordance with the exemplary embodiment.

[0016] FIG. 5 is a flowchart of a method to estimate the instantaneous cross-correlation between a first and second audio signal.

[0017] FIG. 6 is a flowchart to determine when to emit a test signal in accordance with an exemplary embodiment;

[0018] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention;

[0019] FIG. 8 is a block diagram for a method of adjusting the IMS system in accordance with at least one exemplary embodiment;

[0020] FIG. 9 is a block diagram for a method of adjusting IMS pressure in accordance with at least one exemplary embodiment;

[0021] FIG. 10 illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with

at least one exemplary embodiment;

[0022] FIG. 11 illustrates a flowchart of an exemplary method to determine a test signal fundamental;

[0023] FIG. 12 illustrates a flowchart of an exemplary embodiment to determine tonal presence in audio content;

[0024] FIG. 13 illustrates a flowchart of a method to determine when to emit the test signal;

[0025] FIG. 14 illustrates a flowchart of an exemplary method to determine acoustic seal integrity;

[0026] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment; and

[0027] FIG. 16 illustrates the sending of a signal to an inflation controller upon detection of a seal fail to modify the seal pressure in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION

[0028] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0029] Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

[0030] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0031] Additionally exemplary embodiments are not limited to earpieces, for example some functionality can be implemented on other systems with speakers and/or microphones for example computer systems, PDAs, BlackBerry® smart phones, cell and mobile phones, and any other device that emits or measures acoustic energy. Additionally, exemplary embodiments can be used with digital and non-digital acoustic systems. Additionally various receivers and microphones can be used, for example MEMs transducers, diaphragm transducers, for example Knowles' FG and EG series transducers.

[0032] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0033] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0034] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for sealing or partially sealing an ear. FIG. 1 is a diagram of an earpiece inserted in an ear canal 124 in accordance with at least one exemplary embodiment of the invention. FIG. 1 also illustrates portions of the ear including pinna 128, ear canal 124 and eardrum 126. As illustrated, the earpiece comprises an electronic housing unit 100 and a sealing unit 108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 124 of a user 130. The earpiece is an in-ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude the ear canal 124.

[0036] The earpiece includes an Ambient Sound Microphone (ASM) 120 to capture ambient sound, an Ear Canal Receiver (ECR) 114 to deliver audio to an ear canal124, and an Ear Canal Microphone (ECM) 106 to capture and assess a sound exposure level within the ear canal 124. The earpiece can partially or fully occlude the ear canal 124 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 124, and to form an acoustic seal with the walls of the ear canal 124 at a location between the entrance to the ear canal 124 and the tympanic membrane (or ear drum) 126. In general, such a seal is typically achieved by means of a soft and compliant housing of the sealing unit 108. Additionally, the sealing unit 108 can be a pressurized expandable element that fills a portion of the available local space.

[0037] Sealing unit 108 is an acoustic barrier having a first side corresponding to ear canal124 and a second side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 108 includes an ear canal microphone tube 110 and an ear canal receiver tube 112. Sealing unit 108 creates a closed cavity of approximately 5 cc or less between the first side of sealing unit 108 and the tympanic membrane 126 in ear canal124. In at least one exemplary embodiment the sealing facilitates using the ECR

(speaker) 114 to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 126 resulting from the sound field at the entrance to the ear canal 124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0038] In at least one exemplary embodiment and in broader context, the second side of sealing unit 108 corresponds to the side adjacent to electronic housing unit 100. Ambient sound microphone 120 is housed in electronic housing unit 100 and is exposed to the ambient environment for receiving sound from the ambient environment around the user.

[0039] The electronic housing unit 100 can include various system components such as a microprocessor 116, memory 104, battery 102, ECM 106, ASM 120, ECR, 114, and user interface 122, or these components can reside in a separate system or interface operatively connected. Microprocessor 116 (or processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 116 is operatively coupled to memory 104, ECM 106, ASM 120, ECR 114, and user interface 122. An optional wire 118 can provide an external connection to the earpiece. Battery 102 powers the circuits and transducers of the earpiece. Battery 102 can be a rechargeable or replaceable battery.

[0040] In at least one exemplary embodiment, electronic housing unit 100 is adjacent to sealing unit 108. Openings in electronic housing unit 100 receive ECM tube 110 and ECR tube 112 to respectively couple to ECM 106 and ECR 114. ECR tube 112 and ECM tube 110 acoustically couple signals to and from ear canal 124. For example, ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124 where it is received by the tympanic membrane 126 of the user of the earpiece. Conversely, ECM 106 receives an acoustic signal present in ear canal124 though ECM tube 110.

[0041] One function of ECM 106 is that of measuring the sound pressure level in the ear canal cavity 124 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece. In one arrangement, ASM 120 is used to monitor sound pressure at the entrance to the occluded or partially occluded ear canal 124. All transducers shown can receive or transmit audio signals to a processor 116 that undertakes audio signal processing and provides a transceiver for audio via the wired (wire 118) or a wireless communication path. Note also that the acoustic signals can be stored for later retrieval.

[0042] In at least one exemplary embodiment the earpiece can be constructed to actively monitor a sound

pressure level both inside and outside an ear canal 124. In at least one exemplary embodiment monitored data can be used to enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. In at least one exemplary embodiment an earpiece can facilitate at least one of conducting listening tests, filtering sounds in the environment, monitoring warning sounds in the environment, presenting notification based on identified warning sounds, maintaining constant audio content to ambient sound levels, and filtering sound in accordance with a Personalized Hearing Level (PHL).

[0043] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 124 using ECR 114 and ECM 106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 120. For instance, the ECR 114 can deliver an impulse within the ear canal124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. In at least one exemplary embodiment the earpiece can use either the ASM 120 or the ECM 106 to monitor the sound pressure level, which can then be used in a Sound Pressure Level Dosimeter calculation, to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0044] Referring to FIG. 2, a block diagram of an earpiece 201 in accordance with an exemplary embodiment is shown. A power supply 205 (e.g., USB power connection, hearing aid battery (batteries)) powers components of the earpiece 201 including microprocessor 206 (or processor 206, e.g., Texas Instruments TMS320C6713) and a data communication system 216 (e.g., RF or Bluetooth communication chip). As illustrated, the earpiece 201 includes the processor 206 operatively coupled to data communication system 216, ASM 210, ECR 212, and ECM 208. Data communication system 216 may include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 206. The processor 206 can also include a clock to record a time stamp.

[0045] In general, data communication system 216 is a communication pathway to components of the earpiece 201 and components external to the earpiece 201. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system 216 is configured to communicate with ECM 208, ASM 210, visual display 218, and user control interface 214 of the earpiece 201. As shown, user control interface 214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 216 is capable of communication to devices exterior to the

earpiece 201 such as the user's mobile phone 234, a second earpiece 222, and a portable media player 228. Portable media player 228 can be controlled by a manual user control 230.

[0046] The user's mobile phone 234 includes a mobile phone communication system 224. A microprocessor 226 is operatively coupled to mobile phone communication system 224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 220 worn by another person to the user's mobile phone. Similarly, the user's mobile phone 234 can be connected to the data communication system 216 of the earpiece 201 as well as the second earpiece 222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 234 through their respective earpieces.

[0047] As illustrated, a data communication system 216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

[0048] The RAM 202 stores program instructions for execution on the processor 206 as well as captured audio processing data. For instance, memory RAM 202 and ROM 204 can be off-chip and external to the processor 206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. In at least one exemplary embodiment, the data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer is operatively connected with processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0049] Data communication system 216 includes an audio interface operatively coupled to the processor 206 and the VOX to receive audio content, for example from portable media player 228, a cell phone, or any other communication device, and deliver the audio content to the processor 206. The processor 206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 206 (or the VOX of data communication system 216) can lower a volume of the audio content responsive to detecting an event for transmitting the acute sound to the ear canal of the user. The processor 206 by way of the ECM 208 can also actively

monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 216.

[0050] The earpiece 201 and data communication system 216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading over-the-air to the earpiece 201. It should be noted also that next generation access technologies can also be used in exemplary embodiments.

[0051] Data communication system 216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 201.

[0052] The power supply 205 utilizes common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 201 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 205 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0053] Microprocessor 206 is operatively connected with an EarSeal Inflation Management System 232 to control the degree to which the sealing unit 108 is inflated or deflated. In one exemplary embodiment, sealing unit 108 comprises an expandable element (e.g., inflatable balloon mechanism), whereby a cavity can be filled with air or a liquid to change the degree of acoustic isolation between the internal ear canal space 124 and the ambient environment. Alternately, a passive system for sealing ear canal 124 is used such as a flexible rubber or a silicon sealing unit or a foam plug. In one exemplary embodiment, the passive system is a balloon mechanism that is filled with air or liquid. The balloon mechanism conforms to the shape and size of an ear canal and includes a restorative force module that applies a pressure to the balloon mechanism for sealing the ear canal cavity.

[0054] The earpiece is a single operational device or a family of devices configured in a master- slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece are reused in different form factors for the master and slave devices.

[0055] Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user (second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

[0056] The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

[0057] In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

[0058] The degree of sealing between the first and second acoustic volumes is determined in a step 310 and the process ends at step 312. The cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or "leaky") if the cross- correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly

or incorrectly.

[0059] Referring to FIG. 1, the earpiece is used as an example to illustrate a test sequence as disclosed in FIG. 3. Sealing unit 108 occludes an opening of ear canal 124 creating a first volume (ear canal124) and a second volume (the ambient environment). Sealing unit 108 has a first side exposed to ear canal 124 and a second side is exposed to or corresponds to the ambient environment external to the ear.

[0060] In at least one exemplary embodiment, processor 116 is configured to receive a test signal in memory 104. Processor 116 generates the test signal and provides the test signal to Ear Canal Receiver 114 (ECR 114). ECR 114 emits the test signal into Ear Canal Receiver Tube (ECR Tube 112). The test signal propagates through ECR tube 112 and into ear canal 124. Ear Canal Microphone tube 110 (ECM tube 110) is configured to receive an acoustic signal incident on the first side of sealing unit 108. The test signal in ear canal 124 propagates through ECM tube 110 and is received by Ear Canal Microphone 106 (ECM 106). ECM 106 is configured to measure the test signal in ear canal124 and provide the measured test signal to processor 116.

[0061] As shown, electronic housing unit 100 of the earpiece is adjacent to the second side of sealing unit 108. Electronic housing unit 100 is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit 108. Electronic housing unit 100 includes Ambient Sound Microphone 120 (ASM 120), which is configured to measure sounds in the ambient environment. Thus, ASM 120 receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit 108. ASM 120 provides the measured ambient signal to processor 116.

[0062] Ideally, sealing unit 108 is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit 108 and into the ambient environment. Conversely, sealing unit 108 if improperly sealed will pass some of the test signal. Processor 116 compares the test signal to the signal provided by ECM 106 corresponding to the acoustic signal in ear canal124. In particular, processor 116 undertakes the cross-correlation between emitted test signal and the ECM signal.

[0063] Referring to FIG. 4, a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0064] In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively--input) of the test signal data buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time- align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

[0065] A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively--input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10Hz greater than the test signal frequency.

[0066] A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5. The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

[0067] Referring to FIG. 5, a flowchart of an exemplary embodiment to determine the instantaneous cross-correlation between a first audio signal and a second audio signal is illustrated. The process begins at step 500. In at least one exemplary embodiment, the first audio signal is the test signal (i.e. a sine wave) and the second signal is the low-pass-filtered ECM signal.

[0068] The correlation between two signals x and y at time k using an exponential window is defined as:

[0069] $$p(k) = \frac{S_{xy}(k)}{\sqrt{S_{xx}(k)_{yy}(k)}} \quad (1)$$

[0070] Where

[0071] $$S_{xx}(k) = \sum_{l=0}^{\infty} ce^{-\eta^l} x_{k-l} y_{k-l}$$

$$c = 1 - e^{-\eta}$$

[0072] And S_{xx} and S_{yy} are defined similarly as in (2) (replacing y with x for S_{xx} etc.).

[0073] It can be shown (see Aarts et al, 2001) that (1) can be approximated with the recursion:

[0074] $$\hat{\rho}(k) = \hat{\rho}(k-1) + \gamma[\partial_k - \beta_k \hat{\rho}(k-1)] \quad (3)$$

$$\partial_k = 2x_k y_k$$

$$\beta_k = \alpha x_k^2 \alpha^{-1} y_k^2$$

[0075] Where

$$\alpha = \frac{y_{RMS}}{x_{RMS}}$$

$$\gamma = \frac{ce^{\eta}}{2x_{RMS}y_{RMS}} \quad (4)$$

[0077] The cross-correlation estimate using the above recursion is modified for block-wise processing rather than the sample-by-sample basis. This modification replaces the sample values (i.e. x(k) and y(k)) with values for the N-length block mean, i.e.

[0078]

$$x(k) = \frac{1}{N}\sum_{l=0}^{N-1} x(k-l)$$

[0079] Furthermore, the numerator for *y* is replaced with a small constant and so is *a* (replacing *a* with a constant effectively un-normalizes the correlation estimate). It is found that the modified un-normalized block-wise cross-correlation accurately estimates the cross-correlation compared with using the standard cross-correlation for two signals.

[0080] The modified block-wise fast cross-correlation algorithm, as summarized in FIG. 5, comprises the following steps:

1. A first signal buffer 502 is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated 504 (x_{RMS})
3. The mean level of the first buffer is calculated 506.

4. A second signal buffer 508 is accumulated. This signal buffer corresponds to the filtered ECM signal.

5. The RMS level of the second buffer is calculated 510 (y_{RMS}).

6. The mean level of the second buffer is calculated 512.

7. In step 514, γ (gamma) is approximated as:

$$\gamma = \frac{\Gamma}{2x_{RMS}y_{RMS}}$$

Where Γ is a small constant, e.g. 10E-3.

8. In step 516,∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.

9. In step 518, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.

10. In step 520, the new temporary estimate of the correlation newRho_temp is calculated as:

newRho_temp=(delta-beta*rho_old))

11. In step 522, the new estimate of the correlation newRho is updated by summing the previous estimate of the correlation with the product of gamma and the temporary estimate of the correlation newRho_temp.

12. In step 524, the "old" value of the correlation is set to the newest correlation estimate, ready for the next iteration of the update algorithm.

13. In step 526, the current correlation estimate between the emitted test signal and the received and filtered ECM signal is set as equal to the value of newRho.

[0081] Referring to FIG. 6, a flowchart of a method to determine when to emit the test signal is shown. The test signal is emitted when the test can be performed unobtrusively to the user and also provides an accurate test. In at least one exemplary embodiment, a test event to determine if an earpiece is sealed correctly is initiated via a timing methodology. In a first timing scenario, the test event occurs after a delay of a first predetermined time period when the RMS of the Audio Content (AC) is less than a RMS threshold. In a second timing scenario, the delay of the first predetermined time period is allowed to lapse without the test event occurring when the RMS of the audio content is greater than the RMS threshold. A second predetermined time period is started where the test event occurs when the RMS of the audio content is less than the RMS threshold. The test event is then initiated when the second predetermined time period is exceeded independent of the RMS of the audio content.

[0082] A test sequence is initiated in a step 602. The previous seal test event resets the first digital timer in a step 604. A time delay is generated by the loop comprising steps 606 and 608. The first digital timer is time incremented in the step 606. After each added time increment, the first digital timer is compared against a

digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step 608 if the first digital timer is less than the digital_timer_threshold1.

[0083] A second digital timer is reset in a step 610 when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step 612. Audio content (AC) from a signal buffer is retrieved in a step 614. The audio content can be filtered through a low pass filter in an optional step 616. The RMS of the audio content is calculated in a step 618. The calculated RMS of the audio content is compared against an RMS_threshold 622 in a step 620. The second digital timer is compared against a digital_timer_threshold2 in a step 624 if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of 624.

[0084] The audio content signal is mixed with the test signal when the RMS of the audio content is less than the RMS_threshold in a step 626. Also, the audio content signal is mixed with the test signal when the second digital timer is greater than digital_timer_threshold2 in the step 624. The modified audio signal (having the test signal mixed in) is emitted by the ECR in a step 628 for testing the sealing section of the earpiece. The first digital timer is then reset in the step 604 to begin a timing sequence for another sealing section test.

[0085] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention. The estimated un-normalized cross-correlation between the ECM signal and the test signal (i.e. sine wave) is shown for different sine wave frequencies from 30-80Hz. Three different curves are provided corresponding to a good fit (i.e. a tight optimal seal providing approximately 20-30 dB of acoustic attenuation), mid or partial seal (i.e. an ear-seal that could be characterized as "half in" providing approximately 10-15 dB of acoustic attenuation), and a poor seal (i.e. an ear-seal providing less than 10 dB of acoustic attenuation). At lower test frequencies, the change in correlation is more pronounced as the degree of ear seal fitting is changed from "good" to "mid" and "poor". From the data, the threshold used to determine whether the ear seal can be characterized as "good" is approximately -20 dB, (i.e. 0.85 of FIG. 10 which corresponds to the value for XCorr_threshold 414 in FIG. 4).

[0086] In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these

criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40Hz test tone, the cross-correlation for a "good" ear seal is -8 dB, and for a bad ear seal it is -68 dB (i.e. a 60 dB difference). At a test signal frequency of 80Hz, the cross-correlation for a "good" ear seal is -8 dB and for a bad ear seal it is -38 dB (i.e. a 30 dB difference). Thus, above 200Hz the cross- correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

[0087] Using the cross-correlation rather than a level differencing approach improves the accuracy and minimizes errors which occur due to user non-speech body noise, such as teeth chatter; sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0088] FIG. 8 is a flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) in accordance with an exemplary embodiment. The IMS is adjusted depending on the degree of acoustic sealing provided by an earpiece. The method begins at step 802. The acoustic sealing is measured as disclosed in FIG. 7 and the result provided in a step 804 to determine the cross-correlation (XCorr) between a test signal and corresponding ECM signal. In general, the higher the cross-correlation, the higher the degree of acoustic sealing. An exemplary graph showing the relationship between XCorr and acoustic sealing is given in FIG. 10. The degree of acoustic sealing is determined from known XCorr using a look-up (or "hash") table or using a formula (e.g. of a polynomial form) that maps the acoustic sealing to the known XCorr value. The ambient sound level is measured in a step 806. The ambient sound level corresponds to the noise level in proximity to the user. In general, a higher degree of attenuation is desired when the ambient sound levels are high. Conversely, at low ambient sound levels the attenuation level of the IMS may be less of an issue and comfort more of a factor. The IMS is adjusted in a step 808 to meet the attenuation needs. In general, inflating the IMS increases attenuation while deflating the IMS decreases attenuation. The method terminates at step 810.

[0089] Referring to FIG. 9, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) is shown. In general, the attenuation increases when the pressure in the IMS is raised thereby allowing a degree of control to make adjustments. For example, an adjustment is made to increase attenuation when the background noise level rises or a seal check produces a failed result. Adjustments are made until the seal check passes. The pressure level adjustments of the IMS will fall within a comfort range of a user (e.g., between 0.1 bar and 0.3 bar gauge pressure). Typically, the pressure level is set at a minimum level to achieve a predetermined attenuation level.

[0090] The method begins at step 902. The degree of acoustic sealing is determined from cross-correlation between the ECM signal and the generated test signal. The XCorr value is provided in a step 904. In step 906, the attenuation provided by the IMS is calculated (equation) or looked up (table) from data such as that shown in FIG. 10. In one exemplary embodiment, the desired attenuation value is dependant on the ambient sound level of the user. In another exemplary embodiment, the desired attenuation value is dependant on the ear-canal sound level of the user. In yet another exemplary embodiment, the desired attenuation value is dependant on the level of audio content (e.g. speech or music audio) reproduced with the earphone device. In all of the above examples, the desired attenuation value is determined by one or more of the embodiments in a step 907.

[0091] The difference between the degree of acoustic sealing determined in step 906 and the desired attenuation value determined in step 907 is calculated in step 908. The difference value in step 908 is used to determine the change in pressure of the IMS necessary to minimize the difference value in a step 910. In at least one exemplary embodiment, the difference value of the attenuation is converted into a corresponding pressure value change (e.g. in milli-Bars) using a similar look-up table or equation method as described previously. The pressure change in the IMS is then affected with step 912 to meet the desired attenuation level. For example if the desired attenuation is a decrease of 10 dB in sound across the earpiece in the ear canal, then a pressure of a variable volume inflatable system can have a gauge pressure of about 0.15 bar. If the desired attenuation is a decrease of 20 dB across the earpiece in the ear canal then the gauge pressure can be increased to about 0.25 bar, where an increased pressure is associated with an increase in attenuation. An experimental table for each earpiece can be generated in a standard devised experimental setup (e.g. impedance tunnel) and referred to when changes are needed. The method ends at step 914.

[0092] Referring to FIG. 11, a flowchart of an exemplary method to determine a test signal fundamental is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user or made pleasant such that the user is unaware that the test signal is being played. This allows unobtrusive (periodic or non- periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0093] In at least one exemplary embodiment, an audio content 1102 is provided. A step 1104 stores audio content 1102 in a data buffer. In this example, audio content 1102 is music played from a media player and received via a wired or wireless connection to at least one earpiece in the user's ear. An alternate example would be that audio content 1102 is a speech audio signal from a portable telephone device or the like.

[0094] A step 1106 determines if the buffer of audio content 1102 comprises a strong tonal signal

component. Mixing the test signal having a similar fundamental frequency as audio content 1102 will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

[0095] A step 1108 determines whether to update or generate the first fundamental tone for the test signal. The test signal is not updated or generated if buffered audio content 402 does not contain a strong tonal signal component. A return to step 1104 fills the buffer with the next audio content 402 for analysis.

[0096] A step 1110 analyzes the buffer of data of audio content 1102 when it has been determined that it contains a strong tonal signal component. Step 1110 determines the fundamental frequency of the tonal signal. The fundamental tone, often referred to as the fundamental and abbreviate f_0 is the lowest frequency in a harmonic series. The fundamental frequency (also called a natural frequency) of a periodic signal is the inverse of the pitch period length. The pitch period is the smallest repeating unit of a signal. The fundamental frequency of the tonal signal can be calculated using an autocorrelation analysis.

[0097] In one exemplary embodiment, a mathematical operation 1114 is performed where the frequency component of the test signal is limited to a frequency range below a lower minimum and upper maximum frequency range. Fund_ratio is calculated, which is defined as a ratio of the determined fundamental frequency (F_fund) of the tonal signal from step 1110 to an upper threshold value F_fund_threshold 1112, which in one exemplary embodiment, is a fixed constant equal to approximately 100Hz. In general, F_fund_threshold 1112 is chosen to be a low frequency value which is above the lowest (or -3 dB) frequency that a transducer can reproduce, but below a predetermined frequency. In a comparison step 1116, if the estimated F-fund is higher than the F_fund_threshold 1112 (ratio > 1), then F_fund is reduced by an integer multiple to be below F_fund_threshold 1112 corresponding to the mathematical operation of step 1118. Otherwise, the test signal fundamental is equal to F_fund as shown in step 1120. Although not shown, the calculated test signal fundamental is compared and determined to be greater than a predetermined threshold.

[0098] Referring to FIG. 1, in at least one exemplary embodiment, processor 116 is configured to receive or generate audio content. As mentioned previously, the audio content can from external devices such as a portable phone or a media player. Memory 104 can be used as a buffer for the audio content. Processor 116 is configured to receive the buffer of audio content from memory 104. The steps and calculations of the block diagram of FIG. 4 are then performed by processor 116. The result being one of the identification of a strong tonal signal component in the buffer of audio content and the test signal fundamental or loading the buffer with new audio content and starting the process again.

[0099] Referring to FIG. 12, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step 1106 of FIG. 11 that analyzes audio content stored in a buffer. The method begins at step 1202. A step 1204 gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step 1206 filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20Hz to 500Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step 1208 analyzes the audio signal where a strong tonal signal component is represented by peaks in the analysis results. A step 1210 generates Absolute(Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute(Acorr) can be the square of the results from the auto-correlation.

[0100] A crest_factor_Acorr 1218 is generated from the results by calculating an RMS value 1214 (or time-averaged peak value) and peak value 1216 (or time averaged peak value). In at least one exemplary embodiment, the crest_factor_Acorr 1218 is the ratio of the peak value to the RMS value of an absolute auto-correlation sequence of the audio signal.

[0101] A comparison step 1222 is then performed. A strong tonal presence is identified when crest_factor_Acorr 1218 is greater than a threshold Crest_factor_Acorr_threshold 1220. Identification of the strong tonal presence indicates the audio signal would facilitate masking of the test signal to determine sealing of the device (step 1226). The audio signal is not used in conjunction with the test signal if crest_factor_Acorr 1218 is less than Crest_factor_Acorr_threshold 1220 (step 1224). The process would begin again loading a next sequence of the audio signal into the buffer for review.

[0102] Referring to FIG. 1, as mentioned previously, audio content is stored in a buffer, for example memory 104. The audio content in the buffer is provided to processor 116. In at least one exemplary embodiment, processor 116, runs the analysis as described in the block diagram of FIG. 12 thereby determining if a strong tonal presence is found in the audio content in the buffer. New audio content is loaded into the buffer (memory 104) if a strong tonal presence is not found beginning the procedure again.

[0103] Referring to FIG. 13, a flowchart of a method to determine when to emit the test signal is shown. The method begins at step 1302. The test signal is emitted when the test can be performed unobtrusively to the user and also provide an accurate test. In a step 1304, an audio signal is retrieved from a buffer. In at least one exemplary embodiment, the audio signal is received from an ECM or an ASM. The audio signal is measured to determine when the sound level is low in the ear canal, the ambient environment, or both. In

general, the test signal is emitted when the sound level is low.

[0104] A filter step 1306 band pass filters the audio signal. In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150Hz which corresponds to a frequency range of the test signal. In a step 1308, the RMS of the audio signal is calculated. The audio signal is analyzed to detect when the energy within an audio frequency range is below a threshold RMS_threshold 1310. The RMS of the audio signal is the signal level in the volume being measured. A comparison step 1312 compares the measured RMS level of the filtered audio signal against RMS_threshold 1310. In a step 1314, a test signal is emitted when the measured RMS value is less than RMS_threshold 1310. No test signal is emitted when the RMS of the audio signal is greater than RMS_threshold 1310. The method ends at step 1316.

[0105] Referring to FIG. 14, a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the earpiece to ensure that the seal is within normal operating specifications.

[0106] The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

[0107] In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

[0108] A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and

the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity of the signals in the first and second volumes. In particular, the non- difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

[0109] In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

[0110] Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a second comparison step 1428, the Magnitude of Peak is compared against a Peak_threshold 1430. If the Magnitude of Peak is greater than the Peak_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

[0111] In at least one exemplary embodiment and referring briefly to FIG. 1, an earpiece is tested to determine if sealing unit 108 is sealed correctly to the ear canal of the user. Sealing unit 108 creates a first volume in ear canal 124 and a second volume outside the ear canal 124 in the ambient environment. A masking approach is used to perform seal testing unobtrusively to the user. The user is listening to music or speech (audio content) provided to ear canal124 from ECR 114.

[0112] The music or speech is buffered in memory 104 or memory in processor 116. Processor 116 analyzes the audio content in the buffer to identify a strong tonal content. A test signal can

be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor 116 also analyzes the measured signals from ECM 106 and ASM 120 to determine when to emit the test signal. Processor 116 monitors and compares the sound level in the ambient environment and ear canal124. Processor 116 will provide the generated test signal to ECR 114 during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor 116 will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

[0113] Processor 116 monitors the test conditions and then provides the test signal to ECR 114 when an accurate sealing test can be performed. ECR 114 outputs an acoustic test signal which may or may not have other audio content. ECM 106 and ASM 120 respectively measure acoustic signals in ear canal 124 and the ambient environment. Processor 116 is operatively coupled to ECM 106 and ASM 120. The measured signals are buffered in memory 104.

[0114] In an exemplary embodiment, a cross-correlation is used to measure the similarity between the signals in ear canal124 and the ambient environment. Processor 116 performs the cross-correlation calculations using the measured acoustic signals from ECM 106 and ASM 120. In particular, the cross-correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$XCorr(n,l) = \sum_{n=0}^{N} ASM(n)ECM(n-l), \quad (5)$$

Where: l=0, 1, 2, … N

[0115] Where ASM(n) is the nth sample of the ASM signal, and ECM(n-1) is the $(n-1)^{th}$ sample of the ECM signal. A peak of the absolute cross-correlation is estimated using a peak- picking function and also the lag time at which this peak occurs (i.e. the index I at which this occurs). Thus, the Lag-time of Peak and the Magnitude of Peak are known and respectively compared against a Target Lag Limit range and a Peak Threshold. The user of the earpiece is notified or warned that sealing unit 108 is improperly sealed by processor 116 if the measured Lag-time of Peak is within the Target Lag Limit range and the Magnitude of Peak is greater than the Peak Threshold.

[0116] Like Correlation and Cross-Correlation, a coherence function is also a measure of similarity between two signals. Coherence is another non-difference comparison approach that can be used for detecting acoustic seal integrity. Coherence is defined as:

$$\gamma^2{}_{xy} = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (6)$$

[0117] Where Gxy is the cross-spectrum of two signals (e.g. the ASM and ECM signals), and can be calculated by first computing the cross-correlation in equation (5), applying a window function, for example a Hanning window, and transforming to the frequency domain, for example via an FFT. Gxx or Gyy is the auto-power spectrum of either the ASM or ECM signals, and can be calculated by first computing the auto-correlation (using equation 5, but where the two input signals are both from either the ASM or ECM and transforming to the frequency domain. The coherence function gives a frequency-dependent vector between 0 and 1, where a high coherence at a particular frequency indicates a high degree of coherence at this frequency, and can therefore be used to analyze test signal frequencies in the ASM and ECM signals whereby a high coherence indicates the presence of the test signal in the ambient environment (indicating leakage past the acoustic barrier).

[0118] Other approaches such as frequency spectrum analysis and RMS levels can also be used to determine if the earpiece is sealed correctly. Using a non-difference comparison approach such as coherence or cross-correlation between the ASM and ECM signals to determine sealing is more reliable than taking the level difference of the ASM and ECM signals. Using the cross correlation rather than a level differencing approach improves the accuracy and minimizes errors which may occur due to user non-speech body noise, such as teeth chatter, sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0119] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., 1500) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and

has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. 15A and 15B shows a slider actuator 1500, with a moving slide 1510, having attached a pumping arm 1560, with a hole 1550 covering bump 1520. Upon receipt of a signal1505 the actuator can move 1570 such that the bump 1520 covers the hole 1550 in a bellows 1530 pump. The actuation compresses the miniature bellows 1530 pushing 1590 gas through the one-way valve 1540, upon the back stroke the bump 1520 uncovers the hole 1550 and air rushes back into the bellows 1530 for the next pump. For example if the stroke length is 2 mm and the pump arm 1560 contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system 1570 needs to be inflated more (e.g., more gas to increase sealing) then each stroke can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system 1580. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal 1505 sent, which can be specifically tailored depending upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation systems can be used and controlled by signal 1505.

[0120] FIG. 16 illustrates a block diagram of controlled sealing in response to a seal fail signal and/or a request for increased seal attenuation. For example, a command signal 1600 is received (e.g., a seal fail signal in which a default attenuation value is attached for example 15 dB, or a signal requesting an additional amount of attenuation) by a processor. The command signal specifies an additional amount of attenuation or that the seal has failed. An attenuation needed N is identified 1610. For example if the current attenuation is 5 dB loss at f=500 Hz, at a pressure of 0.1 bar, and a command signal is received requesting a 10 dB loss at 500 Hz, then an experimental table is queried to find the pressure needed P which is then subtracted from the current pressure to obtain an increase in pressure DP needed. The increase in pressure is converted into a volume of gas increase needed (e.g., again referring to experimental tables based upon the inflation system volume). The volume of gas increase needed can then be directly linked with the number of cycles of an actuator pump M, 1620. The number of requested cycles **M** can then be sent 1630 to the actuator control circuit to pump the designated number of cycles. The system can then retest the attenuation 1640 and if refinements are needed the process can start again.

[0121] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the

following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

[0122] Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

Electronic Patent Application Fee Transmittal

Application Number:	
Filing Date:	
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-16838277

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 FILING FEE (ELECTRONIC FILING)	4011	1	80	80
UTILITY SEARCH FEE	2111	1	350	350
UTILITY EXAMINATION FEE	2311	1	400	400
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**830**

Electronic Acknowledgement Receipt	
EFS ID:	41996490
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	23-FEB-2021
Filing Date:	
Time Stamp:	13:57:46
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Payment was successfully received in RAM	$830
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Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	WebADS.pdf	146161 a7e8e35eaea7a4b4967f4e3b0afb1136a2c6a451	no	8
Warnings:					
Information:					
2	Abstract	CONT-16838277-ABSTRACT.pdf	89807 43541fcfe5d0a4f8bd3782787896311c8f0eaa59	no	1
Warnings:					
Information:					
3	Claims	CONT-16838277-CLAIMS.pdf	101173 06536d969e6576942108bfc788572058e284ab9f	no	3
Warnings:					
Information:					
4	Drawings-other than black and white line drawings	CONT-16838277-FIGS.pdf	3310893 aa1d2110f2221990c48b590bbbf49ba5ff857dc9	no	15
Warnings:					
Information:					
5	Information Disclosure Statement (IDS) Form (SB08)	CONT-16838277-IDS.pdf	1035190 d461402f03e932e35f04553a2a837bff56ab2f00	no	6
Warnings:					
Information:					
6	Oath or Declaration filed	CONT-16838277-OATH.pdf	146637 0e9029c44af35f4d94410ba917c849704535c256	no	3
Warnings:					
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7	Specification	CONT-16838277-SPEC-FINAL.pdf	380845 58ec988a87fa4676f383002400b82745ac0cc7ff	no	25
Warnings:					
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8	Fee Worksheet (SB06)	fee-info.pdf	34798 58bbba9fb7538cf821aa2ef9e8474b277bb9df48	no	2
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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.

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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.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

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

Secrecy Order 37 CFR 5.2:

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

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

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

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Customer Number	150139		
Email Address	ip@akerman.com	Add Email	Remove Email

Application Information:

Title of the Invention	ACOUSTIC SEALING ANALYSIS SYSTEM		
Attorney Docket Number	CONT-16838277	**Small Entity Status Claimed**	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)	15	**Suggested Figure for Publication (if any)**	14

Filing By Reference:

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

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

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

Publication Information:

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

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

Representative Information:

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

Please Select One:	◉ Customer Number	○ US Patent Practitioner	○ Limited Recognition (37 CFR 11.9)
Customer Number	150139		

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
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Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
Registration Number					

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Domestic Benefit/National Stage Information:

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

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
	Continuation of	16838277	2020-04-02

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
16838277	Continuation of	16414136	2019-05-16	10701499	2020-06-30

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
16414136	Continuation of	15700511	2017-09-11	10299053	2019-05-21

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
15700511	Continuation of	14827332	2015-08-17	9781530	2017-10-03

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
14827332	Continuation of	14054015	2013-10-15	9113267	2015-08-18

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Prior Application Status	Patented			Remove	
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
14054015	Division of	12555864	2009-09-09	8600067	2013-12-03

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
12555864	Claims benefit of provisional	61098250	2008-09-19

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Foreign Priority Information:

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

Application Number	Country	Filing Date (YYYY-MM-DD)	Access Code (if applicable)
	US		

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

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

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Authorization or Opt-Out of Authorization to Permit Access:

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

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

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

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

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

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

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

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

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

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

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

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Applicant Information:

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

Applicant 1

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

Clear

◯ Assignee	◯ Legal Representative under 35 U.S.C. 117	◯ Joint Inventor
◉ Person to whom the inventor is obligated to assign.	◯ Person who shows sufficient proprietary interest	

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

Name of the Deceased or Legally Incapacitated Inventor:	

If the Applicant is an Organization check here. ☒

Organization Name	Staton Techiya LLC

Mailing Address Information For Applicant:

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Country[i]	US	Postal Code	33064
Phone Number		Fax Number	
Email Address	Daniel@techiya.co		

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

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Assignee Information including Non-Applicant Assignee Information:

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

Assignee 1

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

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

Organization Name	Staton Techiya LLC

Mailing Address Information For Assignee including Non-Applicant Assignee:

Address 1	2500 NE 35th Street		
Address 2			
City	Lighthouse Point	**State/Province**	FL
Country i	US	Postal Code	33064
Phone Number		Fax Number	
Email Address	Daniel@techiya.co		

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

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-16838277
	Application Number	
Title of Invention	ACOUSTIC SEALING ANALYSIS SYSTEM	

Signature:

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

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

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

Signature	/John P Keady/			Date (YYYY-MM-DD)	
First Name	John	Last Name	Keady	Registration Number	56389

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ABSTRACT

A device or a method using the device includes a balloon configured to seal a user's orifice, where the balloon is configured to produce an acoustic seal between a first side and a second side of the balloon in an ear canal. At least a second side of the balloon is fitted into the ear canal. Audio processing circuitry produces an audio signal for driving a speaker in the device and to measure sound level using output from the microphone in the device while the speaker is being driven by the audio signal. The device or method further includes control circuitry to evaluate a seal quality of the device. Other embodiments are disclosed.

CLAIMS

We claim:

1. An audio device configured to monitor the seal between a first side and a second side of the audio device comprising:

a microphone configured to measure sound from a first side of the audio device;

a speaker, wherein the speaker is configured to emit an audio signal from the first side of the audio device;

a sealing section, configured to seal the audio device between the first side and a second side of the audio device;

a memory that stores instructions; and

a processor, operatively connected to the microphone, where the processor is operatively connected to the speaker, where the processor is operatively connected to the memory, where the processor is configured to execute the instructions to perform operations, the operations comprising:

sending a playback signal to the speaker;

receiving a microphone signal from the microphone;

comparing the microphone signal to the playback signal to generate a seal quality between the first and second sides; and

generating a filter to the playback signal based upon the seal quality.

2. The audio device according to claim 1, where the operations further comprise:

applying the filter to the playback signal to generate a corrected playback signal.

3. The audio device according to claim 2, where the operations further comprise:

sending the corrected playback signal to the speaker.

4. The audio device according to claim 3, wherein the seal quality is determined at least in part by comparing frequency components below 300Hz of the microphone signal and the playback signal.

5. The audio device according to claim 4, wherein the seal quality is determined at least in part by comparing frequency components below 200Hz of the microphone signal and playback signal.

6. The audio device according to claim 2, wherein the sealing section is a foam eartip.

7. The audio device according to claim 2, wherein the microphone signal is buffered in the memory.

8. The audio device of claim 2, where the audio device is a headphone.

9. The audio device of claim 2, where the audio device is an earphone.

10. An audio device comprising:

an ambient sound microphone configured to measure sound from an ambient side of the audio device;

an ear canal microphone configured to measure sound from an ear canal side of the audio device;

a speaker, wherein the speaker is configured to emit an audio signal;

a sealing section, configured to seal the audio device between the ambient side and the ear canal side of the audio device;

a memory that stores instructions; and

a processor connected to the microphone, where the processor is connected to the speaker, where the processor is connected to the memory, where the processor is configured to execute the instructions to perform operations, the operations comprising:

monitoring the seal quality of the audio device between the ear canal side and the ambient side for a period of time.

11. The audio device according to claim 10, where the operations further comprise:

generating a playback correction to a playback signal.

12. The audio device according to claim 11, where the operations further comprise:

generating an adjusted playback signal using the playback correction.

13. The audio device according to claim 12, where the operations further comprise:

sending the adjusted playback signal to the speaker.

14. The audio device according to claim 10, wherein the seal quality is determined at least in part by comparing frequency components below 1000Hz of the microphone signal and the playback signal.

15. The audio device according to claim 14, wherein the microphone signal is buffered in the memory.

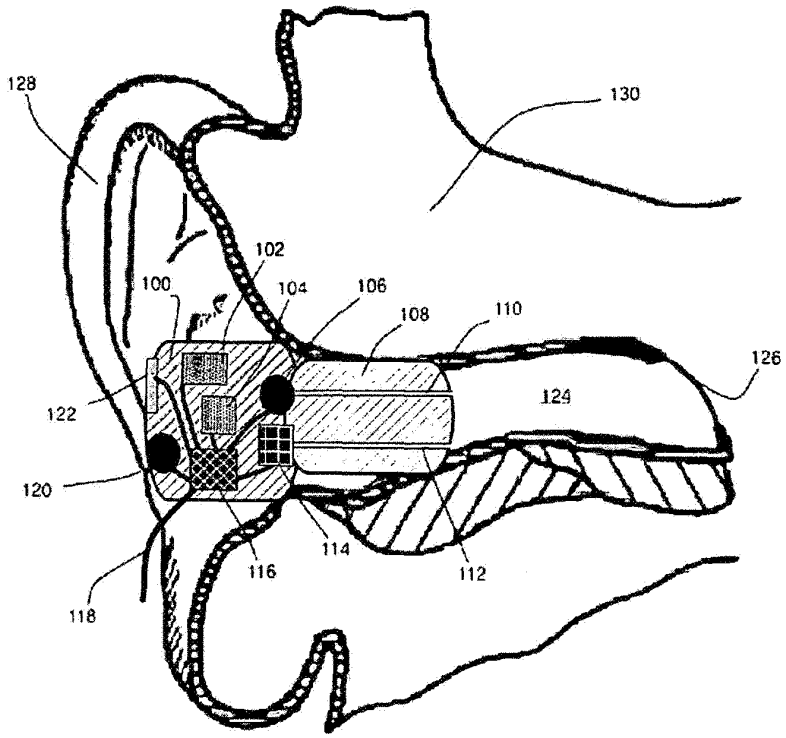
16. The audio device of claim 10, where the audio device is a headphone.

17. The audio device of claim 10, where the audio device is an earphone.

18. The earphone according to claim 17, where the sealing section is an eartip.

19. The earphone according to claim 16, where the sealing section is a foam cup.

FIG. 1



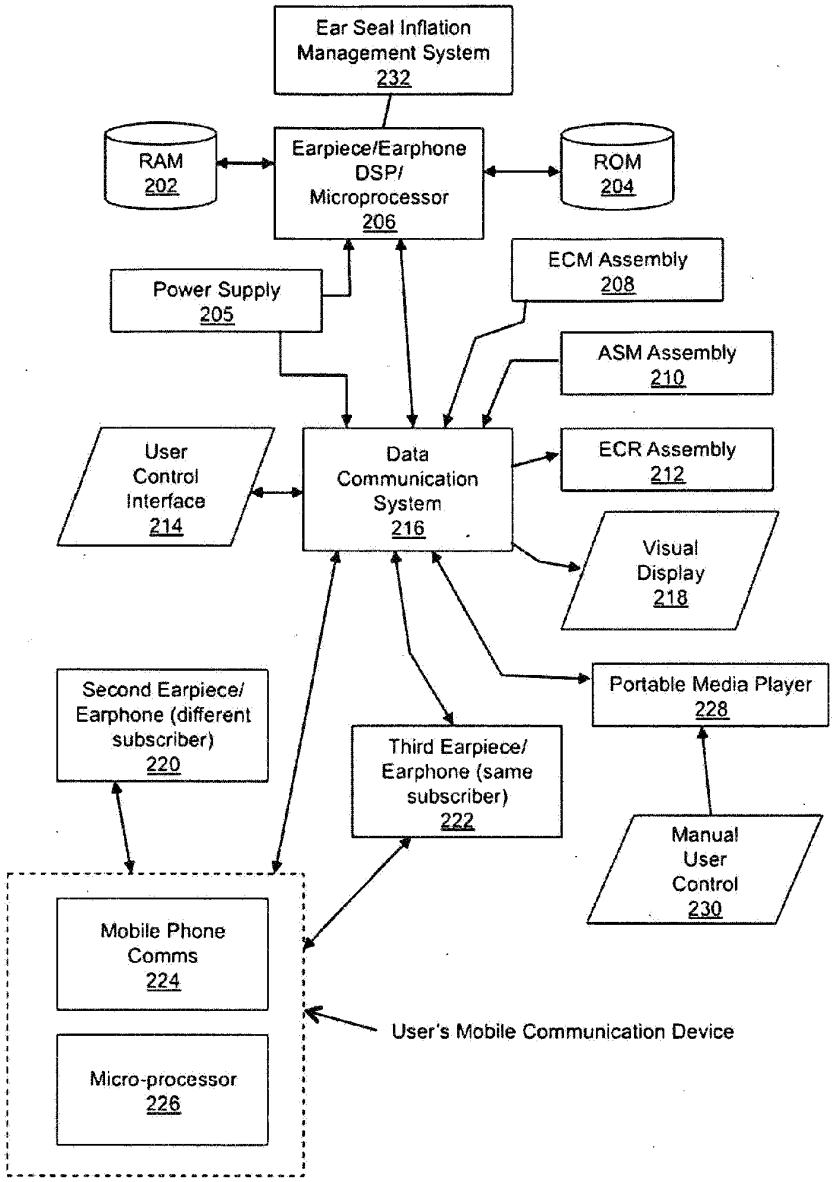

Ear Seal Inflation Management System 232
RAM 202
Earpiece/Earphone DSP/ Microprocessor 206
ROM 204
Power Supply 205
ECM Assembly 208
ASM Assembly 210
User Control Interface 214
Data Communication System 216
ECR Assembly 212
Visual Display 218
Second Earpiece/ Earphone (different subscriber) 220
Third Earpiece/ Earphone (same subscriber) 222
Portable Media Player 228
Manual User Control 230
Mobile Phone Comms 224
Micro-processor 226
User's Mobile Communication Device

FIG. 2

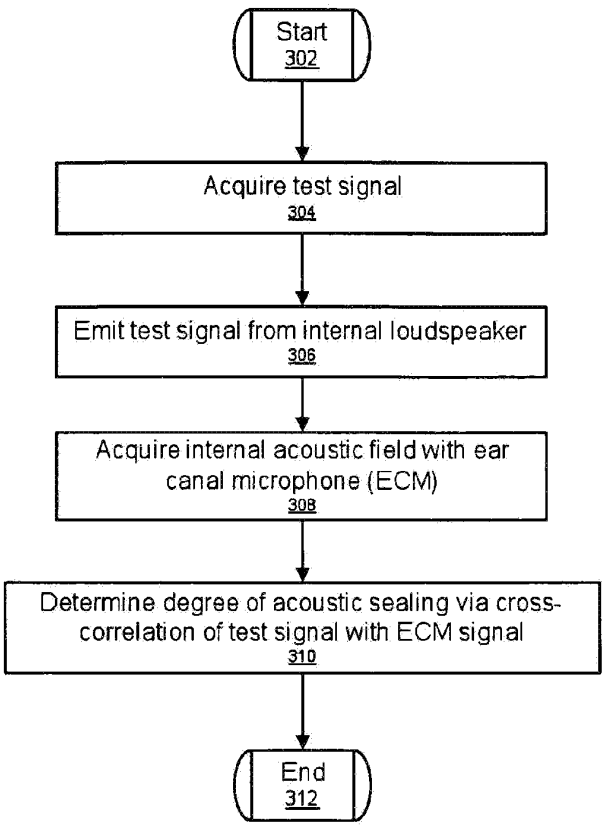

Start
302
Acquire test signal
304
Emit test signal from internal loudspeaker
306
Acquire internal acoustic field with ear canal microphone (ECM)
308
Determine degree of acoustic sealing via cross-correlation of test signal with ECM signal
310
End
312

FIG. 3

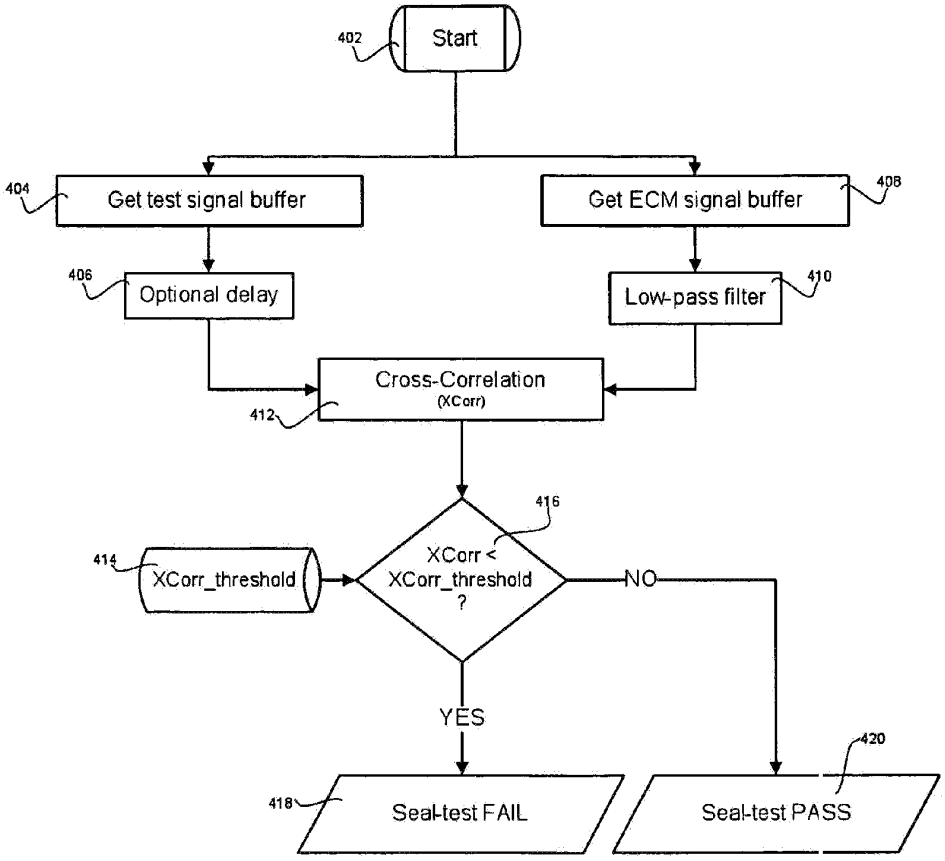

Start
402
Get test signal buffer
404
Get ECM signal buffer
408
Optional delay
406
Low-pass filter
410
Cross-Correlation
(XCorr)
412
XCorr_threshold
414
XCorr <
XCorr_threshold
?
416
NO
YES
Seal-test FAIL
418
Seal-test PASS
420

FIG. 4

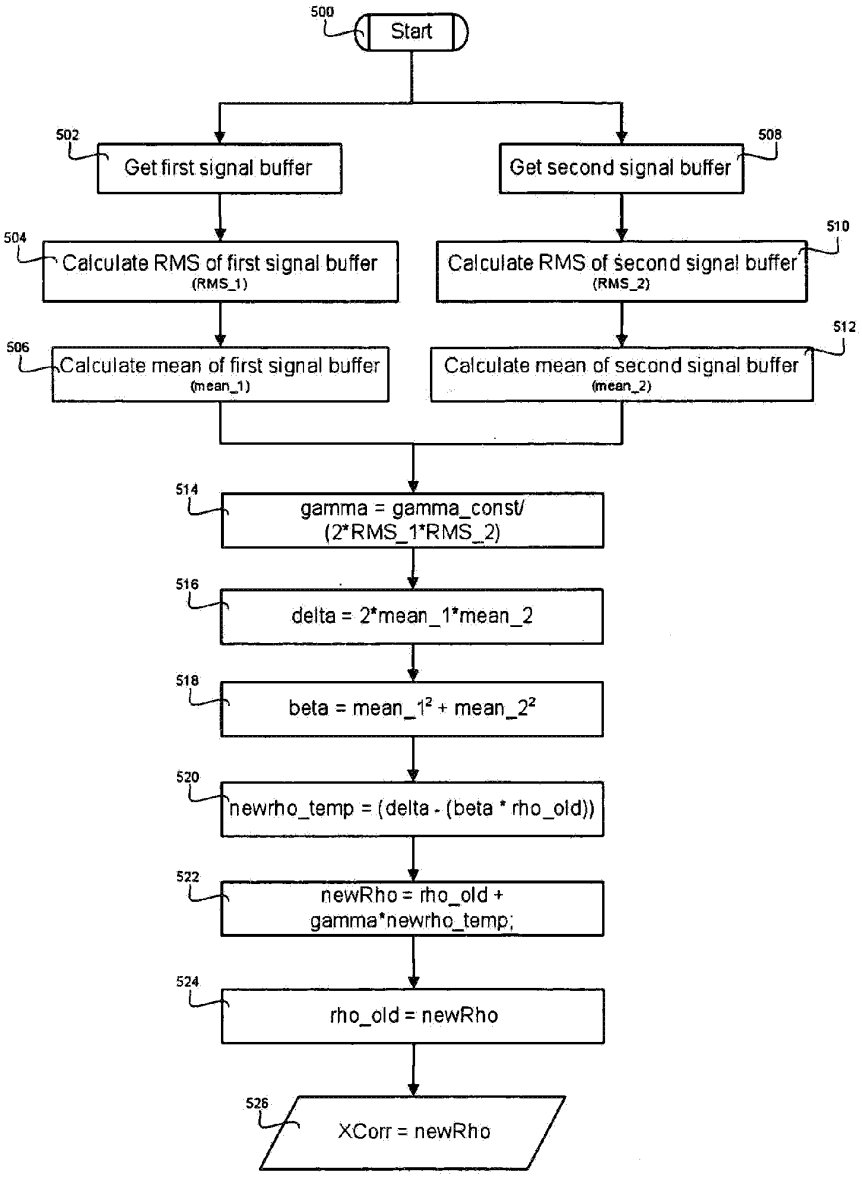

500
Start
502
Get first signal buffer
508
Get second signal buffer
504
Calculate RMS of first signal buffer
(RMS_1)
510
Calculate RMS of second signal buffer
(RMS_2)
506
Calculate mean of first signal buffer
(mean_1)
512
Calculate mean of second signal buffer
(mean_2)
514
gamma = gamma_const/
(2*RMS_1*RMS_2)
516
delta = 2*mean_1*mean_2
518
beta = mean_1^2 + mean_2^2
520
newrho_temp = (delta - (beta * rho_old))
522
newRho = rho_old +
gamma*newrho_temp;
524
rho_old = newRho
526
XCorr = newRho

FIG.5

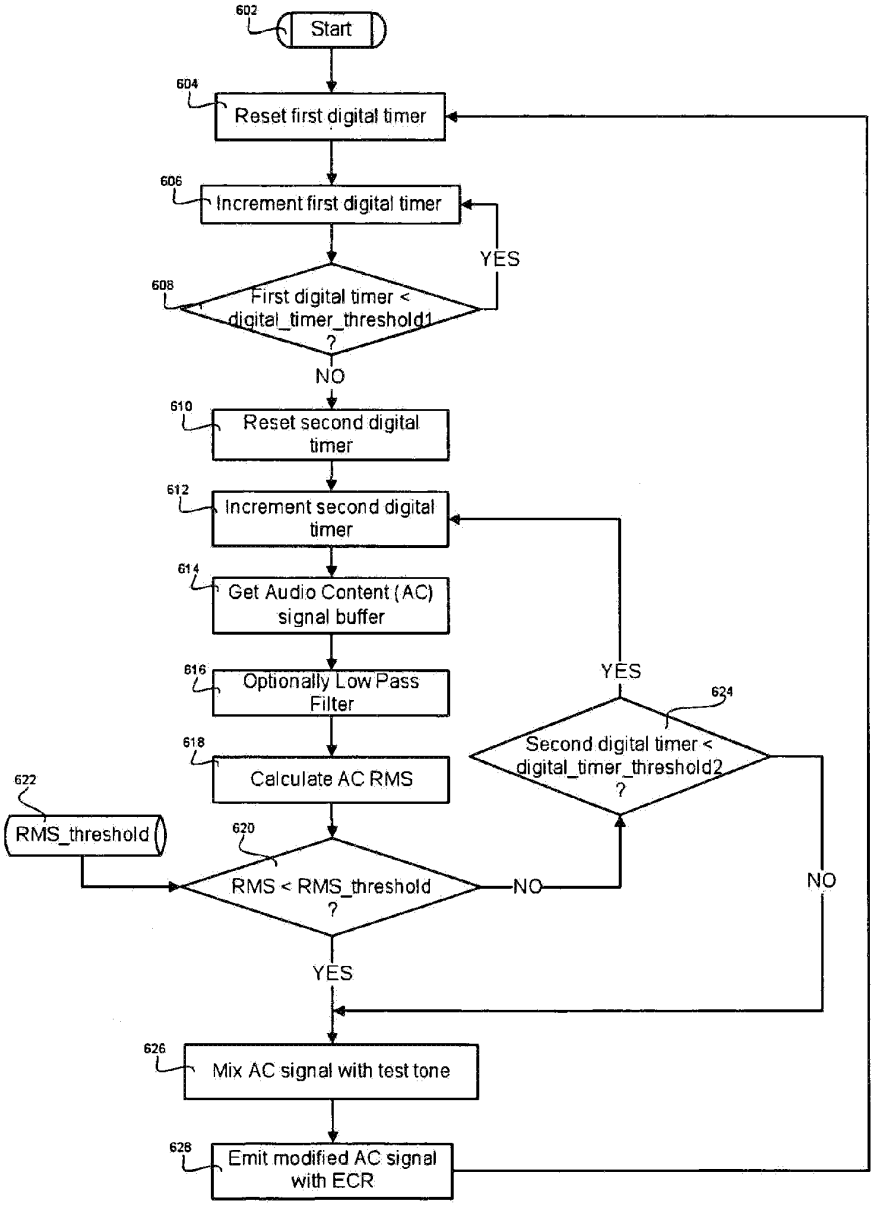

602
Start
604
Reset first digital timer
606
Increment first digital timer
YES
608
First digital timer < digital_timer_threshold1 ?
NO
610
Reset second digital timer
612
Increment second digital timer
614
Get Audio Content (AC) signal buffer
616
Optionally Low Pass Filter
618
Calculate AC RMS
622
RMS_threshold
620
RMS < RMS_threshold ?
NO
YES
624
Second digital timer < digital_timer_threshold2 ?
YES
NO
626
Mix AC signal with test tone
628
Emit modified AC signal with ECR

FIG. 6

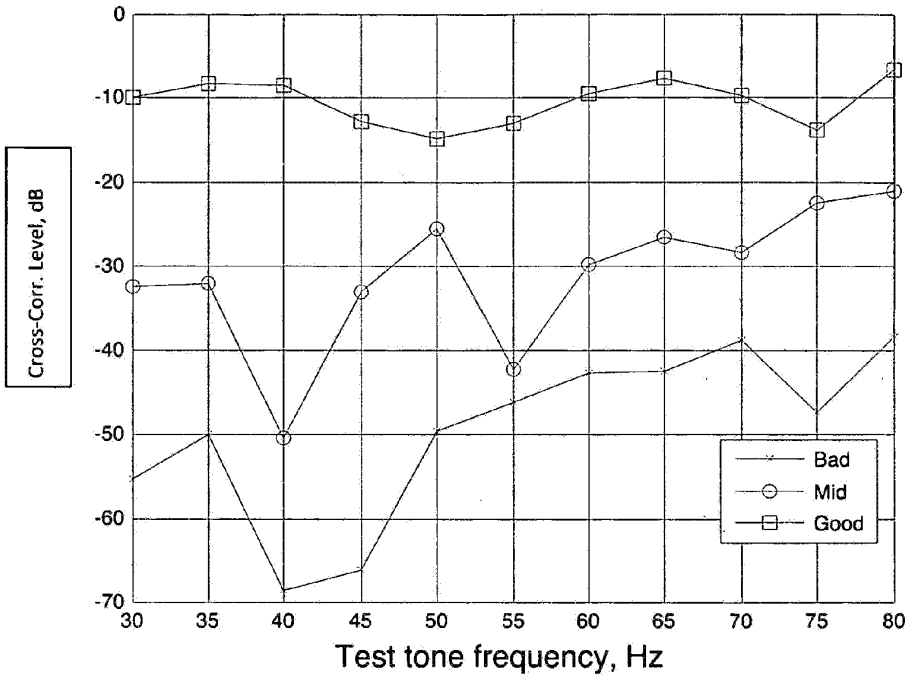

Cross-Corr. Level, dB
0
-10
-20
-30
-40
-50
-60
-70
30
35
40
45
50
55
60
65
70
75
80
Test tone frequency, Hz
Bad
Mid
Good

FIG. 7

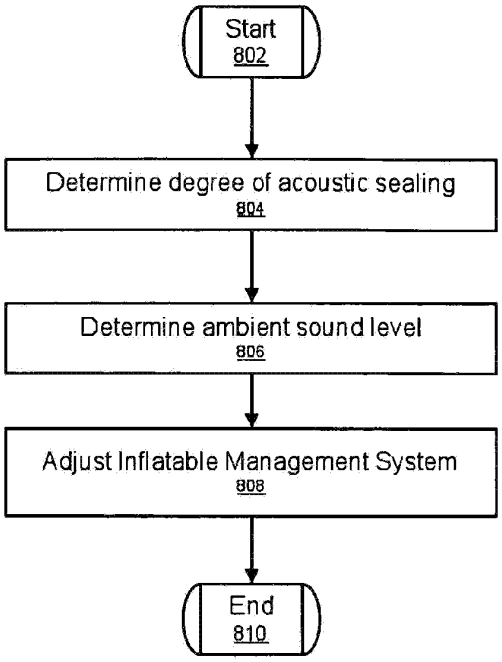

Start
802
Determine degree of acoustic sealing
804
Determine ambient sound level
806
Adjust Inflatable Management System
808
End
810

FIG. 8

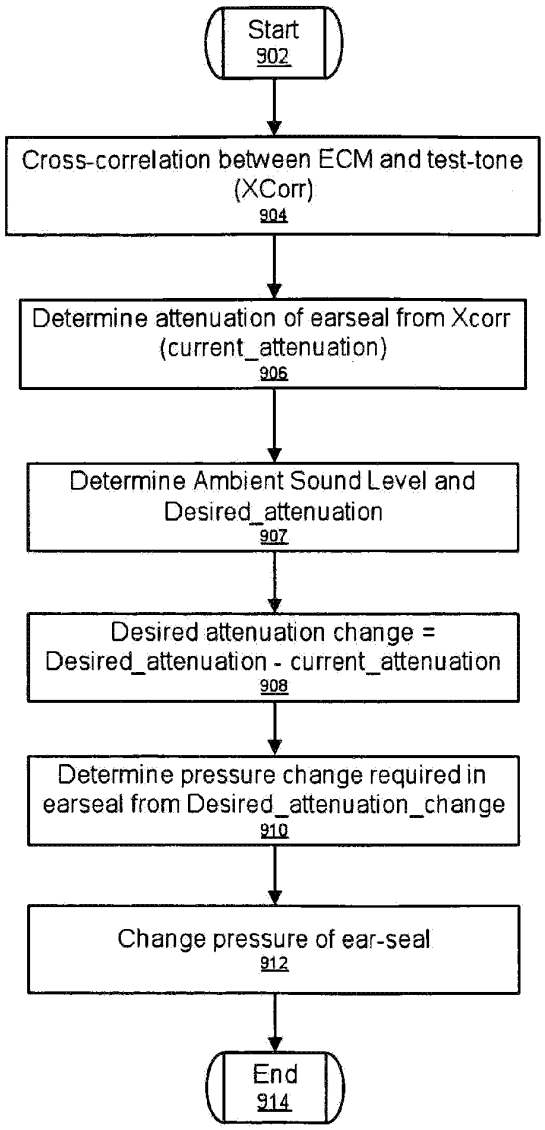

Start
902
Cross-correlation between ECM and test-tone (XCorr)
904
Determine attenuation of earseal from Xcorr (current_attenuation)
906
Determine Ambient Sound Level and Desired_attenuation
907
Desired attenuation change = Desired_attenuation - current_attenuation
908
Determine pressure change required in earseal from Desired_attenuation_change
910
Change pressure of ear-seal
912
End
914

FIG. 9

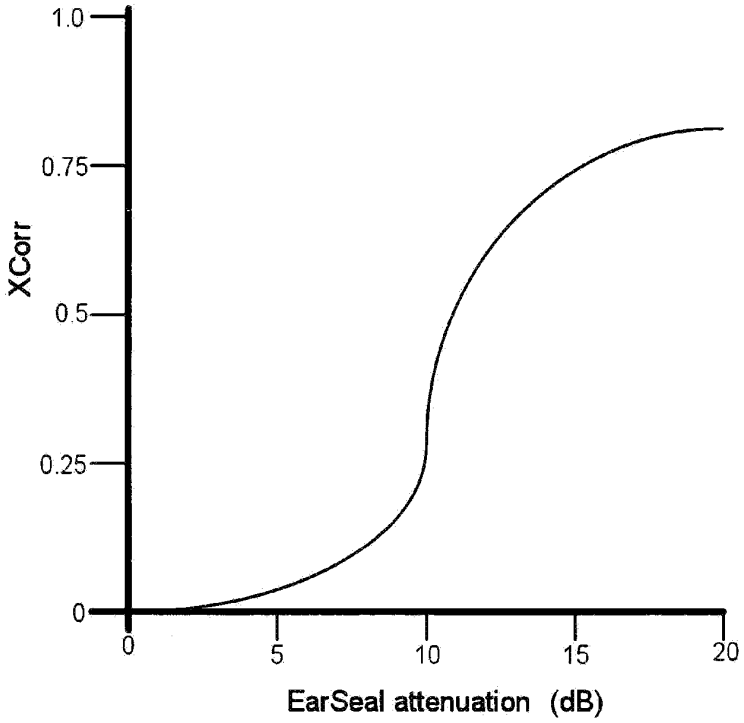


FIG. 10

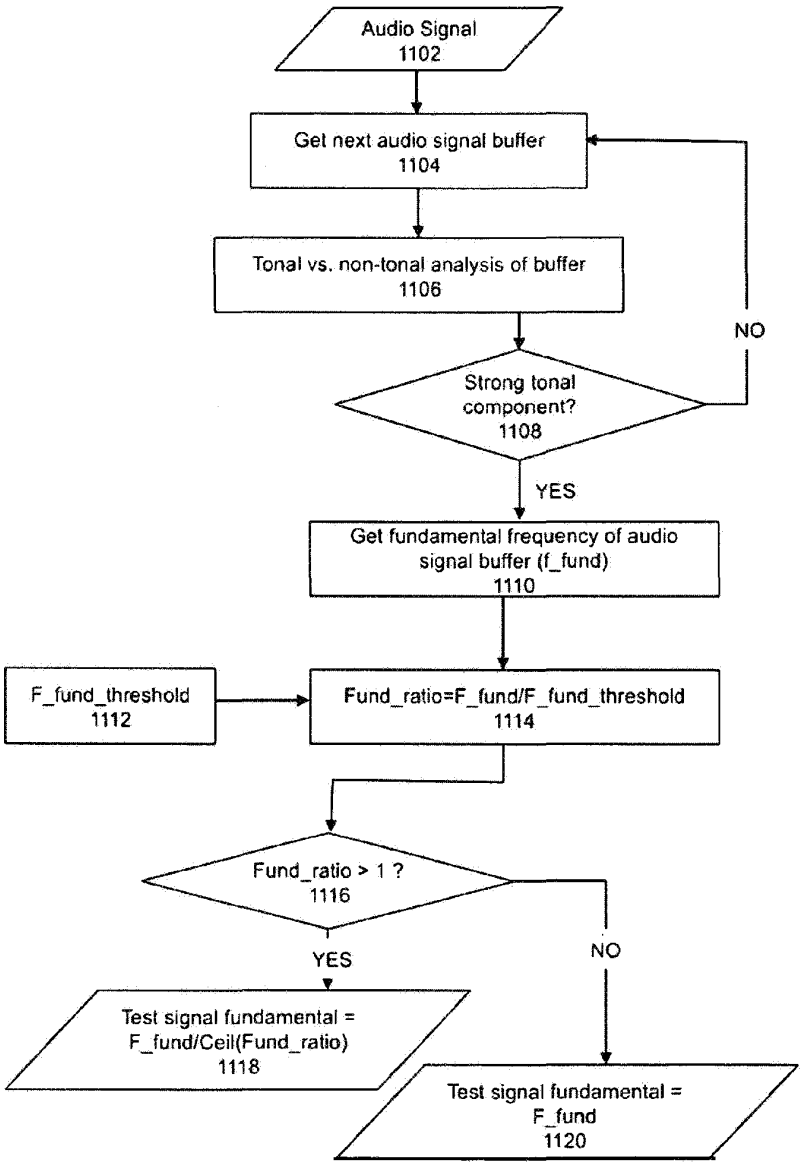

Audio Signal
1102
Get next audio signal buffer
1104
Tonal vs. non-tonal analysis of buffer
1106
Strong tonal component?
1108
NO
YES
Get fundamental frequency of audio signal buffer (f_fund)
1110
F_fund_threshold
1112
Fund_ratio=F_fund/F_fund_threshold
1114
Fund_ratio > 1 ?
1116
YES
NO
Test signal fundamental = F_fund/Ceil(Fund_ratio)
1118
Test signal fundamental = F_fund
1120

FIG. 11

Fig. 12

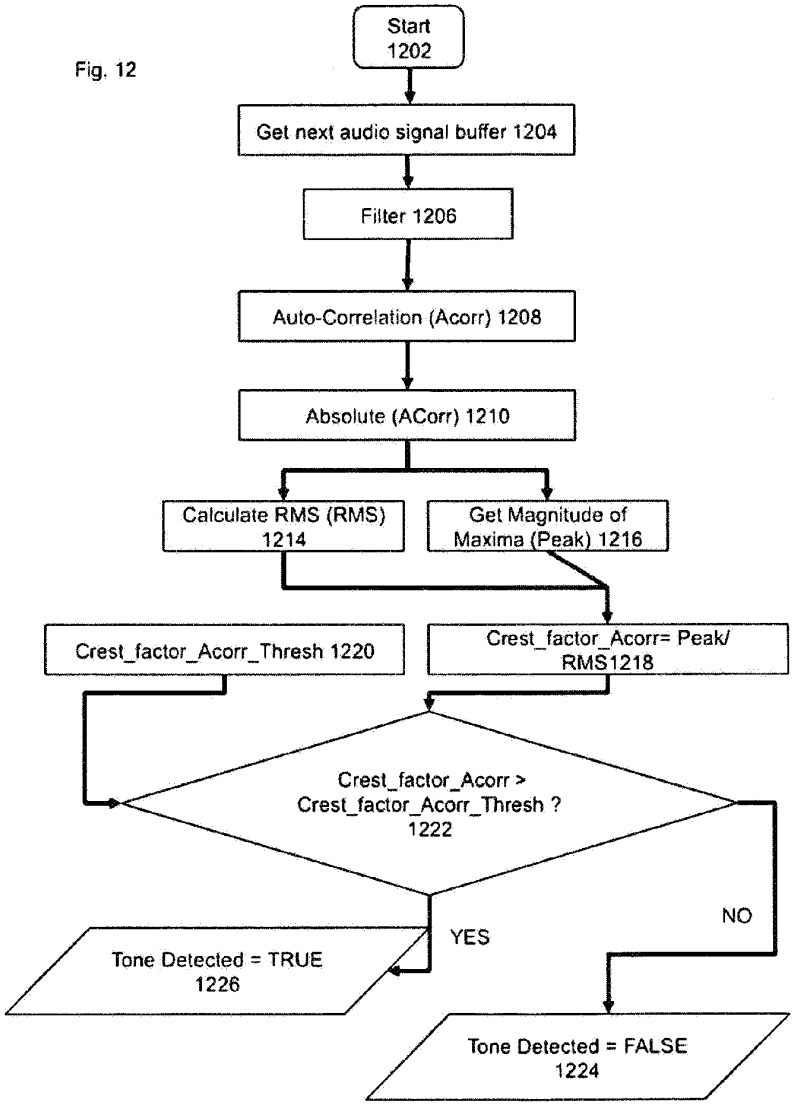

Start
1202
Get next audio signal buffer 1204
Filter 1206
Auto-Correlation (Acorr) 1208
Absolute (ACorr) 1210
Calculate RMS (RMS)
1214
Get Magnitude of
Maxima (Peak) 1216
Crest_factor_Acorr_Thresh 1220
Crest_factor_Acorr= Peak/
RMS1218
Crest_factor_Acorr >
Crest_factor_Acorr_Thresh ?
1222
YES
NO
Tone Detected = TRUE
1226
Tone Detected = FALSE
1224

Fig. 13

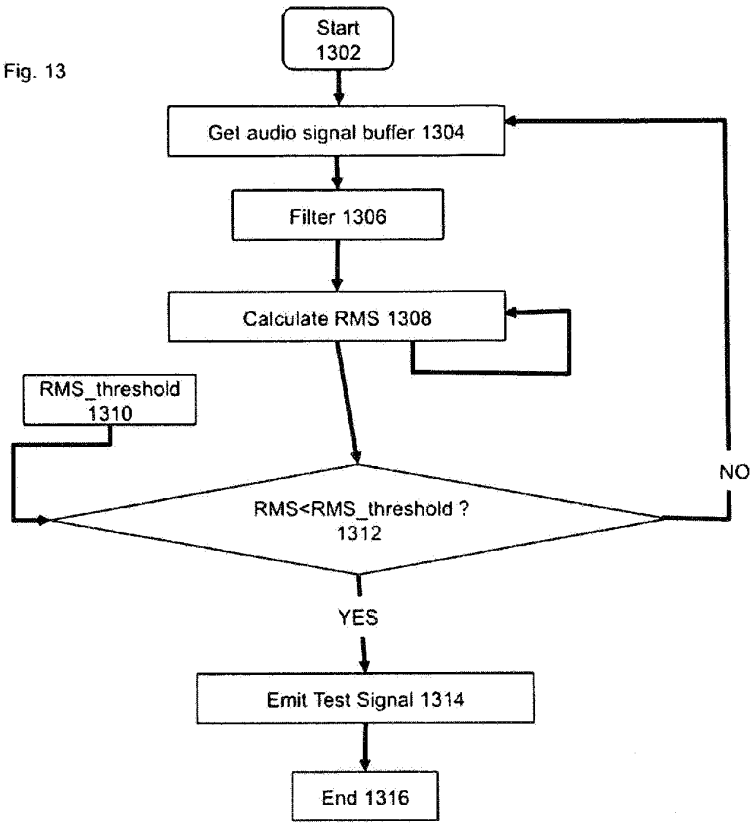

Start
1302
Get audio signal buffer 1304
Filter 1306
Calculate RMS 1308
RMS_threshold
1310
RMS<RMS_threshold ?
1312
NO
YES
Emit Test Signal 1314
End 1316

Fig. 14

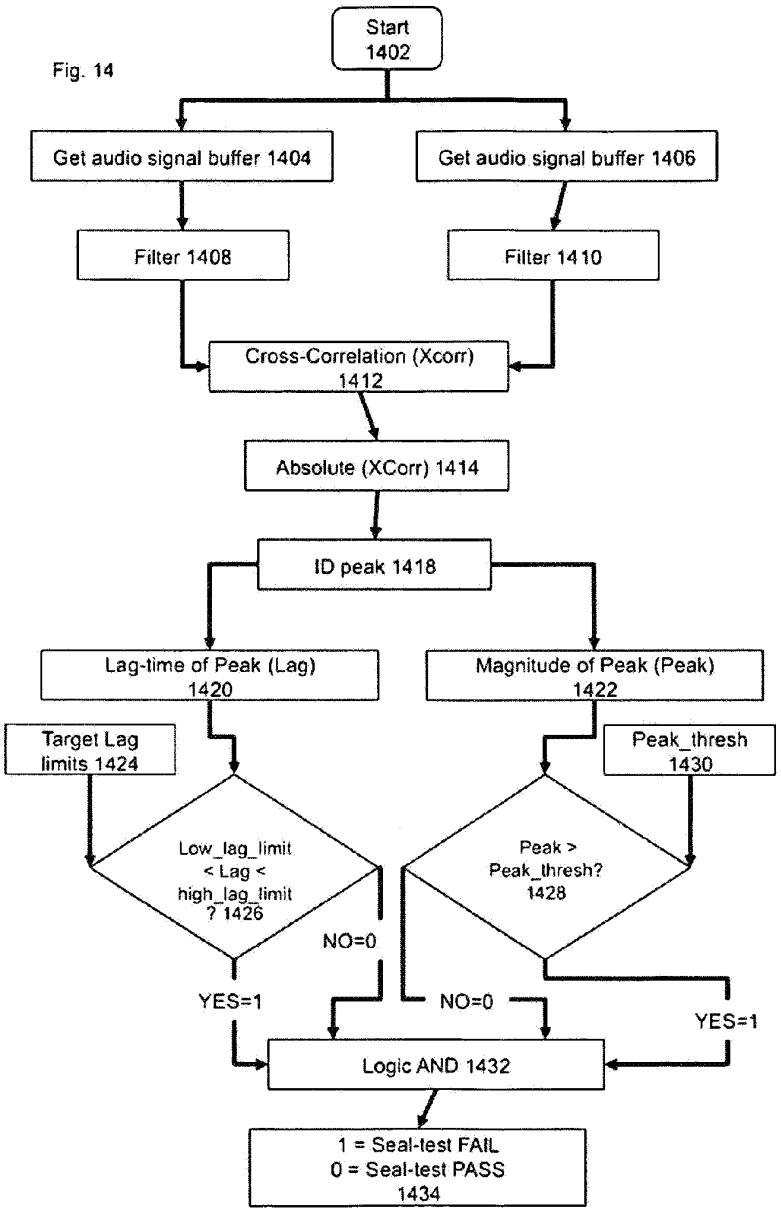

Start
1402
Get audio signal buffer 1404
Get audio signal buffer 1406
Filter 1408
Filter 1410
Cross-Correlation (Xcorr)
1412
Absolute (XCorr) 1414
ID peak 1418
Lag-time of Peak (Lag)
1420
Magnitude of Peak (Peak)
1422
Target Lag
limits 1424
Peak_thresh
1430
Low_lag_limit
< Lag <
high_lag_limit
? 1426
Peak >
Peak_thresh?
1428
NO=0
YES=1
NO=0
YES=1
Logic AND 1432
1 = Seal-test FAIL
0 = Seal-test PASS
1434

FIG. 15A

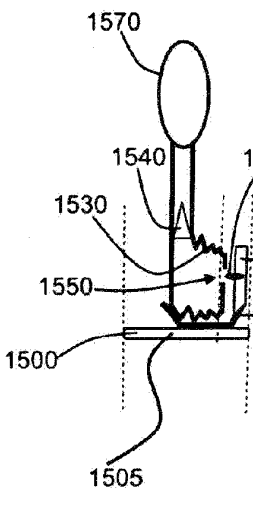


FIG. 15B

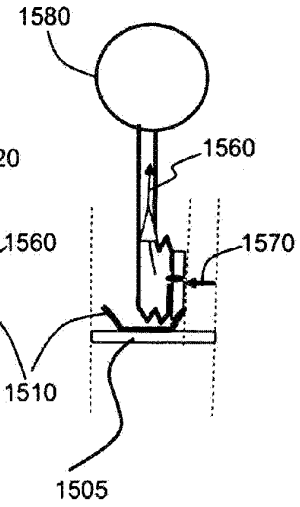
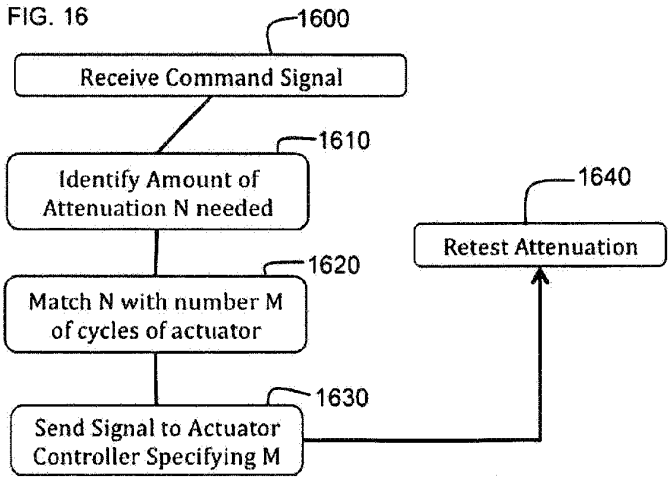


FIG. 16



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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
17/182,570	02/23/2021	2654	830	CONT-16838277	19	2

CONFIRMATION NO. 5671

FILING RECEIPT

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

OC000000123820376

Date Mailed: 03/05/2021

Receipt is acknowledged of this non-provisional utility patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF FIRST INVENTOR, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection.

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Inventor(s)

John Usher, Devon, UNITED KINGDOM;
John P. Keady, Fairfax Station, VA;

Applicant(s)

Staton Techiya LLC, Lighthouse Point, FL

Assignment For Published Patent Application

Staton Techiya LLC, Lighthouse Point, FL

Power of Attorney: None

Domestic Priority data as claimed by applicant

This application is a CON of 16/838,277 04/02/2020
which is a CON of 16/414,136 05/16/2019 PAT 10701499
which is a CON of 15/700,511 09/11/2017 PAT 10299053
which is a CON of 14/827,332 08/17/2015 PAT 9781530
which is a CON of 14/054,015 10/15/2013 PAT 9113267
which is a DIV of 12/555,864 09/09/2009 PAT 8600067
which claims benefit of 61/098,250 09/19/2008

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

Permission to Access Application via Priority Document Exchange: Yes

Permission to Access Search Results: Yes

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

If Required, Foreign Filing License Granted: 03/04/2021

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 17/182,570**

Projected Publication Date: 06/10/2021

Non-Publication Request: No

Early Publication Request: No

**** SMALL ENTITY ****

Title

ACOUSTIC SEALING ANALYSIS SYSTEM

Preliminary Class

381

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

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

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

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PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 17/182,570

APPLICATION AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	80		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	350		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	400		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	19 minus 20 =	*	x 50 =	0.00	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	2 minus 3 =	*	x 240 =	0.00			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	830		TOTAL	

APPLICATION AS AMENDED - PART II

AMENDMENT A

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

AMENDMENT B

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

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

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United States Patent and Trademark Office
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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
17/182,570	02/23/2021	John Usher	CONT-16838277

CONFIRMATION NO. 5671

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

PUBLICATION NOTICE

OC000000126222687

Title:ACOUSTIC SEALING ANALYSIS SYSTEM

Publication No.US-2021-0176576-A1
Publication Date:06/10/2021

NOTICE OF PUBLICATION OF APPLICATION

The above-identified application will be electronically published as a patent application publication pursuant to 37 CFR 1.211, et seq. The patent application publication number and publication date are set forth above.

The publication may be accessed through the USPTO's publically available Searchable Databases via the Internet at www.uspto.gov. The direct link to access the publication is currently http://www.uspto.gov/patft/.

The publication process established by the Office does not provide for mailing a copy of the publication to applicant. A copy of the publication may be obtained from the Office upon payment of the appropriate fee set forth in 37 CFR 1.19(a)(1). Orders for copies of patent application publications are handled by the USPTO's Public Records Division. The Public Records Division can be reached by telephone at (571) 272-3150 or (800) 972-6382, by facsimile at (571) 273-3250, by mail addressed to the United States Patent and Trademark Office, Public Records Division, Alexandria, VA 22313-1450 or via the Internet.

In addition, information on the status of the application, including the mailing date of Office actions and the dates of receipt of correspondence filed in the Office, may also be accessed via the Internet through the Patent Electronic Business Center at www.uspto.gov using the public side of the Patent Application Information and Retrieval (PAIR) system. The direct link to access this status information is currently https://portal.uspto.gov/pair/PublicPair. Prior to publication, such status information is confidential and may only be obtained by applicant using the private side of PAIR.

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

PTO/SB/08a (02-18)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17182570
Filing Date	2021-02-23
First Named Inventor	John Usher
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-16838277

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	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	

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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
	1	20030165246		2003-09-04	Kvaloy et al.	

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	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	2	20080037801		2008-02-14	Alves et al.	
	3	20060204014		2006-09-14	Isenberg et al.	
	4	20050078838		2005-04-14	Simon	

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Examiner Signature		Date Considered	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

[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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	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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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)).

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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.

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First Named Inventor	John Usher
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-16838277

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	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
	5	7562020		2009-06-14	Le et al.	
	6	9135797		2015-09-15	Couper et al.	
	7	9123343		2015-09-01	Kurki-Suonio	
	8	7756285		2010-07-13	Sjursen et al.	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	9	5903868		1999-05-11	Yuen et al.	
	10	3876843		1975-04-08	Moen	
	11	5208867		1993-05-04	Stites, III	
	12	6226389		2001-05-01	Lemelson et al.	
	13	8493204		2013-07-23	Wong et al.	
	14	6661901		2003-12-09	Svean et al.	
	15	7430299		2008-09-30	Armstrong et al.	
	16	6567524		2003-05-20	Svean et al.	
	17	7107109		2006-09-12	Nathan et al.	
	18	4947440		1990-08-07	Bateman et al.	
	19	9037458		2015-05-19	Park et al.	

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	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	20	4054749		1977-10-18	Suzuki et al.	
	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
	25	6021325		2000-02-01	Hall	
	26	8750295		2014-06-10	Liron	
	27	7920557		2011-04-05	Moote	

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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
	1	20030165246		2003-09-04	Kvaloy et al.	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	First Named Inventor	John Usher
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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	2	20050288057		2005-12-29	Lai et al.	
	3	20060195322		2006-08-31	Broussard et al.	
	4	20070043563		2007-02-22	Comerford et al.	
	5	20010046304		2001-11-29	Rast	
	6	20070189544		2007-08-16	Rosenberg	
	7	20060092043		2006-05-04	Lagassey	
	8	20020106091		2002-08-08	Furst et al.	
	9	20040109668		2004-06-10	Stuckman	
	10	20020118798		2002-08-29	Langhart et al.	
	11	20040042103		2004-03-04	Mayer	
	12	20030161097		2003-08-28	Le et al.	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	13	20160104452		2016-04-14	Guan et al.	
	14	20040203351		2004-10-14	Shearer et al.	
	15	20080165988		2008-07-10	Terlizzi et al.	
	16	20070291953		2007-12-20	Ngia et al.	
	17	20110096939		2011-04-28	Ichimura	
	18	20060067551		2006-03-30	Cartwright et al.	
	19	20090024234		2009-01-22	Archibald	
	20	20090010456		2009-01-08	Goldstein et al.	
	21	20070086600		2007-04-19	Boesen	
	22	20040190737		2004-09-30	Kuhnel et al.	
	23	20040196992		2004-10-07	Ryan	

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	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	24	20050123146		2005-06-09	Voix et al.	
	25	20040125965		2004-07-01	Alberth Jr., et al.	
	26	20060083395		2006-04-20	Allen et al.	
	27	20110264447		2011-10-27	Visser et al.	
	28	20100296668		2010-11-25	Lee et al.	
	29	20110293103		2011-12-01	Park et al.	
	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] 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	EP1519625	EP	A2		Victorian		
	2	2006037156	WO	A1		Mejia et al.		

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

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

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]
	1	Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan 9-12, 2005, p. 284-286.	
	2	Bernard Widrow, John R. Glover Jr., John M. McCool, John Kaunitz, Charles S. Williams, Robert H. Hearn, James R. Zeidler, Eugene Dong Jr, and Robert C. Goodlin, Adaptive Noise Cancelling: Principles and Applications, PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975	
	3	Mauro Dentino, John M. McCool, and Bernard Widrow, Adaptive Filtering in the Frequency Domain, PROCEEDINGS OF THE IEEE, VOL. 66, NO. 12, DECEMBER 1978	

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

EXAMINER SIGNATURE

Examiner Signature		Date Considered	

*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	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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

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

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

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

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 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.

Electronic Acknowledgement Receipt

EFS ID:	44707337
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	10-JAN-2022
Filing Date:	23-FEB-2021
Time Stamp:	15:07:09
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	17182570_transmittal_ltr.pdf	124895 75134c0da3f48d0fdbd74d6d8c89b51fc7b81e05	no	1

Warnings:

Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARWARE-IDS.pdf	1035152 9f1f7ba6626189771b374035f409adc6e1117470	no	6
Warnings:					
Information:					
3	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS.pdf	1036606 6155612effb85f771f9176b080ae9d42b1a8c11b	no	9
Warnings:					
Information:					
4	Non Patent Literature	ref2-pdf2.pdf	3762561 39646ce913fa97e4a1efb547fcce8a48fc8e6d08	no	10
Warnings:					
Information:					
5	Non Patent Literature	ref1.pdf	1115022 01c5b583e9d89f0e10dda2f41631f1a7c8c19dda	no	2
Warnings:					
Information:					
6	Non Patent Literature	ref3.pdf	2858776 2418fbe7a8c340eaa35ef3922c589767859789c3	no	3
Warnings:					
Information:					
			Total Files Size (in bytes):	9933012	

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

New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/182,570	CONFIRMATION NO.	5671
APPLICATION OF	:	John Usher		

TRANSMITTAL LETTER

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

Date: January 10, 2022

Sir:

Please find attached:

- **IDS:** The attached IDS, consisting of two parts, list references that were cited in several IPRs for related Patents with some of the same inventors in the Staton Techiya LLC Patent Portfolio. Applicant appreciates the Examiner for taking the time to consider the extensive list for the present application. Attached are also three non-patent references cited. Should the Examiner have any questions, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

/John P. Keady/

John P. Keady
Ph.D. (Physics), Ph.D.(Comp. Sci.)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
Tel: 571-606-1291

Electronic Acknowledgement Receipt

EFS ID:	44861100
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	28-JAN-2022
Filing Date:	23-FEB-2021
Time Stamp:	11:44:47
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Preliminary Amendment	17182570-PRELIMINARY-AMDT.pdf	175994 dd85e1e557280aef04c8188fa0516bd1dfc72b2c	no	5

Warnings:

Information:	
Total Files Size (in bytes):	175994

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

New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/182,570	CONFIRMATION NO. 5671
APPLICATION OF	:	John Usher	
DATE FILED	:	23 Feb. 2021	
EXAMINER	:	UNKNOWN	
TC/A.U.	:	OPAP	
DOCKET	:	CONT-16838277	
TITLE	:	ACOUSTIC SEALING ANALYSIS SYSTEM	

PRELIMINARY AMENDMENT

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

Date: January 28, 2022

Sir:

Please find herein a preliminary amendment, replacing the previously filed claims with new claims, for the above identified application. Please amend the above-identified patent application as follows:

- **Amendments to the Claims** begin on page 2 of this paper.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

Date: January 28, 2022

/John P. Keady/

John P. Keady
CTO and Assistant General Counsel IP
Ph.D (Physics), Ph.D(Comp. Sci), JD, MBA
Reg. No. 56,389
Va Bar No. 47027
571-606-1291

AMENDMENTS TO THE CLAIMS

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

Listing of Claims:

Claims 1-19 (cancelled)

20.(new) A wearable device that can test the seal quality of an earphone comprising:

a memory that stores instructions; and

a processor that executes the instructions to perform operations, the operations comprising:

generating a test signal, wherein the test signal includes frequency components between the range of 30Hz and 200Hz;

sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user;

receiving a first microphone signal from the first microphone;

calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;

comparing the value to a threshold to determine a seal quality between the first side and the second side;

generating a message to convey the seal quality; and

sending the message to the user.

21.(new) The wearable according to claim 1, wherein the message is an acoustic message.

22.(new) The wearable according to claim 1, wherein the message is a visual message sent to a display of the wearable device.

23.(new) The wearable according to claim 1, wherein the speaker plays the test signal from the first side.

24.(new) The wearable according to claim 23, wherein the first microphone measures sound from the first side.

25.(new) The wearable according to claim 23, wherein the first microphone measures sound from the second side.

26.(new) The wearable according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.

27.(new) The wearable according to claim 1, wherein the first signal is the test signal and the second signal is a modified microphone signal.

28.(new) The wearable according to claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal.

29.(new) The wearable according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.

30.(new) The wearable according to claim 27, where the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than 200Hz.

31.(new) The wearable according to claim 29, where the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than 200Hz.

32.(new) The wearable according to claim 1, wherein the earphone includes a second microphone.

33.(new) The wearable according to claim 32, wherein the operations further include:

receiving a second microphone signal from the second microphone.

34.(new) The wearable device according to claim 33, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.

35.(new) The wearable device according to claim 33, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.

36.(new) The wearable device according to claim 33, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal.

37.(new) The wearable device according to claim 1, wherein the seal quality is good if the value is at or above the threshold.

38.(new) The wearable according to claim 36, wherein the first microphone measures sound from the first side and the second microphone measures sound from the second side.

39.(new) The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination.

26.(new) The audio device of claim 25, wherein the eartip is a flexible polymer.

27.(new) The audio device according to claim 1, wherein the message is sent to a communication device connected to the audio device.

PTO/SB/06 (09-11)
Approved for use through 1/31/2014. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875	Application or Docket Number 17/182,570	Filing Date 02/23/2021	☐ To be Mailed

ENTITY: ☐ LARGE ☑ SMALL ☐ MICRO

APPLICATION AS FILED - PART I

	(Column 1)	(Column 2)			
FOR	NUMBER FILED	NUMBER EXTRA		RATE ($)	FEE ($)
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A		N/A	
☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A		N/A	
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	minus 20 =	*		x $50 =	
INDEPENDENT CLAIMS (37 CFR 1.16(h))	minus 3 =	*		x $240 =	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).				
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))					
* If the difference in column 1 is less than zero, enter "0" in column 2.				TOTAL	

APPLICATION AS AMENDED - PART II

		(Column 1)		(Column 2)	(Column 3)			
AMENDMENT	01/28/2022	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA		RATE ($)	ADDITIONAL FEE ($)
	Total (37 CFR 1.16(i))	* 22	Minus	** 20	= 2		x $50 =	100
	Independent (37 CFR 1.16(h))	* 1	Minus	*** 3	= 0		x $240 =	0
	☐ Application Size Fee (37 CFR 1.16(s))							
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							
							TOTAL ADD'L FEE	100

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

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

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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

UNITED STATES PATENT AND TRADEMARK OFFICE

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

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	02/23/2021	John Usher	CONT-16838277	5671

150139 7590 02/09/2022

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

EXAMINER
KIM, PAUL

ART UNIT	PAPER NUMBER
2654	

NOTIFICATION DATE	DELIVERY MODE
02/09/2022	ELECTRONIC

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

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

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

ip@akerman.com
jp.keady@techiya.co
personics_docketing@cardinal-ip.com

Notice of Non-Compliant Amendment (37 CFR 1.121)	Application No. 17/182,570	Applicant(s) Usher et al.
	Art Unit OED2	AIA (FITF) Status Yes

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

The amendment document filed on 01/28/2022 is considered non-compliant because it has failed to meet the requirements of 37 CFR 1.121 or 1.4. In order for the amendment document to be compliant, correction of the following item(s) is required.

THE FOLLOWING MARKED (X) ITEM(S) CAUSE THE AMENDMENT DOCUMENT TO BE NON-COMPLIANT:

- ☐ 1. Amendments to the specification:
  - ☐ A. Amended paragraph(s) do not include markings.
  - ☐ B. New paragraph(s) should not be underlined.
  - ☐ C. Other _____ .
- ☐ 2. Abstract:
  - ☐ A.Not presented on a separate sheet. 37 CFR 1.72.
  - ☐ B.Other _____ .
- ☐ 3. Amendments to the drawings:
  - ☐ A. The drawings are not properly identified in the top margin as "Replacement Sheet", "New Sheet", or "Annotated Sheet" as required by 37 CFR 1.121(d).
  - ☐ B. The practice of submitting proposed drawing correction has been eliminated. Replacement drawings showing amended figures, without markings, in compliance with 37 CFR 1.84 are required .
  - ☐ C. Other _____ .
- ☑ 4. Amendments to the claims:
  - ☐ A. A complete listing of all of the claims is not present.
  - ☐ B. The listing of claims does not include the text of all pending claims (including withdrawn claims)
  - ☐ C. Each claim has not been provided with the proper status identifier, and as such, the individual status of each claim cannot be identified. Note: the status of every claim must be indicated after its claim number by using one of the following status identifiers: (Original), (Currently amended), (Canceled), (Previously presented), (New), (Not entered), (Withdrawn) and (Withdrawn-currently amended).
  - ☑ D. The claims of this amendment paper have not been presented in ascending numerical order.
  - ☐ E. Other: _____ .
- ☐ 5. Other (e.g., the amendment is unsigned or not signed in accordance with 37 CFR 1.4): For further explanation of the amendment format required by 37 CFR 1.121, see MPEP § 714.

TIME PERIODS FOR FILING A REPLY TO THIS NOTICE:

1. Applicant is given **no new time period** if the non-compliant amendment is an after-final amendment or an amendment filed after allowance, or a drawing submission (only) If applicant wishes to resubmit the non-compliant after-final amendment with corrections, the **entire corrected amendment** must be resubmitted.
2. Applicant is given **two months** from the mail date of this notice to supply the correction, if the non-compliant amendment is one of the following: a preliminary amendment, a non-final amendment (including a submission for a request for continued examination (RCE) under 37 CFR 1.114), a supplemental amendment filed within a suspension period under 37 CFR 1.103(a) or (c), and an amendment filed in response to a Quayle action. If any of above boxes 1 to 4 are checked, the correction required is only the corrected section of the non-compliant amendment in compliance with 37 CFR 1.121.

Extensions of time are available under 37 CFR 1.136(a) only if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action.

Failure to timely respond to this notice will result in:

Abandonment of the application if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action; or

Non-entry of the amendment if the non-compliant amendment is a preliminary amendment or supplemental amendment.

Legal Instruments Examiner (LIE), if applicable /TINA M. BELL/ Telephone No: (571)272-1553

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
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www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	02/23/2021	John Usher	CONT-16838277	5671

150139 7590 02/11/2022

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

EXAMINER	
KIM, PAUL	
ART UNIT	**PAPER NUMBER**
2654	
NOTIFICATION DATE	**DELIVERY MODE**
02/11/2022	ELECTRONIC

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

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

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

ip@akerman.com
jp.keady@techiya.co
personics_docketing@cardinal-ip.com

NOTICE REQUIRING EXCESS CLAIMS FEES	Application No.	Applicant(s)
	17/182,570	Usher et al.
	Art Unit	AIA (FITF) Status
	OED2	Yes

The excess claim(s) filed on 28 January 2022 is/are not accompanied by the appropriate payment of excess claims fees set forth in 37 CFR 1.16(h)-(j) or 1.492(d)-(f). Excess claims fees are required for each claim in independent form in excess of three (§ 1.16(h)), each claim (whether dependent or independent) in excess of twenty (note that § 1.75(c) indicates how multiple dependent claims are considered for fee calculation purposes) (§ 1.16(i)), and each application that contains a multiple dependent claim (§ 1.16(j)).

Since the application is not under a final rejection, applicant is given a time period of **TWO (2) MONTHS** from the mailing date of this notice to submit either: (1) the fee payment of $ 100 , or (2) an amendment in compliance with 37 CFR 1.121 that cancels the excess claim(s), in order to avoid ABANDONMENT. Extensions of this time period may be granted under 37 CFR 1.136, unless the excess claim(s) was/were presented in a preliminary amendment.

☐ 1. The funds in Deposit Account No. _____ are insufficient to cover the entire fee due. The balance is due within the time period set forth in this notice. See note below regarding the appropriate service charge.

☐ 2. The Credit Card payment to cover the entire fee due to Account _____ (**Card type + last 4 digits ONLY**) was refused. The balance is due within the time period set forth in this notice. See note below regarding the appropriate service charge.

☑ 3. The amendment that includes the excess claim(s) has not been entered, since applicant has failed to remit (or authorize charge to a Deposit Account or Credit Card) the fee as indicated on the attached Patent Application Fee Determination Record (PTO/SB/06). Remittance or authorization is due within the time period set forth in this notice .

☐ 4. The fee submitted in this application is insufficient. A balance of $ _____ is due for presentation of excess claims (37 CFR 1.16(h)-(j) or 1.492(d)-(f)).

☑ 5. Other.

Explanation (*Provide specific details of the required correction in order to assist the applicant. Indicate whether a service charge has been added to the fee due*): Please submit $100 for two dependent claims over 20

THE AMOUNT OF THE FEE(S) DUE IS SUBJECT TO CHANGE, GENERALLY ON OCTOBER 1 OF EACH YEAR (37 CFR 1.16, 1.21 & 1.492). THE **AMOUNT OF THE FEE(S) DUE IS DETERMINED AS OF THE DATE A COMPLETE REPLY WITH THE APPROPRIATE FEE(S) IS RECEIVED BY THE OFFICE** (37 CFR 1.8 & 1.10). BECAUSE THE AMOUNT DUE IS SUBJECT TO CHANGE, IT IS RECOMMENDED THAT APPLICANT CHECK THE CURRENT FEE SCHEDULE WHICH IS AVAILABLE ON THE USPTOS WEBSITE AT: ***http://www.uspto.gov/web/offices/ac/qs/ope/fees.htm***

Service Charges: There is a $50 service charge for processing each payment refused (including a check returned unpaid) or charged back by a financial institution (37 CFR 1.21(m)). There is a $25.00 service charge for each month when the balance of a deposit account is below $1000 at the end of the month (37 CFR 1.21(b)(2)).

Technical Support Staff (TSS): /TINA M. BELL/ Phone Number: (571)272-1553

Note to TSS: Please do NOT use this notice if the application is under a final rejection.

Electronic Acknowledgement Receipt

EFS ID:	44986364
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	14-FEB-2022
Filing Date:	23-FEB-2021
Time Stamp:	12:08:23
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Preliminary Amendment	17182570-PRELIMINARY-AMDT-2nd.pdf	184668 85c79b4e3a9ea663867a09033cefeb4650ab0069	no	6

Warnings:

Information:	
Total Files Size (in bytes):	184668

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

New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/182,570	CONFIRMATION NO. 5671
APPLICATION OF	:	John Usher	
DATE FILED	:	23 Feb. 2021	
EXAMINER	:	UNKNOWN	
TC/A.U.	:	OPAP	
DOCKET	:	CONT-16838277	
TITLE	:	ACOUSTIC SEALING ANALYSIS SYSTEM	

PRELIMINARY AMENDMENT

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

Date: February 14, 2022

Sir: This preliminary amendment and response is filed in response to the notice of Additional Fees Due and Notice to the Applicant of 11 Feb. 2022, regarding an alleged non-compliant amendment filed 28 Jan. 2022. First note this preliminary amendment replaces the not entered preliminary amendment of 28 Jan. 2022. No fees are due since no additional independent claims nor claims over 20 have been added.

Please amend the above-identified patent application as follows:

- **Amendments to the Claims** begin on page 2 of this paper.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

Date: February 14, 2022

/John P. Keady/

John P. Keady
CTO and Assistant General Counsel IP
Ph.D (Physics), Ph.D(Comp. Sci), JD, MBA
Reg. No. 56,389
Va Bar No. 47027
571-606-1291

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 audio device configured to ~~monitor~~ test the seal ~~between a first side and a second side~~ of ~~the audio device~~ an earphone comprising:

~~a microphone configured to measure sound from a first side of the audio device;~~

~~a speaker, wherein the speaker is configured to emit an audio signal from the first side of the audio device;~~

~~a sealing section, configured to seal the audio device between the first side and a second side of the audio device;~~

a memory that stores instructions; and

a processor, ~~operatively connected to the microphone, where the processor is operatively connected to the speaker, where the processor is operatively connected to the memory,~~ wherein the processor is configured to execute the instructions to perform operations, the operations comprising:

~~sending a playback signal to the speaker;~~

~~receiving a microphone signal from the microphone;~~

~~comparing the microphone signal to the playback signal to generate a seal quality between the first and second sides; and~~

~~generating a filter to the playback signal based upon the seal quality~~

generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between the range of 30Hz and 200Hz;

sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user;

receiving a first microphone signal from the first microphone;

calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;

comparing the value to a threshold to determine a seal quality between the first side and the second side;

generating a message to convey the seal quality; and

sending the message to the user.

.

2.(currently amended) The audio device according to claim 1, wherein the message is an acoustic message. ~~the operations further comprise:~~

~~applying the filter to the playback signal to generate a corrected playback signal.~~

3(currently amended). The audio device according to claim [[2]]1, wherein ~~the operations further comprise:~~

~~sending the corrected playback signal to the speaker~~ the message is a visual message sent to a display of the audio device.

4.(currently amended) The audio device according to claim [[3]]1, wherein the speaker plays the test signal from the first side. ~~seal quality is determined at least in part by comparing frequency components below 300Hz of the microphone signal and the playback signal.~~

5.(currently amended) The audio device according to claim 4, wherein the first microphone measures sound from the first side. ~~seal quality is determined at least in part by comparing frequency components below 200Hz of the microphone signal and playback~~ signal.

6.(currently amended) The audio device according to claim [[2]]4, wherein the first microphone measures sound from the second side. ~~the sealing section is a foam eartip.~~

7.(currently amended) The audio device according to claim [[2]]1, wherein the ~~microphone signal is buffered in the memory~~ first signal is the test signal and the second signal is the first microphone signal .

8.(currently amended) The audio device of claim [[2]]1, wherein the first signal is the test signal and the second signal is a modified first microphone signal. ~~audio device is a headphone.~~

9.(currently amended) The audio device of claim [[2]]1, wherein the first signal is a modified test signal and the second signal is the first microphone signal. ~~audio device is an earphone.~~

10.(currently amended) ~~An~~ The audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal. ~~comprising:~~

~~an ambient sound microphone configured to measure sound from an ambient side of the audio device;~~

~~an ear canal microphone configured to measure sound from an ear canal side of the audio device;~~

~~a speaker, wherein the speaker is configured to emit an audio signal;~~

~~a sealing section, configured to seal the audio device between the ambient side and the ear canal side of the audio device;~~

~~a memory that stores instructions; and~~

~~a processor connected to the microphone, where the processor is connected to the speaker, where the processor is connected to the memory, where the processor is configured to execute the instructions to perform operations, the operations comprising:~~

~~monitoring the seal quality of the audio device between the ear canal side and the ambient side for a period of time.~~

11.(currently amended) The audio device according to claim [[10]]8, ~~where the operations further comprise:~~

~~generating a playback correction to a playback signal.~~ wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

12.(currently amended) The audio device according to claim [[11]]10, ~~where the operations further comprise:~~

~~generating an adjusted playback signal using the playback correction.~~ wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

13.(currently amended) The audio device according to claim [[12]]1, wherein the earphone includes a second microphone. ~~where the operations further comprise:~~

~~sending the adjusted playback signal to the speaker.~~

14.(currently amended) The audio device according to claim [[10]]13, wherein the operations further comprise:

receiving a second microphone signal from the second microphone. ~~seal quality is determined at least in part by comparing frequency components below 1000Hz of the microphone signal and the playback signal.~~

15.(currently amended) The audio device according to claim 14, wherein the first signal is the fist microphone signal and the second signal is the second microphone signal. ~~microphone signal is buffered in the memory~~.

16.(currently amended) The audio device of claim [[10]]14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal ~~the audio device is a headphone.~~

17.(currently amended) The audio device of claim [[10]]14, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal. ~~the audio device is an earphone.~~

18.(currently amended) The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value. ~~earphone according to claim 17, where the sealing section is an eartip.~~

19.(currently amended) The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof. ~~The earphone according to claim 16, where the sealing section is a foam cup.~~

20.(new) The audio device of claim 1, wherein the sealing section is an eartip.

Electronic Patent Application Fee Transmittal

Application Number:	17182570
Filing Date:	23-Feb-2021
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-16838277

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Pages:				
Claims:				
CLAIMS IN EXCESS OF 20	2202	2	50	100
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Extension-of-Time:				
Extension - 1 month with $0 paid	2251	1	110	110
Miscellaneous:				
		Total in USD ($)		**210**

Electronic Acknowledgement Receipt	
EFS ID:	45491649
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	18-APR-2022
Filing Date:	23-FEB-2021
Time Stamp:	09:14:49
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$210
RAM confirmation Number	E20224H016392405
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	Fee Worksheet (SB06)	fee-info.pdf	40145 b4eb4b5b9e2423c98b8b759c4d3ed0cc19ff8464	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	40145		

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New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

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

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
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

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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]

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

	1	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00282, December 21, 2021	
	2	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00242, December 23, 2021	
	3	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00243, December 23, 2021	
	4	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00234, December 21, 2021	
	5	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00253, January 18, 2022	
	6	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00324, January 13, 2022	
	7	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00281, January 18, 2022	
	8	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00302, January 13, 2022	
	9	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00369, February 18, 2022	
	10	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00388, February 18, 2022	
	11	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00410, February 18, 2022	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

If you wish to add additional non-patent literature document citation information please click the Add button	Add		
EXAMINER SIGNATURE			
Examiner Signature		Date Considered	

*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	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-04-18
Name/Print	John P. Keady	Registration Number	56389

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

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

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

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

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 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.

Electronic Patent Application Fee Transmittal

Application Number:	17182570
Filing Date:	23-Feb-2021
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-16838277

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

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

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Miscellaneous:				
SUBMISSION- INFORMATION DISCLOSURE STMT	2806	1	130	130
		Total in USD ($)		**130**

Electronic Acknowledgement Receipt	
EFS ID:	45491763
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	18-APR-2022
Filing Date:	23-FEB-2021
Time Stamp:	09:34:56
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$130
RAM confirmation Number	E20224H036312556
Deposit Account	
Authorized User	
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	17182570-trns-ltr.pdf	120757 0717b14177481ca64618de0eef695d45328b12b9	no	1
Warnings:					
Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	IPR-IDS.pdf	1034835 4df4ab5d46c07e582b2d4f74fcadc3a1651965d6	no	5
Warnings:					
Information:					
A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
3	Non Patent Literature	IPR1A.pdf	1065114 b32633d9277d1d5cbb0af2066ff5867d862385c1	no	80
Warnings:					
Information:					
4	Non Patent Literature	IPR2A.pdf	1943835 da801697fc2947375e698af3715a85d228c3169b	no	79
Warnings:					
Information:					
5	Non Patent Literature	IPR3A.pdf	1479136 586f5587012890efc764c06c62325f6658d3e6b0	no	80
Warnings:					
Information:					

6	Non Patent Literature	IPR4A.pdf	5856275 bbb6e06c8da39022bf9185c6f1f5e759e4c972a1	no	88
Warnings:					
Information:					
7	Non Patent Literature	IPR5A.pdf	8407299 889345276652cbf08e8b18236b62f4a383c4f305	no	93
Warnings:					
Information:					
8	Non Patent Literature	IPR6A.pdf	2934158 e3cdedca197bf8edfaca52fd01beac9ae8e9dc1f	no	96
Warnings:					
Information:					
9	Non Patent Literature	IPR7A.pdf	8015419 daf57ea06e8b5a9fac276fdf4986680d2d48e57a	no	91
Warnings:					
Information:					
10	Non Patent Literature	IPR8A.pdf	1207844 e3045ff9a612a959a7d0b32668a27ada07cb7f9f	no	79
Warnings:					
Information:					
11	Non Patent Literature	IPR9A.pdf	2841009 9ef3d0e90e5f4822674c42f5aafb0d7a2ac41139	no	84
Warnings:					
Information:					
12	Non Patent Literature	IPR10A.pdf	1712217 0fc9ba7b23d69dc17b42f5eceb527eae3739084f	no	58
Warnings:					
Information:					

13	Non Patent Literature	IPR11A.pdf	2025912 bc48559c25842c20f54e91f5b4ff83950ac0ced29	no	84
Warnings:					
Information:					
14	Fee Worksheet (SB06)	fee-info.pdf	37649 8ebb3d755e40a8794ec9cdbf165a7576f626931c	no	2
Warnings:					
Information:					
			Total Files Size (in bytes):	38681459	

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

New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/182,570	CONFIRMATION NO. 5671
APPLICATION OF	:	John Usher	
DATE FILED	:	23 Feb. 2021	
EXAMINER	:	KIM, PAUL	
TC/A.U.	:	2654	
DOCKET	:	CONT-16838277	
TITLE	:	ACOUSTIC SEALING ANALYSIS SYSTEM	

Transmittal Letter

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

Date: April 18, 2022

Sir:

The attached IDS contains the IPRs whose references have already been previously submitted to the Examiner, however in the abundance of due diligence, Applicant has included the IPRs in the attached IDS for the Examiner's consideration. Note the Patent Owner POPR responses have not been included. Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

Date: April 18, 2022

/John P. Keady/
John P. Keady
Reg. No. 56,389
571-606-1291

UNITED STATES PATENT AND TRADEMARK OFFICE

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

NOTICE OF ALLOWANCE AND FEE(S) DUE

150139 7590 11/16/2022

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

EXAMINER
KIM, PAUL

ART UNIT	PAPER NUMBER
2654	

DATE MAILED: 11/16/2022

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	02/23/2021	John Usher	CONT-16838277	5671

TITLE OF INVENTION: ACOUSTIC SEALING ANALYSIS SYSTEM

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$600	$0.00	$0.00	$600	02/16/2023

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 1/2 the amount of undiscounted fees, and micro entity fees are 1/2 the amount of small entity fees.

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

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

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

PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), 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 including the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

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

150139 7590 11/16/2022
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.
17/182,570	02/23/2021	John Usher	CONT-16838277	5671

TITLE OF INVENTION: ACOUSTIC SEALING ANALYSIS SYSTEM

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$600	$0.00	$0.00	$600	02/16/2023

EXAMINER	ART UNIT	CLASS-SUBCLASS
KIM, PAUL	2654	381-059000

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) ❑Advance Order - # of Copies _____

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

❑ Electronic Payment via 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

❑ Applicant asserting small entity status. See 37 CFR 1.27

❑ Applicant changing to regular undiscounted fee status.

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.
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.
NOTE: Checking this box will be taken to be a notification of loss of entitlement to small or micro entity status, as applicable.

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

Authorized Signature _____ Date _____

Typed or printed name _____ Registration No. _____

UNITED STATES 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.
17/182,570	02/23/2021	John Usher	CONT-16838277	5671

150139 7590 11/16/2022

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

EXAMINER	
KIM, PAUL	
ART UNIT	**PAPER NUMBER**
2654	

DATE MAILED: 11/16/2022

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. 17/182,570	Applicant(s) Usher et al.	
	Examiner PAUL KIM	Art Unit 2654	AIA (FITF) Status Yes

-- 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 2/23/2021.
   ☐ 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 1-20 . As a result of the allowed claim(s), you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see **http://www.uspto.gov/patents/init_events/pph/index.jsp** or send an inquiry to **PPHfeedback@uspto.gov.**

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

   Certified copies:

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

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

   * Certified copies not received: _____ .

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

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

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

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

Attachment(s)

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

/PAUL KIM/
Primary Examiner, Art Unit 2654

DETAILED ACTION

1. The present application, filed on or after March 16, 2013, is being examined under the first inventor to file provisions of the AIA.

EXAMINER'S AMENDMENT

2. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

The application has been amended as follows:

Claim 1 (amended): An audio device configured to test [[the]] a seal of an earphone comprising:

a memory that stores instructions; and

a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:

generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between [[the]] a range of 30Hz and 200Hz;

sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing

section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user,

receiving a first microphone signal from the first microphone;

calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;

comparing the value to a threshold to determine a seal quality between the first side and the second side;

generating a message to convey the seal quality; and

sending the message to the user.

Claim 16 (amended): The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.

Allowable Subject Matter

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

Regarding independent claim 1, the closest prior art of record, Svean et al. (US Pat. 6567524) discloses an audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: wherein the earphone includes a first microphone, the speaker, and a

sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user.

However, Svean fails to teach the combination of an audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30Hz and 200Hz; sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination; comparing the value to a threshold to determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user. The distinct features, as disclosed in independent claim 1, render the claims allowable.

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

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to PAUL KIM whose telephone number is (571)270-7697. The examiner can normally be reached on 9 AM - 5 PM, PST.

Examiner interviews are available via telephone, in-person, and video conferencing using a USPTO supplied web-based collaboration tool. To schedule an interview, applicant is encouraged to use the USPTO Automated Interview Request (AIR) at http://www.uspto.gov/interviewpractice.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, VIVIAN CHIN can be reached on (571) 272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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

/PAUL KIM/
Primary Examiner, Art Unit 2654

Notice of References Cited	Application/Control No. 17/182,570	Applicant(s)/Patent Under Reexamination Usher et al.	
	Examiner PAUL KIM	Art Unit 2654	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-6567524-B1	05-2003	Svean; Jarle	A61F11/08	381/317
	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.

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

CPC								
Symbol							**Type**	**Version**
H04R	/	29	/	001			F	2013-01-01
H04R	/	25	/	70			I	2013-01-01
H04R	/	1	/	1016			I	2013-01-01
H04R	/	1	/	1091			I	2013-01-01
H04R	/	29	/	00			I	2013-01-01
H04R	/	2460	/	15			A	2013-01-01
H04R	/	2460	/	07			A	2013-01-01

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

NONE		Total Claims Allowed:	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	05 November 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office — Part of Paper No.: 20221105

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

INTERNATIONAL CLASSIFICATION					
CLAIMED					
H04R29/00	/	29	/	00	
H04R1/10	/	1	/	10	

NON-CLAIMED					
H04R25/00	/	25	/	00	

US ORIGINAL CLASSIFICATION	
CLASS	**SUBCLASS**
381	59

CROSS REFERENCES(S)						
CLASS	**SUBCLASS (ONE SUBCLASS PER BLOCK)**					
381	60	58				

NONE		**Total Claims Allowed:**	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	05 November 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office — Part of Paper No.: 20221105

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

☐ 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	11	10	19	19										
2	2	9	11	20	20										
3	3	12	12												
4	4	13	13												
5	5	14	14												
6	6	15	15												
7	7	16	16												
8	8	17	17												
10	9	18	18												

NONE		**Total Claims Allowed:**	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	05 November 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office Part of Paper No.: 20221105

Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	Art Unit
	PAUL KIM	2654

CPC - Searched*		
Symbol	**Date**	**Examiner**
H04R25/70 OR H04R29/001 OR H04R29/00 OR H04R2460/15 OR H04R1/1016 OR H04R1/1091 OR H04R2460/07	11/05/2022	PK

CPC Combination Sets - Searched*		
Symbol	**Date**	**Examiner**

US Classification - Searched*			
Class	**Subclass**	**Date**	**Examiner**
381	58, 59, 60	11/05/2022	PK

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

/PAUL KIM/ Primary Examiner, Art Unit 2654	

Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	Art Unit
	PAUL KIM	2654

Search Notes		
Search Notes	**Date**	**Examiner**
Inventor search - John Usher, John Keady	11/05/2022	PK
PE2E search - see PE2E search note	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 01/14/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 05/18/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 06/14/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 06/23/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 07/1/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 08/12/2022	11/05/2022	PK

Interference Search			
US Class/CPC Symbol	**US Subclass/CPC Group**	**Date**	**Examiner**
	See attached search history	11/05/2022	PK

/PAUL KIM/ Primary Examiner, Art Unit 2654	

PE2E SEARCH - Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
L1	1	"17182570"	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 02:30 AM
L2	12	"12555864"	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 02:30 AM
L3	117	((("Staton") near3 ("Techiya") near3 ("LLC"))).AS,AANM.	(USPAT)	OR	ON	ON	2022/11/05 02:31 AM
L4	289	((("Usher") near3 ("John"))).INV.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:32 AM
L5	149	((("Keady") near3 ("John") near3 ("P"))).INV.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:32 AM
L6	245	((("Staton") near3 ("Techiya") near3 ("LLC"))).AS,AANM.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:33 AM
L7	917	(381/59).ccls.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:34 AM
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L9	5965	H04R1/1016	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L10	2719	H04R1/1091	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L12	2674	H04R25/70	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L13	8905	H04R29/00	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:36 AM
L14	892	H04R2460/07	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:36 AM
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L16	19612	L8 OR L9 OR L10 OR L12 OR L13 OR L14 OR l15	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:37 AM
L17	13	L8 AND L9 AND L10 AND L12 AND L13 AND L14 AND L15	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:49 AM
L18	652	L7 AND L16	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:51 AM
L19	1	"20060188105"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 04:47 AM
L20	128	"6567524"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 04:48 AM
L21	2764	(381/58,59,60).ccls.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:00 PM

L22	42	("5142587"|"5764781"|"8774435"|"20050078838"|"5267321"|"6021207"|"7072482"|"10860287"|"7801318"|"8971561"|"9293128"|"7477756"|"20140247948"|"4020297"|"5319163"|"7688983"|"8550206"|"20080037801"|"6567524"|"6728385"|"6754359"|"20130177179"|"5138722"|"5712453"|"8774433"|"6321070"|"8401200"|"6415034"|"7450730"|"8014553"|"20140247952"|"20060204014"|"4088849"|"7778434"|"RE38351"|"20180176681"|"9313572"|"6359993"|"9084060"|"20030165246"|"5524056"|"7209569").PN.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:01 PM
L23	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) SAME (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) SAME (microphone$2 WITH acoustic WITH barrier$2) SAME (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) SAME (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:08 PM
L24	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:09 PM

		earbud$2))) AND (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) AND (microphone$2 WITH acoustic WITH barrier$2) AND (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) AND (compar$4 WITH threshold WITH qualit$4)					
L25	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2) AND (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:10 PM
L26	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:10 PM
L27	358040	((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5) OR (headset$2 OR earphone$2 OR headphone$2 OR	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:11 PM

		earpiece$2 OR earbud$2))					
L28	1606	L27 AND (test$4 WITH seal$3)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:12 PM
L29	422	L28 AND L16	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:13 PM
L30	65	L29 AND L21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:13 PM
L31	119	L29 AND (test$3 near3 signal$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:15 PM
L32	25	L31 AND l21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:15 PM
L33	2215	L27 AND ((test$4 OR analy$3) WITH seal$3)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:16 PM
L34	706	l33 AND (test$3 near3 signal$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L35	123	L34 AND L16	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L36	27	L34 AND L21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L37	44	(Artificial Intelligence) Similar to: 16/838,277 with 0 CPC Selections and 0 Text Selections Text: (US-10986432-B2 OR EP-3646614-A1 OR WO-2019005492-A1 OR US-20190007762-A1 OR EP-2737720-A2 OR US-20220256270-A1 OR EP-2737720-B1 OR US-20200068283-A1 OR US-10972823-B2 OR WO-2013016336-A2 OR US-20210022996-A1 OR US-10462549-B2 OR US-8737669-B2 OR FR-3062014-A1 OR KR-102213388-B1 OR US-7837005-B2 OR CN-113491504-A OR JP-6037483-B2 OR KR-	(US-PGPUB; USPAT; USOCR; FIT (AP, AT, AU, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DK, EA, EE, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, RU, SE, SG, SI, SK, SU, TH, TN, TR, TW, UA, VN, WO))	OR	ON	ON	2022/11/05 05:19 PM

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L41	41	(Artificial Intelligence) More like doc: US-20060188105-A1 Text: (CA-2497710-A1 OR CA-2477010-C OR US-10645505-B2 OR AU-2002220394-B2 OR CN-100505921-C OR CN-1483298-A OR EP-4075830-A1 OR CN-112534221-A OR US-20090220099-A1 OR AU-2004310732-B2 OR US-20050123146-A1 OR CA-2952220-C OR AU-2004310732-A1 OR WO-2005055656-A1 OR EP-1702497-A1 OR CA-2549279-A1 OR CA-2549279-C OR US-8254586-B2 OR AU-2004310732-A2 OR US-20200196078-A1).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	2022/11/05 05:21 PM
L42	5	("2006/0188105").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:22 PM
L43	37	(Artificial Intelligence) More like doc: US-7801318-B2 Text: (US-20060285709-A1 OR EP-1737270-B2 OR EP-1737270-B1 OR EP-1737270-A1 OR DE-102005028742-B3 OR JP-H06189397-A OR AU-2008306923-B2 OR EP-2046073-B1 OR US-20100254555-A1 OR EP-0581261-B1 OR US-10009695-B2 OR AU-660818-B2 OR US-9432783-B2 OR EP-2661103-A1 OR JP-	(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	2022/11/05 05:23 PM

		WO2010061539-A1 OR JP-4652488-B2 OR EP-0581261-A1 OR DK-1690252-T3 OR US-20130294610-A1 OR CA-2100015-A1).did.					
L44	21	("4689820" | "5259033" | "5402496" | "5524150" | "6219427" | "6398858" | "20020176584" | "20030161492" | "20040013280" | "20050047620" | "20050190929").pn. OR ("7801318").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:23 PM

PE2E SEARCH - Search History (Interference)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
N1	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) SAME (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) SAME (microphone$2 WITH acoustic WITH barrier$2) SAME (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) SAME (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:08 PM
N2	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:09 PM

		earbud$2))) AND (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) AND (microphone$2 WITH acoustic WITH barrier$2) AND (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) AND (compar$4 WITH threshold WITH qualit$4)					
N3	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2) AND (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:10 PM
N4	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:10 PM

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

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

Application Number	17182570
Filing Date	2021-02-23
First Named Inventor	John Usher
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-16838277

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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
	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	

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	1	20030165246		2003-09-04	Kvaloy et al.	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	2	20080037801		2008-02-14	Alves et al.	
	3	20060204014		2006-09-14	Isenberg et al.	
	4	20050078838		2005-04-14	Simon	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

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	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

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

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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	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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

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Application Number	17182570
Filing Date	2021-02-23
First Named Inventor	John Usher
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-16838277

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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
	1	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
	5	7562020		2009-06-14	Le et al.	
	6	9135797		2015-09-15	Couper et al.	
	7	9123343		2015-09-01	Kurki-Suonio	
	8	7756285		2010-07-13	Sjursen et al.	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	9	5903868		1999-05-11	Yuen et al.	
	10	3876843		1975-04-08	Moen	
	11	5208867		1993-05-04	Stites, III	
	12	6226389		2001-05-01	Lemelson et al.	
	13	8493204		2013-07-23	Wong et al.	
	14	6661901		2003-12-09	Svean et al.	
	15	7430299		2008-09-30	Armstrong et al.	
	16	6567524		2003-05-20	Svean et al.	
	17	7107109		2006-09-12	Nathan et al.	
	18	4947440		1990-08-07	Bateman et al.	
	19	9037458		2015-05-19	Park et al.	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	20	4054749		1977-10-18	Suzuki et al.	
	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
	25	6021325		2000-02-01	Hall	
	26	8750295		2014-06-10	Liron	
	27	7920557		2011-04-05	Moote	

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	1	20030165246		2003-09-04	Kvaloy et al.	

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	First Named Inventor	John Usher
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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	2	20050288057		2005-12-29	Lai et al.	
	3	20060195322		2006-08-31	Broussard et al.	
	4	20070043563		2007-02-22	Comerford et al.	
	5	20010046304		2001-11-29	Rast	
	6	20070189544		2007-08-16	Rosenberg	
	7	20060092043		2006-05-04	Lagassey	
	8	20020106091		2002-08-08	Furst et al.	
	9	20040109668		2004-06-10	Stuckman	
	10	20020118798		2002-08-29	Langhart et al.	
	11	20040042103		2004-03-04	Mayer	
	12	20030161097		2003-08-28	Le et al.	

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	13	20160104452		2016-04-14	Guan et al.	
	14	20040203351		2004-10-14	Shearer et al.	
	15	20080165988		2008-07-10	Terlizzi et al.	
	16	20070291953		2007-12-20	Ngia et al.	
	17	20110096939		2011-04-28	Ichimura	
	18	20060067551		2006-03-30	Cartwright et al.	
	19	20090024234		2009-01-22	Archibald	
	20	20090010456		2009-01-08	Goldstein et al.	
	21	20070086600		2007-04-19	Boesen	
	22	20040190737		2004-09-30	Kuhnel et al.	
	23	20040196992		2004-10-07	Ryan	

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

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	1	EP1519625	EP	A2		Victorian		
	2	2006037156	WO	A1		Mejia et al.		

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	1	Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan 9-12, 2005, p. 284-286.	
	2	Bernard Widrow, John R. Glover Jr., John M. McCool, John Kaunitz, Charles S. Williams, Robert H. Hearn, James R. Zeidler, Eugene Dong Jr, and Robert C. Goodlin, Adaptive Noise Cancelling: Principles and Applications, PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975	
	3	Mauro Dentino, John M. McCool, and Bernard Widrow, Adaptive Filtering in the Frequency Domain, PROCEEDINGS OF THE IEEE, VOL. 66, NO. 12, DECEMBER 1978	

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Examiner Signature	/PAUL KIM/	Date Considered	11/05/2022

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See attached certification statement.

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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

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Application Number	
Filing Date	
First Named Inventor	John Usher
Art Unit	
Examiner Name	Unknown
Attorney Docket Number	CONT-16838277

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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
	1	7801318		2010-09-21	Barthel	
	2	8774433		2014-07-08	Goldstein	
	3	7801318		2010-09-21	Barthel, Roland	
	4	8401200		2009-11-19	Tiscareno	
	5	5138722		2010-09-21	Urella	
	6	5764781		1998-06-09	Ding	
	7	6321070		2001-11-20	Clark	
	8	9293128		2016-03-22	Goldstein	

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	Filing Date	
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	Examiner Name	Unknown
	Attorney Docket Number	CONT-16838277

	9	4020297		1977-04-26	Brodie S Dan	
	10	5142587		1992-08-25	Kobayashi	
	11	5319163		1994-06-07	Scott	
	12	5712453		1998-01-27	Bungardt	
	13	9084060		2015-07-14	Liu	
	14	8774435		2014-07-08	Ambrose	
	15	7688983		2010-03-30	Voix	
	16	8550206		2013-10-08	Keady	
	17	10860287		2020-12-08	Konagai	
	18	8971561		2015-03-03	Howes et al.	
	19	9313572		2016-04-12	Dusan et al.	

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	1	20140247948		2014-09-04	Goldstein	
	2	20140247952		2014-09-04	Goldstein	
	3	20130177179		2013-07-11	Ambrose	
	4	20180176681		2018-06-21	Asada	

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	1	107071123	CN		2017-08-18			

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	Examiner Name	Unknown
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Bibliographic Data

Application No: 17/182,570

Foreign Priority claimed: ○ Yes ◉ No

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

Verified and Acknowledged: /PAUL KIM/ Examiner's Signature | Initials

Title: ACOUSTIC SEALING ANALYSIS SYSTEM

FILING or 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
02/23/2021	381	2654	CONT-16838277
RULE			

APPLICANTS

Staton Techiya LLC, Lighthouse Point, FL, UNITED STATES

INVENTORS

John Usher, Devon, UNITED KINGDOM

John P. Keady, Fairfax Station, VA, UNITED STATES

CONTINUING DATA

This application is a CON of 16838277 04/02/2020 PAT 10979836

16838277 is a CON of 16414136 05/16/2019 PAT 10701499

16414136 is a CON of 15700511 09/11/2017 PAT 10299053

15700511 is a CON of 14827332 08/17/2015 PAT 9781530

14827332 is a CON of 14054015 10/15/2013 PAT 9113267

14054015 is a DIV of 12555864 09/09/2009 PAT 8600067

12555864 has PRO of 61098250 09/19/2008

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

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Filing Date	2021-02-23
First Named Inventor	John Usher
Art Unit	2654
Examiner Name	KIM, PAUL
Attorney Docket Number	CONT-16838277

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	1	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00282, December 21, 2021	
	2	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00242, December 23, 2021	
	3	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00243, December 23, 2021	
	4	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00234, December 21, 2021	
	5	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00253, January 18, 2022	
	6	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00324, January 13, 2022	
	7	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00281, January 18, 2022	
	8	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00302, January 13, 2022	
	9	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00369, February 18, 2022	
	10	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00388, February 18, 2022	
	11	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00410, February 18, 2022	

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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.

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

DocCode - SCORE

SCORE Placeholder Sheet for IFW Content

Application Number: 17182570 Document Date: 11/18/2022

The presence of this form in the IFW record indicates that the following document type was received in electronic format on the date identified above. This content is stored in the SCORE database.

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Form Revision Date: March 1, 2019

Doc code: RCEX
Doc description: Request for Continued Examination (RCE)

PTO/SB/30EFS (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.

REQUEST FOR CONTINUED EXAMINATION(RCE)TRANSMITTAL
(Submitted Only via EFS-Web)

Application Number	17/182,570	Filing Date	2021-02-23	Docket Number (if applicable)	CONT-16838277	Art Unit	2654
First Named Inventor	John Usher			Examiner Name	Kim, Paul		

This is a Request for Continued Examination (RCE) under 37 CFR 1.114 of the above-identified application.
Request for Continued Examination (RCE) practice under 37 CFR 1.114 does not apply to any utility or plant application filed prior to June 8, 1995, or to any design application. The Instruction Sheet for this form is located at WWW.USPTO.GOV

SUBMISSION REQUIRED UNDER 37 CFR 1.114

Note: If the RCE is proper, any previously filed unentered amendments and amendments enclosed with the RCE will be entered in the order in which they were filed unless applicant instructs otherwise. If applicant does not wish to have any previously filed unentered amendment(s) entered, applicant must request non-entry of such amendment(s).

☐ Previously submitted. If a final Office action is outstanding, any amendments filed after the final Office action may be considered as a submission even if this box is not checked.

- ☐ Consider the arguments in the Appeal Brief or Reply Brief previously filed on _____
- ☐ Other _____

☒ Enclosed

- ☐ Amendment/Reply
- ☐ Information Disclosure Statement (IDS)
- ☐ Affidavit(s)/ Declaration(s)
- ☒ Other IDS, Replacement Sheets 1-15

MISCELLANEOUS

☐ Suspension of action on the above-identified application is requested under 37 CFR 1.103(c) for a period of months _____ (Period of suspension shall not exceed 3 months; Fee under 37 CFR 1.17(i) required)

☐ Other _____

FEES

☐ **The RCE fee under 37 CFR 1.17(e) is required by 37 CFR 1.114 when the RCE is filed.**
The Director is hereby authorized to charge any underpayment of fees, or credit any overpayments, to Deposit Account No _____

SIGNATURE OF APPLICANT, ATTORNEY, OR AGENT REQUIRED

☒ Patent Practitioner Signature

Applicant Signature

PTO/SB/30EFS (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.

Signature of Registered U.S. Patent Practitioner			
Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-11-18
Name	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.114. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450.

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

Privacy Act Statement

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

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

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these records.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent 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)).

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

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
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	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

	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		Date Considered	

*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.

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	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-11-18
Name/Print	John P. Keady	Registration Number	56389

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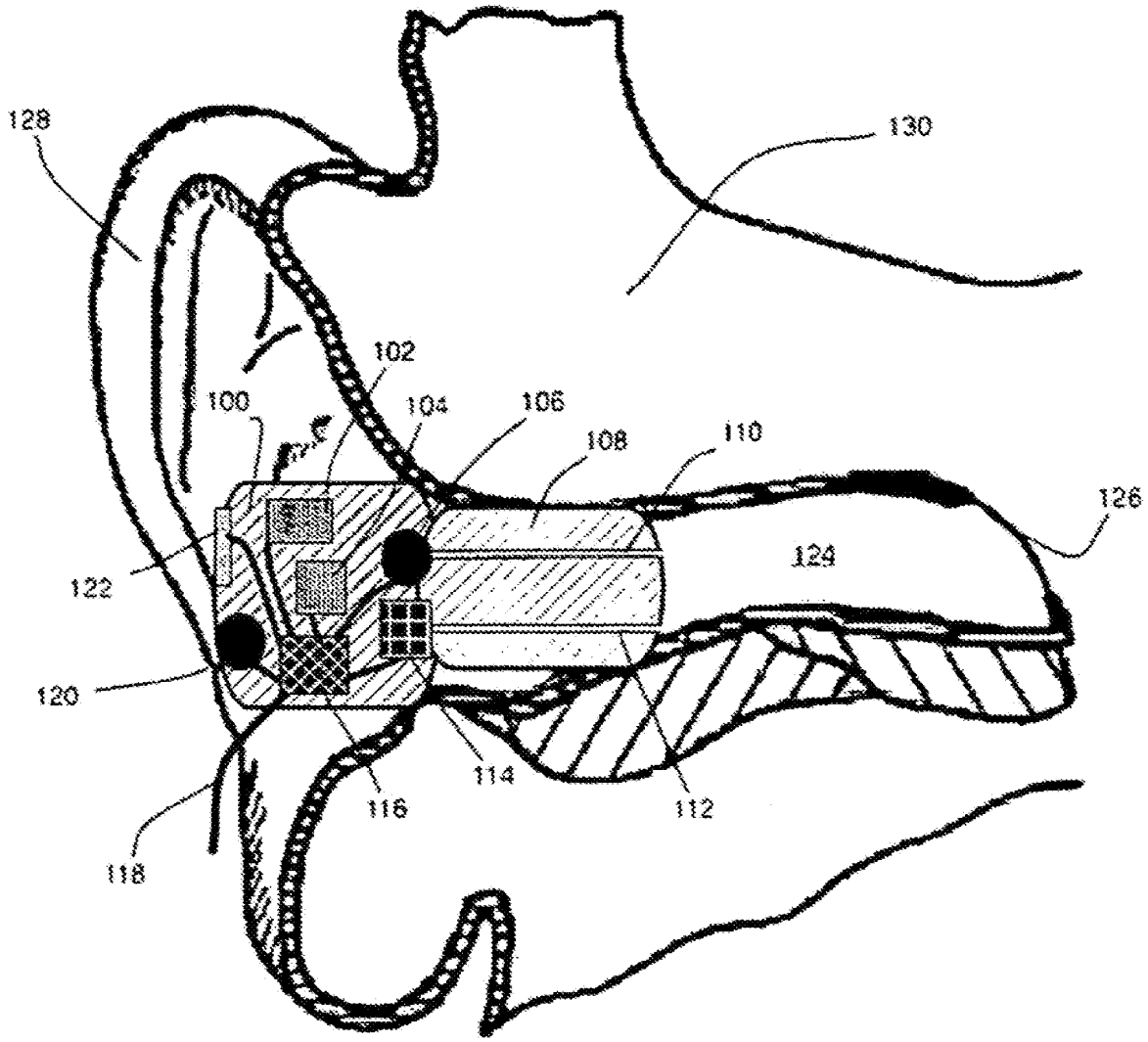
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FIG 1



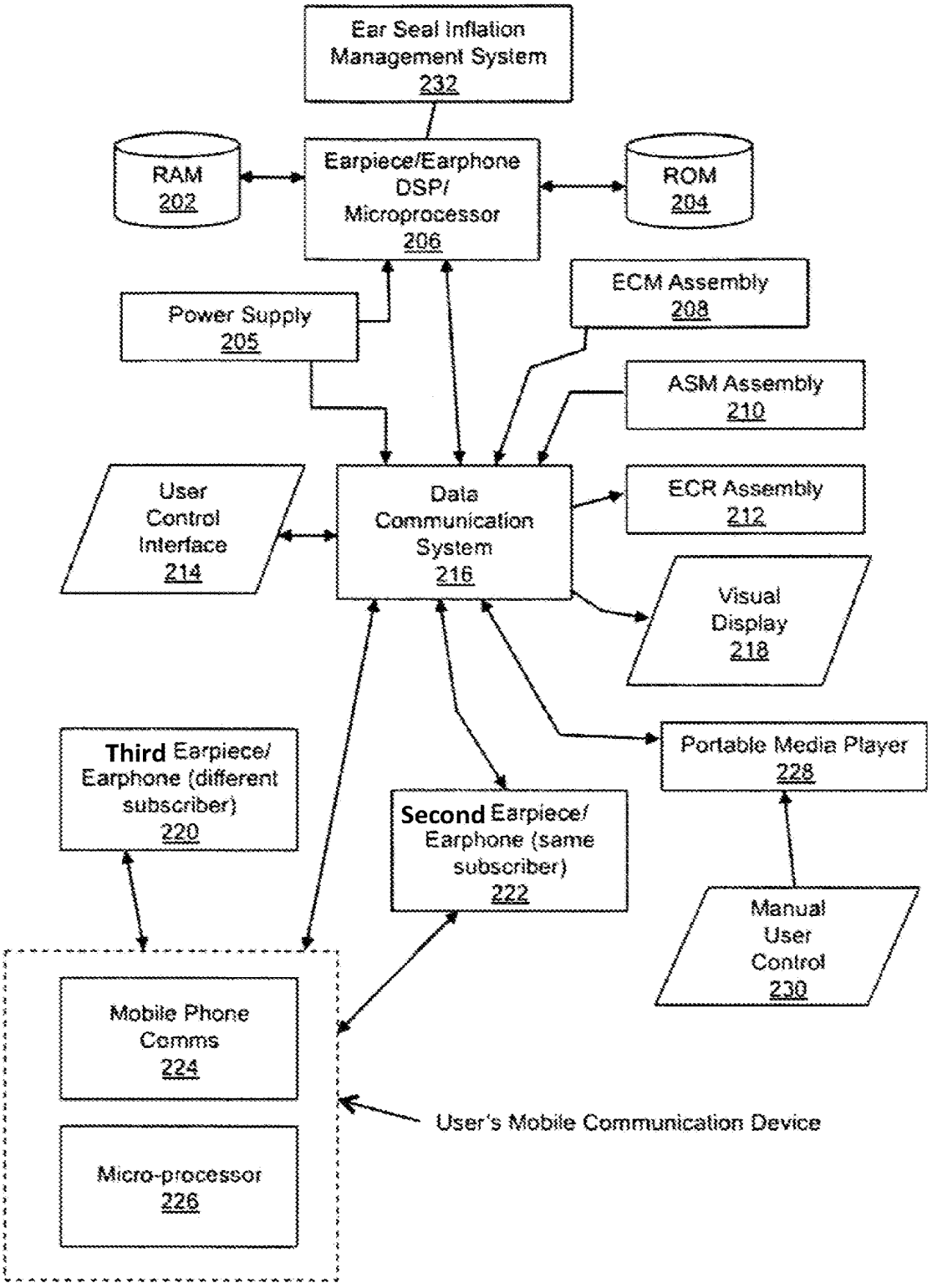

Ear Seal Inflation Management System 232
RAM 202
Earpiece/Earphone DSP/ Microprocessor 206
ROM 204
Power Supply 205
ECM Assembly 208
ASM Assembly 210
User Control Interface 214
Data Communication System 216
ECR Assembly 212
Visual Display 218
Third Earpiece/ Earphone (different subscriber) 220
Second Earpiece/ Earphone (same subscriber) 222
Portable Media Player 228
Manual User Control 230
Mobile Phone Comms 224
Micro-processor 226
User's Mobile Communication Device

FIG. 2

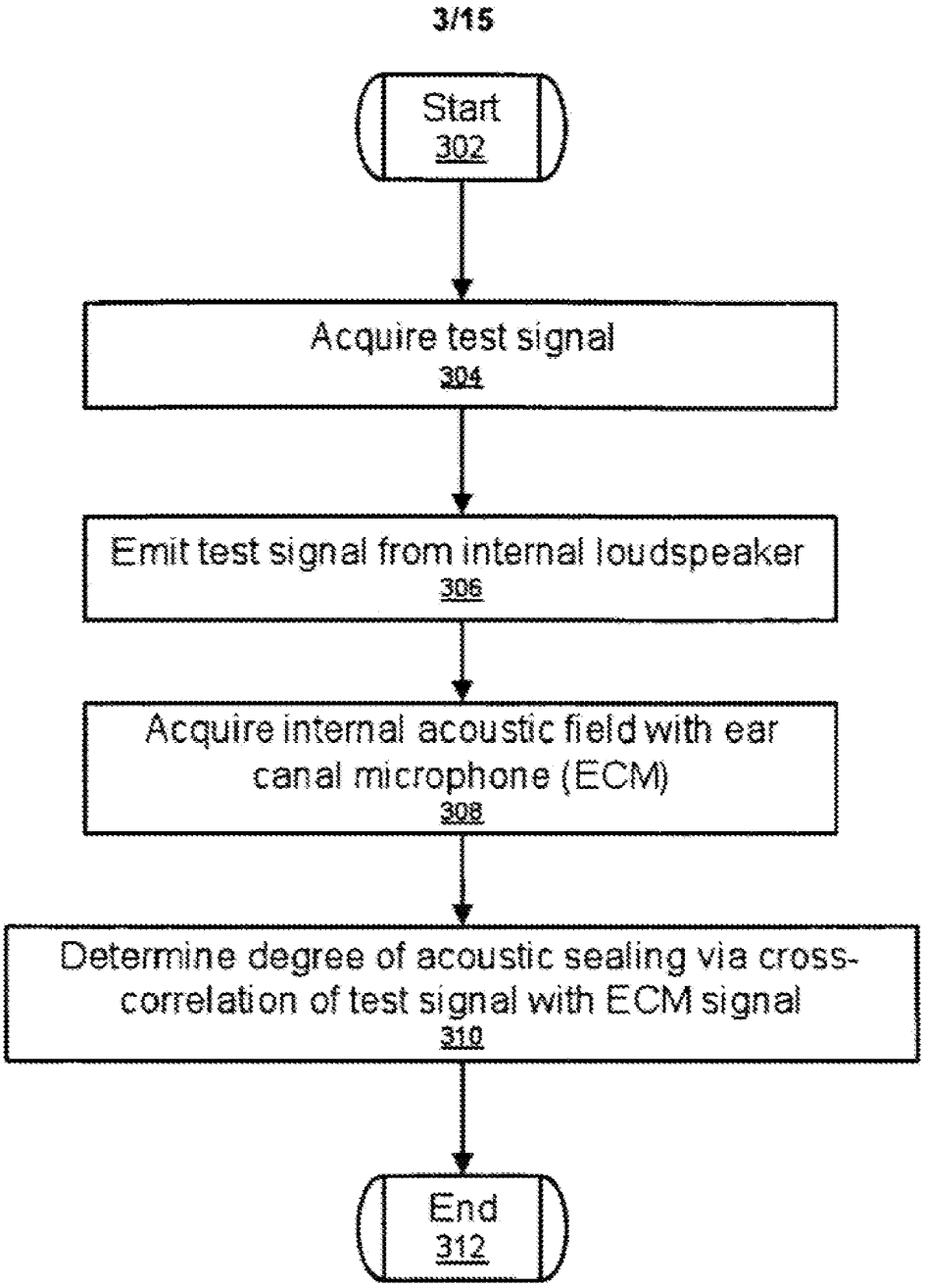

3/15
Start
302
Acquire test signal
304
Emit test signal from internal loudspeaker
306
Acquire internal acoustic field with ear canal microphone (ECM)
308
Determine degree of acoustic sealing via cross-correlation of test signal with ECM signal
310
End
312

FIG. 3

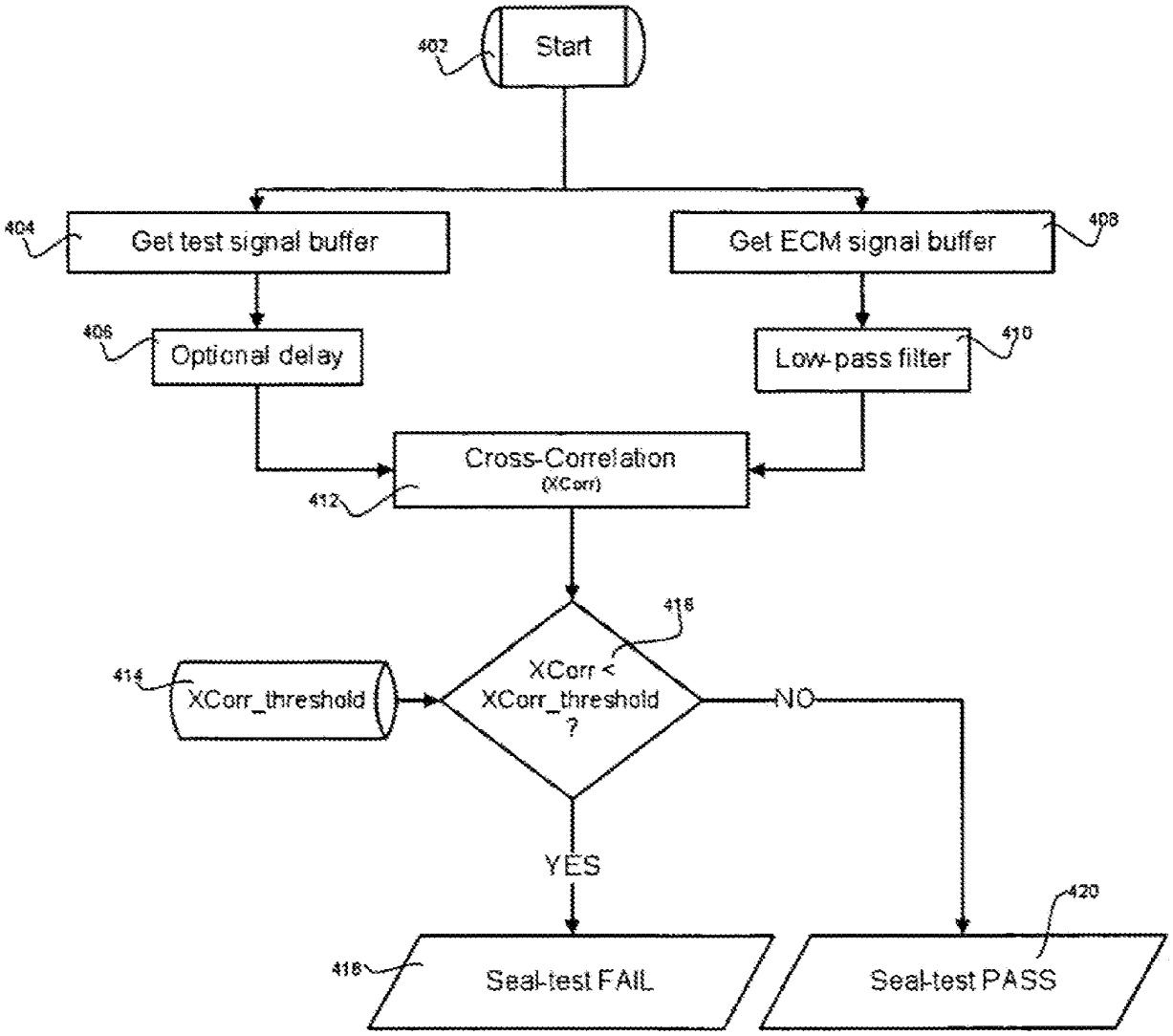

402
Start
404
Get test signal buffer
408
Get ECM signal buffer
406
Optional delay
410
Low-pass filter
Cross-Correlation
(XCorr)
412
416
XCorr < XCorr_threshold ?
414
XCorr_threshold
NO
YES
420
418
Seal-test FAIL
Seal-test PASS

FIG. 4

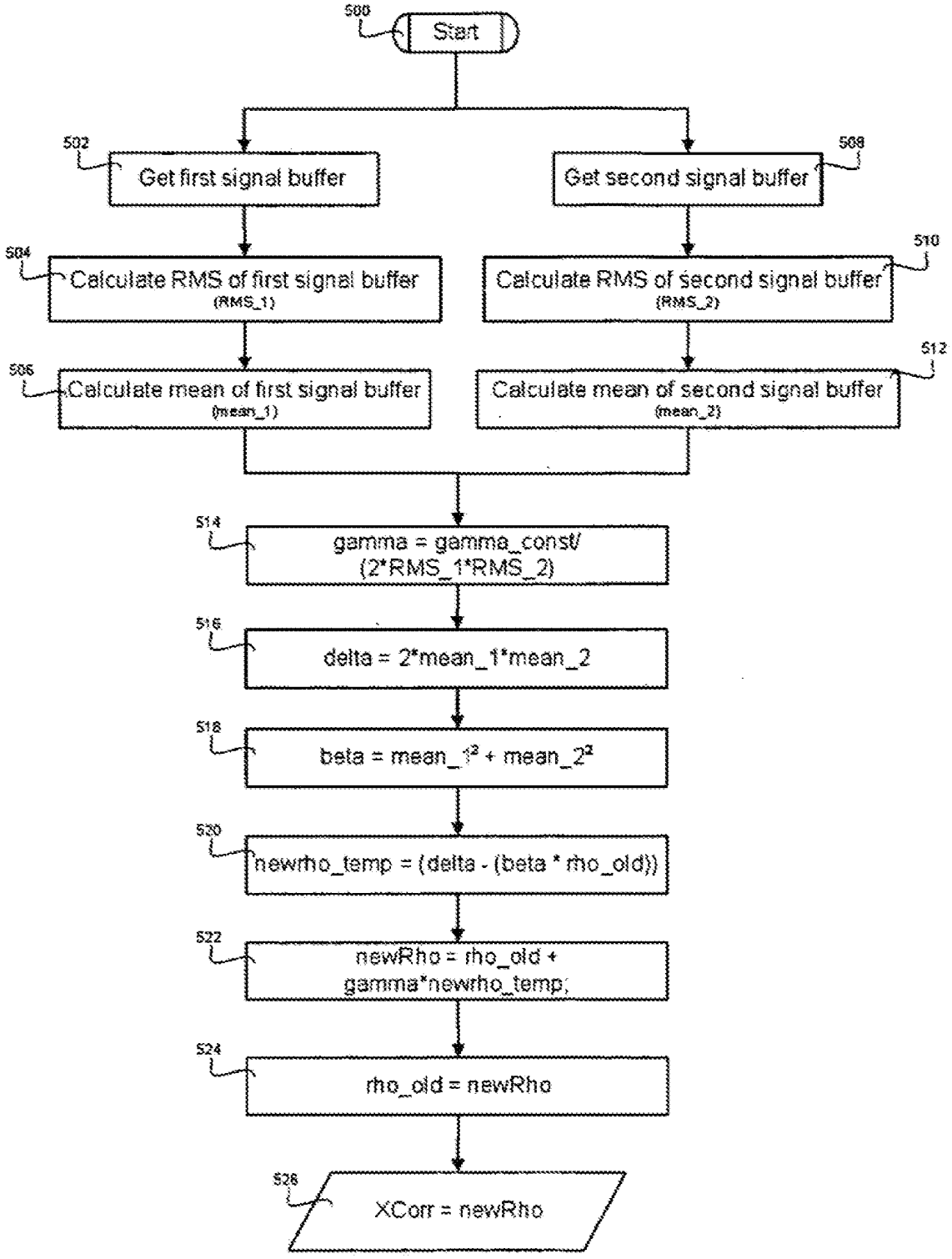

500
Start
502
Get first signal buffer
508
Get second signal buffer
504
Calculate RMS of first signal buffer
(RMS_1)
510
Calculate RMS of second signal buffer
(RMS_2)
506
Calculate mean of first signal buffer
(mean_1)
512
Calculate mean of second signal buffer
(mean_2)
514
gamma = gamma_const/
(2*RMS_1*RMS_2)
516
delta = 2*mean_1*mean_2
518
beta = mean_1^2 + mean_2^2
520
newrho_temp = (delta - (beta * rho_old))
522
newRho = rho_old +
gamma*newrho_temp;
524
rho_old = newRho
526
XCorr = newRho

FIG.5

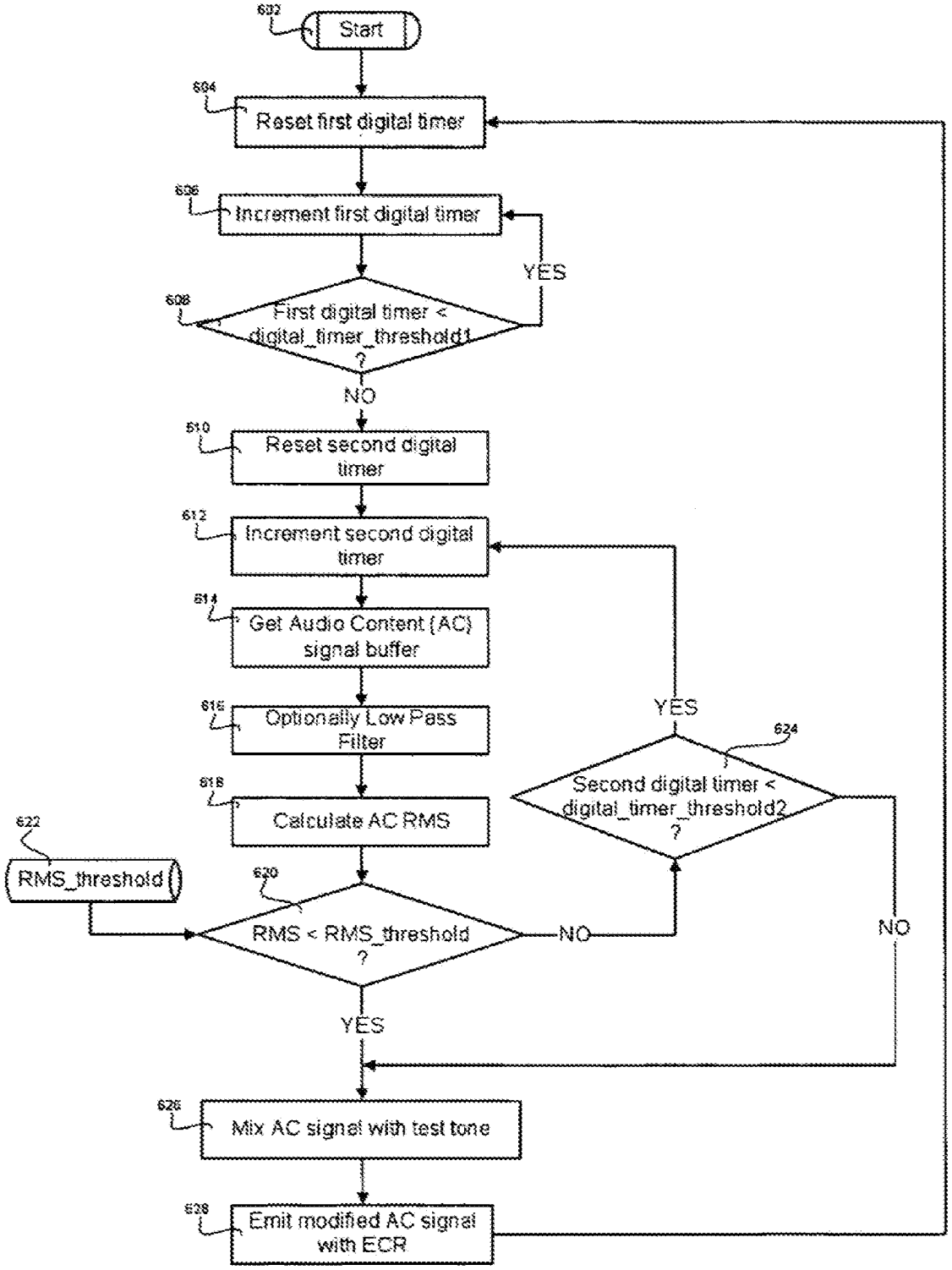

602
Start
604
Reset first digital timer
606
Increment first digital timer
608
First digital timer < digital_timer_threshold1 ?
YES
NO
610
Reset second digital timer
612
Increment second digital timer
614
Get Audio Content (AC) signal buffer
616
Optionally Low Pass Filter
618
Calculate AC RMS
622
RMS_threshold
620
RMS < RMS_threshold ?
NO
YES
624
Second digital timer < digital_timer_threshold2 ?
YES
NO
626
Mix AC signal with test tone
628
Emit modified AC signal with ECR

FIG. 6

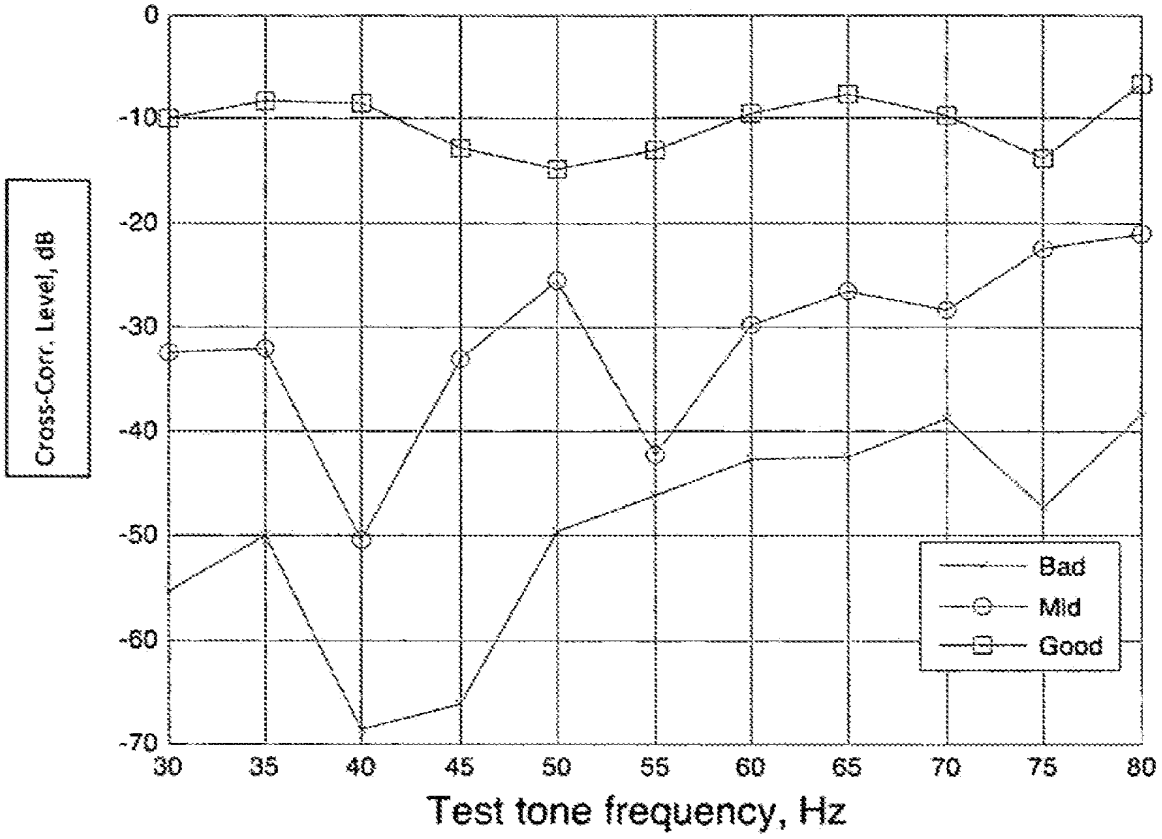

Cross-Corr. Level, dB
0
-10
-20
-30
-40
-50
-60
-70
30
35
40
45
50
55
60
65
70
75
80
Test tone frequency, Hz
Bad
Mid
Good

FIG. 7

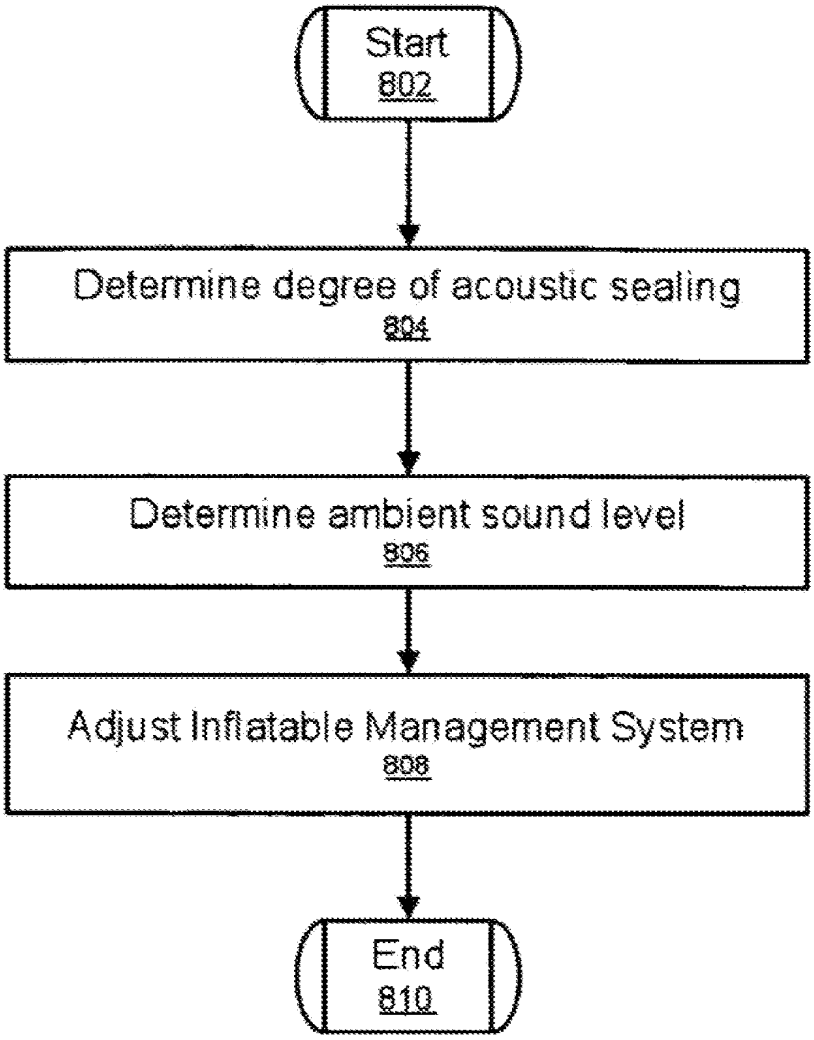

Start
802
Determine degree of acoustic sealing
804
Determine ambient sound level
806
Adjust Inflatable Management System
808
End
810

FIG. 8

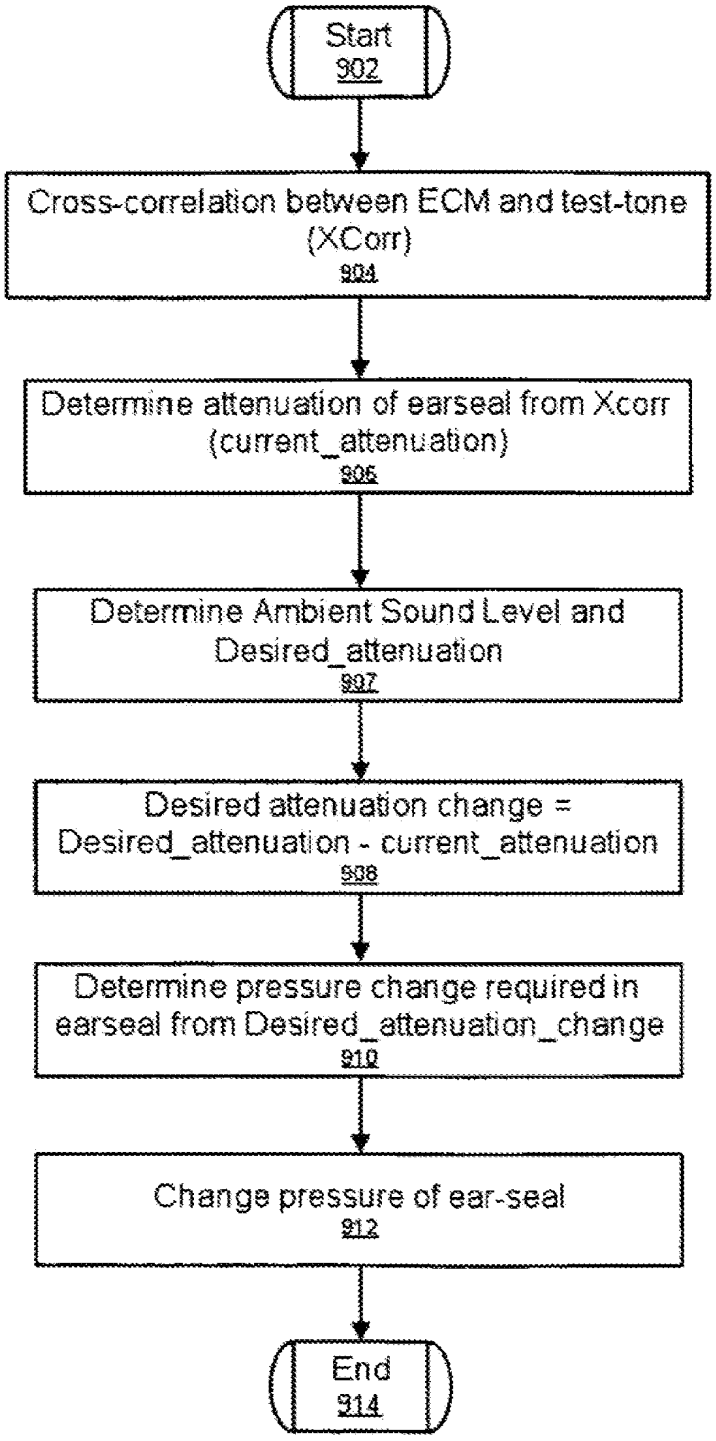

Start
902
Cross-correlation between ECM and test-tone (XCorr)
904
Determine attenuation of earseal from Xcorr (current_attenuation)
906
Determine Ambient Sound Level and Desired_attenuation
907
Desired attenuation change = Desired_attenuation - current_attenuation
908
Determine pressure change required in earseal from Desired_attenuation_change
910
Change pressure of ear-seal
912
End
914

FIG. 9

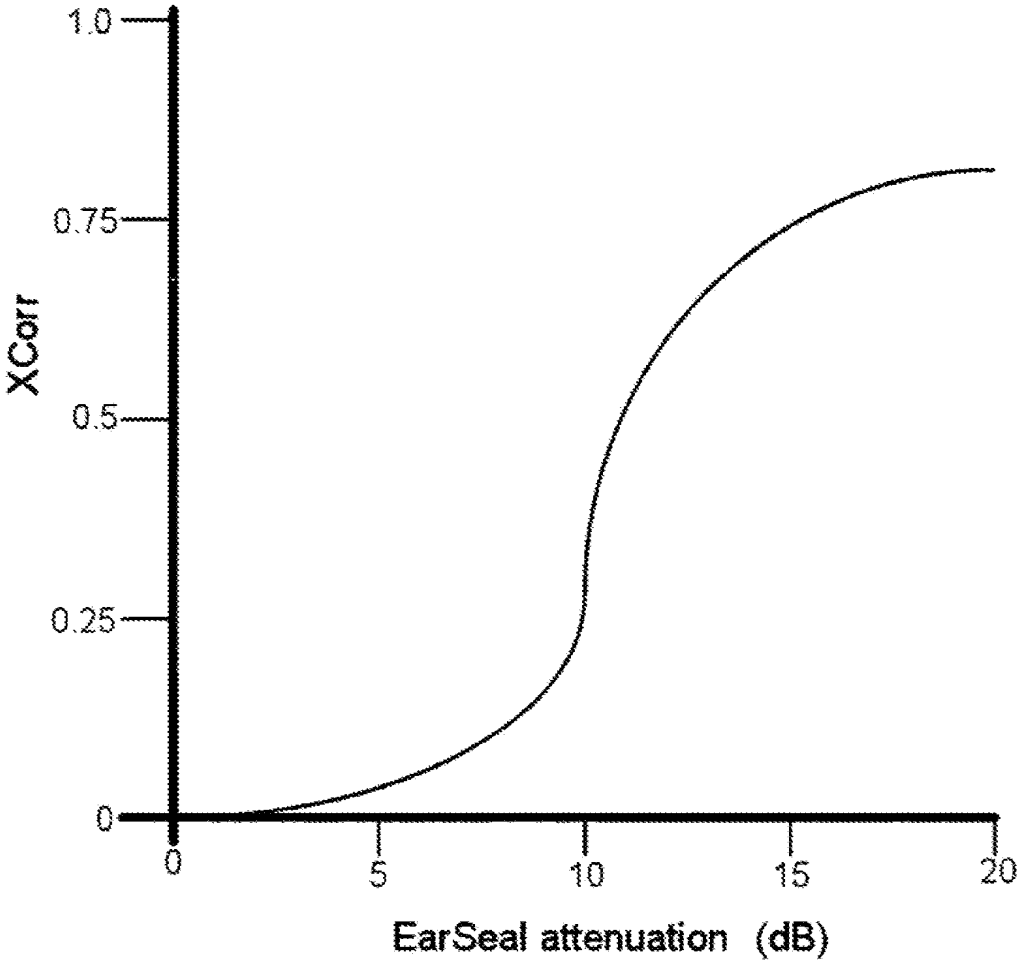


FIG. 10

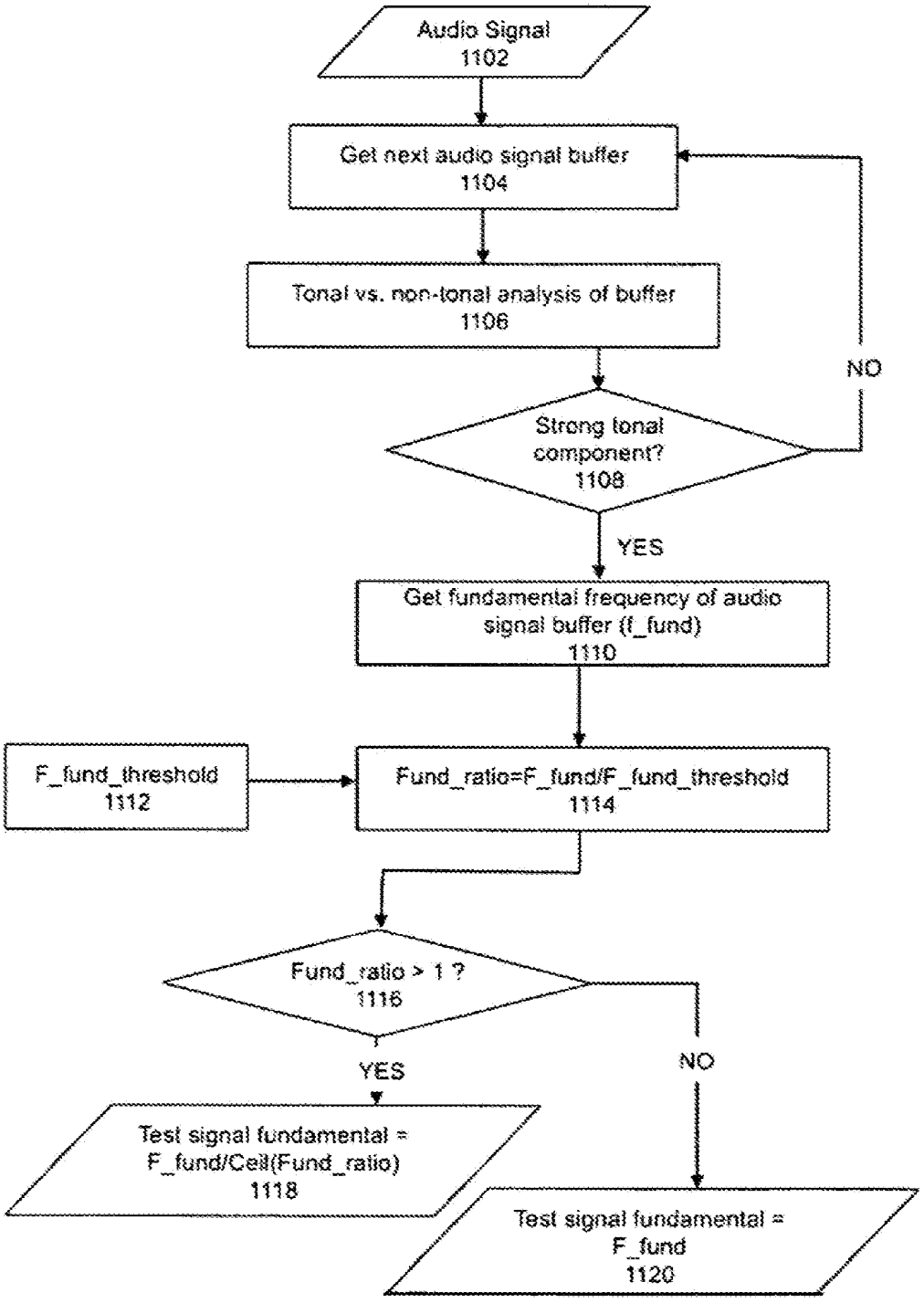

Audio Signal
1102
Get next audio signal buffer
1104
Tonal vs. non-tonal analysis of buffer
1106
Strong tonal component?
1108
NO
YES
Get fundamental frequency of audio signal buffer (f_fund)
1110
F_fund_threshold
1112
Fund_ratio=F_fund/F_fund_threshold
1114
Fund_ratio > 1 ?
1116
YES
NO
Test signal fundamental = F_fund/Ceil(Fund_ratio)
1118
Test signal fundamental = F_fund
1120

FIG. 11

Fig. 12

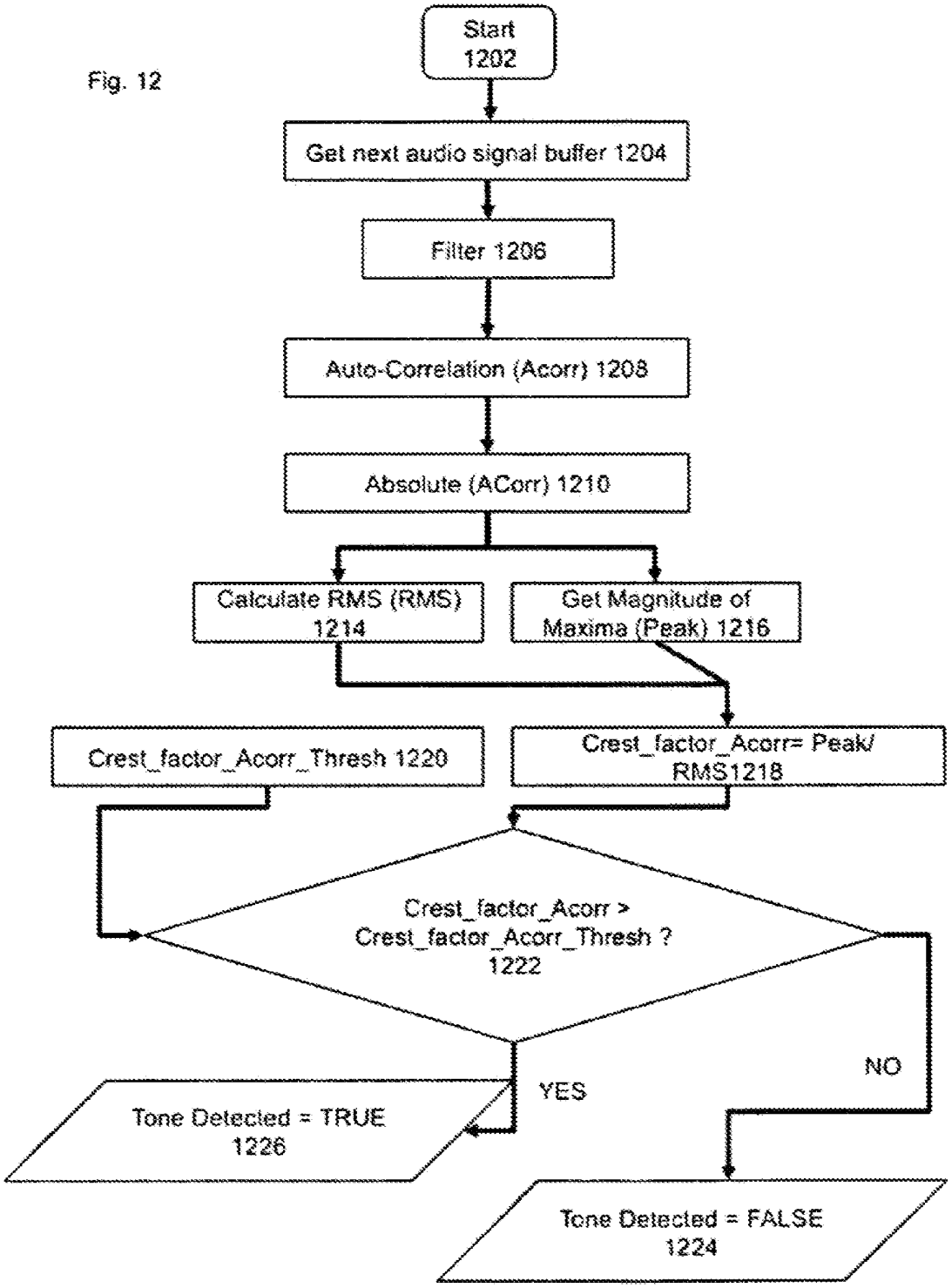

Start
1202
Get next audio signal buffer 1204
Filter 1206
Auto-Correlation (Acorr) 1208
Absolute (ACorr) 1210
Calculate RMS (RMS)
1214
Get Magnitude of
Maxima (Peak) 1216
Crest_factor_Acorr_Thresh 1220
Crest_factor_Acorr= Peak/
RMS1218
Crest_factor_Acorr >
Crest_factor_Acorr_Thresh ?
1222
YES
NO
Tone Detected = TRUE
1226
Tone Detected = FALSE
1224

Fig. 13

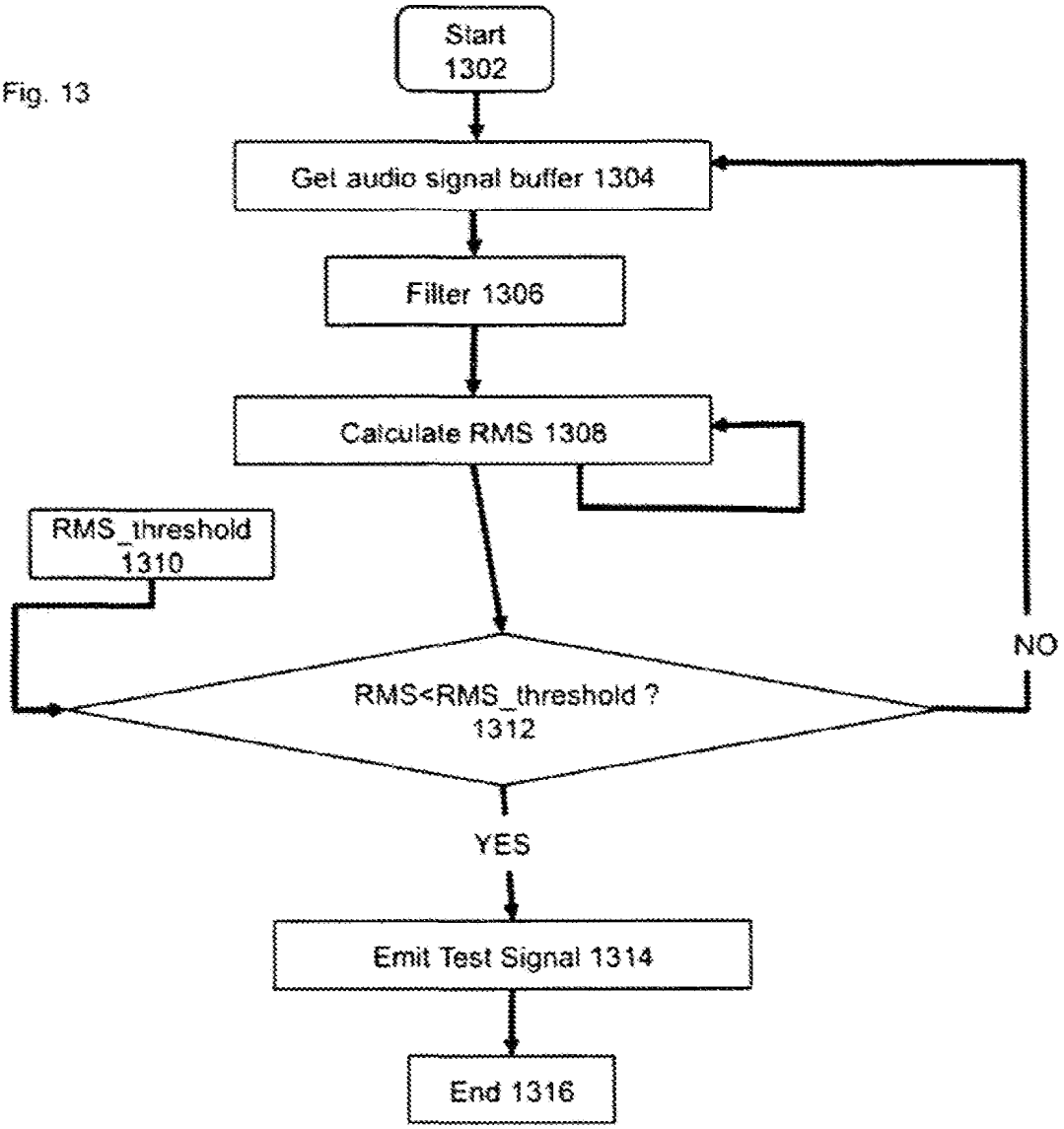

Start
1302
Get audio signal buffer 1304
Filter 1306
Calculate RMS 1308
RMS_threshold
1310
RMS<RMS_threshold ?
1312
NO
YES
Emit Test Signal 1314
End 1316

Fig. 14

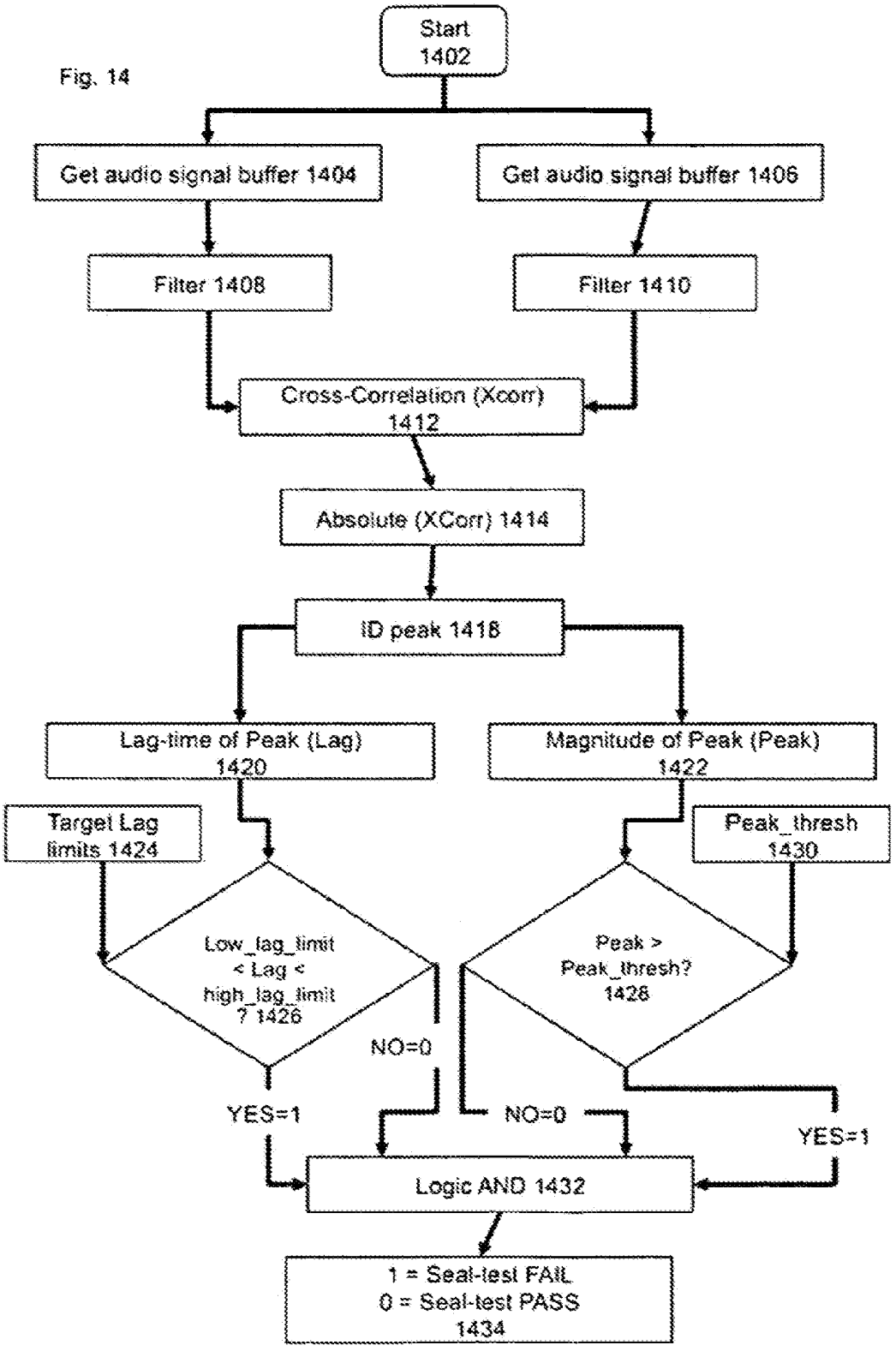

Start
1402
Get audio signal buffer 1404
Get audio signal buffer 1406
Filter 1408
Filter 1410
Cross-Correlation (Xcorr)
1412
Absolute (XCorr) 1414
ID peak 1418
Lag-time of Peak (Lag)
1420
Magnitude of Peak (Peak)
1422
Target Lag
limits 1424
Peak_thresh
1430
Low_lag_limit
< Lag <
high_lag_limit
? 1426
Peak >
Peak_thresh?
1428
NO=0
YES=1
NO=0
YES=1
Logic AND 1432
1 = Seal-test FAIL
0 = Seal-test PASS
1434

FIG. 15A

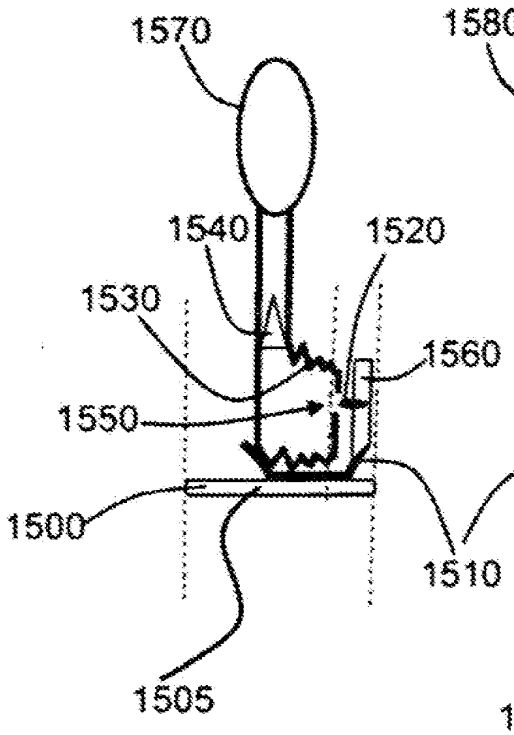


FIG. 15B

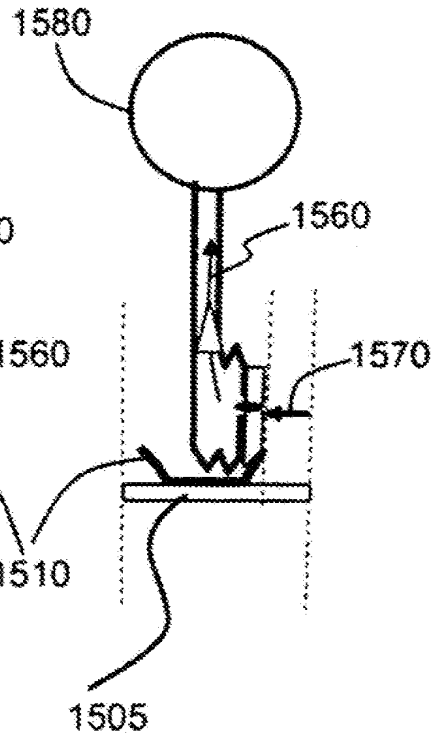
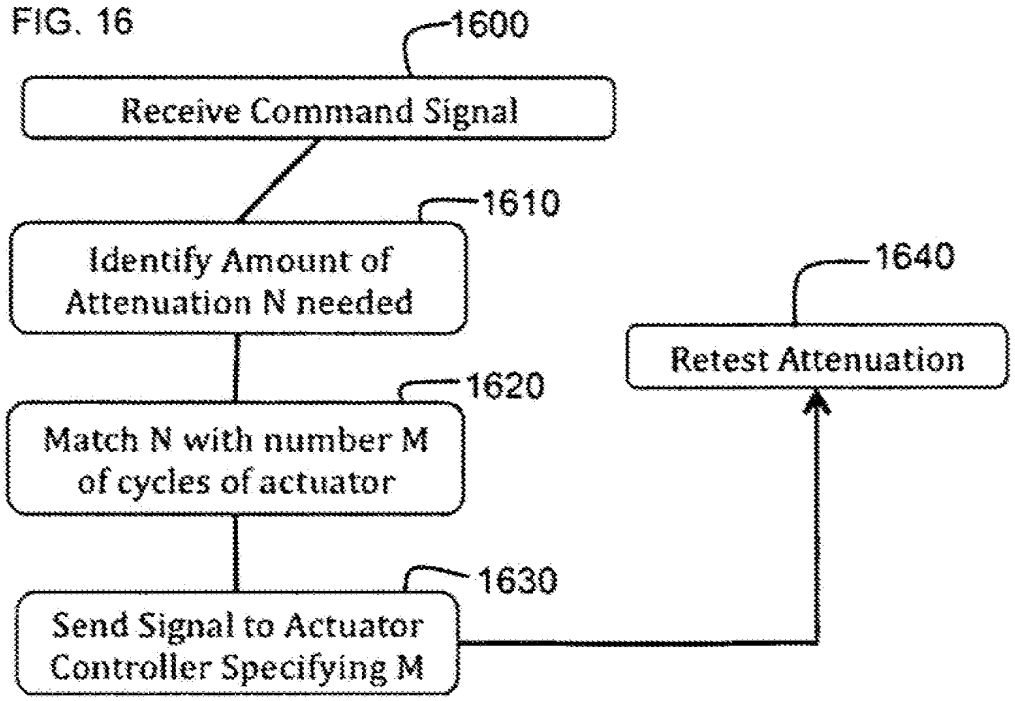


FIG. 16



Electronic Patent Application Fee Transmittal

Application Number:	17182570
Filing Date:	23-Feb-2021
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-16838277

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

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

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Miscellaneous:				
RCE- 1ST REQUEST	2801	1	680	680
		Total in USD ($)		**680**

Electronic Acknowledgement Receipt	
EFS ID:	47058673
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-16838277
Receipt Date:	18-NOV-2022
Filing Date:	23-FEB-2021
Time Stamp:	23:16:45
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$680
RAM confirmation Number	E2022AHN18040556
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	Request for Continued Examination (RCE)	17182570_RCE.pdf	1349948 6bad239e2368bd53477334656d102b878cde3371	no	3
Warnings:					
Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	17182570_RCE-IDS.pdf	1034424 e07475121ad3fc35d42573040651d41b97168d95	no	4
Warnings:					
Information:					
A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
3	Amendment Submitted/Entered with Filing of Continued Prosecution Application (CPA)/Request for Continued Examination(RCE)	17182570_RCE-LTR.pdf	168412 203367740e538cb15fa98f64c4074e7b6748c615	no	6
Warnings:					
Information:					
4	Drawings-other than black and white line drawings	17182570-FIGS-REPLACEMENT.pdf	2822352 5e575ed34dd26dac2ba02c8c89dab8bbfce6fdab	no	15
Warnings:					
Information:					
5	Non Patent Literature	ExParte_Request_10979836.PDF	2466545 7dff27b8c9b00c21db3f8d2c1d0cce7482382b31	no	133
Warnings:					
Information:					

6	Fee Worksheet (SB06)	fee-info.pdf	37777 50d1226088c7ddd27cd547ab76b9d232d3e2f64a	no	2
Warnings:					
Information:					
			Total Files Size (in bytes):	7879458	

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

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New International Application Filed with the USPTO as a Receiving Office

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

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

RESPONSE TO OFFICE ACTION

Commissioner of Patents
Alexandria, VA 22313-1450

DUE DATE: 26 Nov. 2021

Sir:

Applicant appreciates the Examiner's Notice of Allowance dated 16 Nov. 2022, and after a review of the IDS (e.g., 1449) and Figures, the following additional IDS is submitted for the Examiner's consideration, which includes an Exparte Re-Examination filing for the Parent Patent No. 10,979,836, grammatical correction to claims 5 and 15, and Figure replacement sheets 1-15, where in particular Figure 2 reference number 222 should read "Second Earpiece" in keeping with specification ([0045]-[0046]) and as also reflected in Parent Patent 10,979,836. Remarks are respectfully submitted in connection with the above-identified application as is a Request for Continued Examination.

Amendment(s) to the Claims begin on page 2 of this paper.

Amendment(s) to the Drawings begin on page 6 of this paper.

If the Examiner has any questions concerning this application, the Examiner is requested to contact the undersigned at 571-606-1291 in the Washington D.C. Metro area.

Respectfully submitted,
TECHIYA LLC
/John P. Keady/

Date: November 18, 2022

John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci), J.D.
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389

Amendment(s) to the Claim(s):

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

Listing of Claims:

1.(previously presented) An audio device configured to test a seal of an earphone comprising:

a memory that stores instructions; and

a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:

generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30Hz and 200Hz;

sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user;

receiving a first microphone signal from the first microphone;

calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;

comparing the value to a threshold to determine a seal quality between the first side and the second side;

generating a message to convey the seal quality; and

sending the message to the user.

2.(previously presented) The audio device according to claim 1, wherein the message is an acoustic message.

3(previously presented). The audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.

4.(previously presented) The audio device according to claim 1, wherein the speaker plays the test signal from the first side.

5.(currently amended) The audio device according to claim 4, wherein the first microphone measures sound from the first side. ~~signal.~~

6.(previously presented) The audio device according to claim 4, wherein the first microphone measures sound from the second side.

7.(previously presented) The audio device according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.

8.(previously presented) The audio device of claim 1, wherein the first signal is the test signal and the second signal is a modified first microphone signal.

9.(previously presented) The audio device of claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal.

10.(previously presented) The audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.

11.(previously presented) The audio device according to claim 8, wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

12.(previously presented) The audio device according to claim 10, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

13.(previously presented) The audio device according to claim 1, wherein the earphone includes a second microphone.

14.(previously presented) The audio device according to claim 13, wherein the operations further comprise:

receiving a second microphone signal from the second microphone.

15.(currently amended) The audio device according to claim 14, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.

16.(previously presented) The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.

17.(previously presented) The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal.

18.(previously presented) The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.

19.(previously presented) The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof.

20.(previously presented) The audio device of claim 1, wherein the sealing section is an eartip.

Amendments to the Drawings:

The attached sheet of drawings, sheets 1-15, includes changes to Figure 2, where reference 222 should refer to “Second Earpiece” in accordance with referral to specification [0045] and [0046].

Attachment: Replacement Sheets

PTO/SB/80 (11-08)
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POWER OF ATTORNEY TO PROSECUTE APPLICATIONS BEFORE THE USPTO

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

I hereby appoint:

☑ Practitioners associated with the Customer Number: 150139

OR

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

Name	Registration Number	Name	Registration Number

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

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

☑ The address associated with Customer Number: 150139

OR

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

Assignee Name and Address:

Staton Techiya LLC. 16193 Bridlewood Circle, Delray Beach, FL 33445

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

SIGNATURE of Assignee of Record

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

Signature	[signature]	Date	8-14-17
Name	DANIEL C - STATON	Telephone	573-404-0099
Title	MANAGING MEMBER		

This collection of information is required by 37 CFR 1.31, 1.32 and 1.33. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 3 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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PTO/AIA/96 (08-12)
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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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STATEMENT UNDER 37 CFR 3.73(c)

Applicant/Patent Owner: Staton Techiya, LLC

Application No./Patent No.: 17/182,570 Filed/Issue Date: February 23, 2021

Titled: ACOUSTIC SEALING ANALYSIS SYSTEM

Staton Techiya, LLC, a limited liability company

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

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

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

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

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

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

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

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

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

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

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

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

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

1. From: John Usher To: PERSONICS HOLDINGS, LLC

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

   Reel 057620, Frame 0870, or for which a copy thereof is attached.

2. From: John P. Keady To: PERSONICS HOLDINGS, LLC

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

   Reel 057620, Frame 0943, or for which a copy thereof is attached.

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STATEMENT UNDER 37 CFR 3.73(c)

3. From: PERSONICS HOLDINGS, INC. and PERSONICS HOLDINGS, LLC To: DM STATON FAMILY LIMITED PARTNERSHIP

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4. From: DM STATON FAMILY LIMITED PARTNERSHIP To: STATON TECHIYA, LLC

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5. From: _____ To: _____

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6. From: _____ To: _____

The document was recorded in the United States Patent and Trademark Office at
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☐ Additional documents in the chain of title are listed on a supplemental sheet(s).

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

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

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

/Peter A. Chiabotti/	December 15, 2022
Signature	Date
Peter A. Chiabotti	54603
Printed or Typed Name	Title or Registration Number

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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.
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ELECTRONIC ACKNOWLEDGEMENT RECEIPT

APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/182,570	**12/15/2022 04:23:13 PM ET**	**11472-36-3_CONT-16838277**

Title of Invention

ACOUSTIC SEALING ANALYSIS SYSTEM

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	-
CONFIRMATION #	5671	FILED BY	Yoshi Hassan
PATENT CENTER #	61307289	FILING DATE	02/23/2021
CUSTOMER #	150139	FIRST NAMED INVENTOR	John Usher
CORRESPONDENCE ADDRESS	-	AUTHORIZED BY	Peter Chiabotti

Documents

TOTAL DOCUMENTS: 2

DOCUMENT	PAGES	DESCRIPTION	SIZE (KB)
11472-36-3 POA-Executed.PDF	1	Power of Attorney	69 KB
11472-36-3 Statement Under 37 CFR 3_73.pdf	3	Assignee showing of ownership per 37 CFR 3.73	239 KB

Digest

DOCUMENT	MESSAGE DIGEST(SHA-512)
11472-36-3 POA-Executed.PDF	C2CCCA2BE90B29224908557544D64DAEB158FDDBE3B9370 702642BE7F5C5A750A87870A48FFA0DA0A5FD2B8B2798EE50

	AF46F9956498999852D56E3EFBDA46F
11472-36-3 Statement Under 37 CFR 3_73.pdf	02A74B12F6198850A4BE5C34BB3916E93BE8B0B7D625EA76E584D058F8EE1503B6788FF6315F55B65E5367E7E7B954EC5AA6FB4B807F93FE7B83D26C5ABF0130

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150139 7590 12/23/2022

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

EXAMINER
KIM, PAUL

ART UNIT	PAPER NUMBER
2654	

DATE MAILED: 12/23/2022

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277	5671

TITLE OF INVENTION: ACOUSTIC SEALING ANALYSIS SYSTEM

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$600	$0.00	$0.00	$600	03/23/2023

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 1/2 the amount of undiscounted fees, and micro entity fees are 1/2 the amount of small entity fees.

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

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

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

PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), 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 including the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

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

150139 7590 12/23/2022
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.
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277	5671

TITLE OF INVENTION: ACOUSTIC SEALING ANALYSIS SYSTEM

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$600	$0.00	$0.00	$600	03/23/2023

EXAMINER	ART UNIT	CLASS-SUBCLASS
KIM, PAUL	2654	381-059000

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) ❑Advance Order - # of Copies _____

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

❑ Electronic Payment via 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

❑ Applicant asserting small entity status. See 37 CFR 1.27

❑ Applicant changing to regular undiscounted fee status.

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.
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.
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. _____

IPR2024-01015 Page 00223

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.
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277	5671

150139 7590 12/23/2022

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

EXAMINER	
KIM, PAUL	

ART UNIT	PAPER NUMBER
2654	

DATE MAILED: 12/23/2022

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.** 17/182,570	**Applicant(s)** Usher et al.	
	Examiner PAUL KIM	**Art Unit** 2654	**AIA (FITF) Status** Yes

-- 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 11/18/2022.
   ☐ 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 1-20 . As a result of the allowed claim(s), you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see **http://www.uspto.gov/patents/init_events/pph/index.jsp** or send an inquiry to **PPHfeedback@uspto.gov.**

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

   Certified copies:

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

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

   * Certified copies not received: _____ .

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

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

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

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

Attachment(s)

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

/PAUL KIM/
Primary Examiner, Art Unit 2654

DETAILED ACTION

1. The present application, filed on or after March 16, 2013, is being examined under the first inventor to file provisions of the AIA.

Allowable Subject Matter

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

Regarding independent claim 1, the closest prior art of record, Svean et al. (US Pat. 6567524) discloses an audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user.

However, Svean fails to teach the combination of an audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one

component lies between a range of 30Hz and 200Hz; sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination; comparing the value to a threshold to determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user. The distinct features, as disclosed in independent claim 1, render the claims allowable.

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

2. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Ryan (US Pub. 20040196992) discloses a system and method for detecting the insertion or removal of a hearing instrument from the ear canal.

Siltmann (US Pub. 20070019817) discloses a hearing device with automatic determination of its fit in the ear and corresponding method.

Padhi (US Pub. 20050096899) discloses an apparatus, method, and computer program for comparing audio signals.

Oliveira (US Pat. 5002151) discloses an ear piece having disposable compressible polymeric foam sleeve.

Rast (US Pub. 20010046304) discloses a system and method for selective control of acoustic isolation in headset.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to PAUL KIM whose telephone number is (571)270-7697. The examiner can normally be reached on 9 AM - 5 PM, PST.

Examiner interviews are available via telephone, in-person, and video conferencing using a USPTO supplied web-based collaboration tool. To schedule an interview, applicant is encouraged to use the USPTO Automated Interview Request (AIR) at http://www.uspto.gov/interviewpractice.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, VIVIAN CHIN can be reached on (571) 272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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

/PAUL KIM/
Primary Examiner, Art Unit 2654

Notice of References Cited	Application/Control No. 17/182,570	Applicant(s)/Patent Under Reexamination Usher et al.	
	Examiner PAUL KIM	Art Unit 2654	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-6567524-B1	05-2003	Svean; Jarle	A61F11/08	381/317
*	B	US-20040196992-A1	10-2004	Ryan, Jim G.	H04R1/10	381/317
*	C	US-20070019817-A1	01-2007	Siltmann; Simone	H04R25/305	381/60
*	D	US-20050096899-A1	05-2005	Padhi, Kabi P.	G10L15/10	704/216
*	E	US-5002151-A	03-1991	Oliveira; Robert J.	H04R25/656	381/328
*	F	US-20010046304-A1	11-2001	Rast, Rodger H.	H04R1/1083	381/26
	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.

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

CPC								
Symbol							**Type**	**Version**
H04R	/	29	/	001			F	2013-01-01
H04R	/	25	/	70			I	2013-01-01
H04R	/	1	/	1016			I	2013-01-01
H04R	/	1	/	1091			I	2013-01-01
H04R	/	29	/	00			I	2013-01-01
H04R	/	2460	/	15			A	2013-01-01
H04R	/	2460	/	07			A	2013-01-01

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

NONE		Total Claims Allowed:	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	13 December 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office — Part of Paper No.: 20221213

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

INTERNATIONAL CLASSIFICATION					
CLAIMED					
H04R29/00	/	29	/	00	
H04R1/10	/	1	/	10	
NON-CLAIMED					
H04R25/00	/	25	/	00	

US ORIGINAL CLASSIFICATION	
CLASS	**SUBCLASS**
381	59

CROSS REFERENCES(S)						
CLASS	**SUBCLASS (ONE SUBCLASS PER BLOCK)**					
381	60	58				

NONE		**Total Claims Allowed:**	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	13 December 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office — Part of Paper No.: 20221213

Issue Classification	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	**Art Unit**
	PAUL KIM	2654

☐ 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	11	10	19	19										
2	2	9	11	20	20										
3	3	12	12												
4	4	13	13												
5	5	14	14												
6	6	15	15												
7	7	16	16												
8	8	17	17												
10	9	18	18												

NONE		**Total Claims Allowed:**	
(Assistant Examiner)	(Date)	20	
/PAUL KIM/ Primary Examiner, Art Unit 2654	13 December 2022	O.G. Print Claim(s)	O.G. Print Figure
(Primary Examiner)	(Date)	1	1,3

U.S. Patent and Trademark Office Part of Paper No.: 20221213

Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	Art Unit
	PAUL KIM	2654

CPC - Searched*		
Symbol	Date	Examiner
H04R25/70 OR H04R29/001 OR H04R29/00 OR H04R2460/15 OR H04R1/1016 OR H04R1/1091 OR H04R2460/07	11/05/2022	PK

CPC Combination Sets - Searched*		
Symbol	Date	Examiner

US Classification - Searched*			
Class	Subclass	Date	Examiner
381	58, 59, 60	11/05/2022	PK

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

/PAUL KIM/ Primary Examiner, Art Unit 2654	

Search Notes	Application/Control No.	Applicant(s)/Patent Under Reexamination
	17/182,570	Usher et al.
	Examiner	Art Unit
	PAUL KIM	2654

Search Notes		
Search Notes	**Date**	**Examiner**
Inventor search - John Usher, John Keady	11/05/2022	PK
PE2E search - see PE2E search note	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 01/14/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 05/18/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 06/14/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 06/23/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 07/1/2022	11/05/2022	PK
IPR2022-00410 Reviewed Petition for Inter Partes Review of U.S. Patent No. 10979836 on 08/12/2022	11/05/2022	PK
Updated search - see PE2E search note	12/13/2022	PK

Interference Search			
US Class/CPC Symbol	**US Subclass/CPC Group**	**Date**	**Examiner**
	See attached search history	11/05/2022	PK

/PAUL KIM/ Primary Examiner, Art Unit 2654	

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

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(Not for submission under 37 CFR 1.99)

Application Number	17182570
Filing Date	2021-02-23
First Named Inventor	John Usher
Art Unit	2654
Examiner Name	KIM, PAUL
Attorney Docket Number	CONT-16838277

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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
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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]

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

	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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EXAMINER SIGNATURE

Examiner Signature	/PAUL KIM/	Date Considered	12/13/2022

*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.

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	2654
	Examiner Name	KIM, PAUL
	Attorney Docket Number	CONT-16838277

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-11-18
Name/Print	John P. Keady	Registration Number	56389

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.**

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

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

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

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.

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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 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.

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

Bibliographic Data

Application No: 17/182,570

Foreign Priority claimed: ○ Yes ◉ No

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

Verified and Acknowledged: /PAUL KIM/ Examiner's Signature | Initials

Title: ACOUSTIC SEALING ANALYSIS SYSTEM

FILING or 371(c) DATE	CLASS	GROUP ART UNIT	ATTORNEY DOCKET NO.
02/23/2021	381	2654	CONT-16838277
RULE			

APPLICANTS

Staton Techiya LLC, Lighthouse Point, FL, UNITED STATES

INVENTORS

John Usher, Devon, UNITED KINGDOM

John P. Keady, Fairfax Station, VA, UNITED STATES

CONTINUING DATA

This application is a CON of 16838277 04/02/2020 PAT 10979836

16838277 is a CON of 16414136 05/16/2019 PAT 10701499

16414136 is a CON of 15700511 09/11/2017 PAT 10299053

15700511 is a CON of 14827332 08/17/2015 PAT 9781530

14827332 is a CON of 14054015 10/15/2013 PAT 9113267

14054015 is a DIV of 12555864 09/09/2009 PAT 8600067

12555864 has PRO of 61098250 09/19/2008

FOREIGN APPLICATIONS

IF REQUIRED, FOREIGN LICENSE GRANTED**

03/04/2021

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UNITED KINGDOM

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

UNITED STATES

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PE2E SEARCH - Search History (Prior Art)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
L1	1	"17182570"	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 02:30 AM
L2	12	"12555864"	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 02:30 AM
L3	117	((("Staton") near3 ("Techiya") near3 ("LLC"))).AS,AANM.	(USPAT)	OR	ON	ON	2022/11/05 02:31 AM
L4	289	((("Usher") near3 ("John"))).INV.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:32 AM
L5	149	((("Keady") near3 ("John") near3 ("P"))).INV.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:32 AM
L6	245	((("Staton") near3 ("Techiya") near3 ("LLC"))).AS,AANM.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:33 AM
L7	917	(381/59).ccls.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:34 AM
L8	3298	H04R29/001	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L9	5965	H04R1/1016	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L10	2719	H04R1/1091	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L12	2674	H04R25/70	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:35 AM
L13	8905	H04R29/00	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:36 AM
L14	892	H04R2460/07	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:36 AM
L15	496	H04R2460/15	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:36 AM
L16	19612	L8 OR L9 OR L10 OR L12 OR L13 OR L14 OR l15	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:37 AM
L17	13	L8 AND L9 AND L10 AND L12 AND L13 AND L14 AND L15	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:49 AM
L18	652	L7 AND L16	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 02:51 AM
L19	1	"20060188105"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 04:47 AM
L20	128	"6567524"	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 04:48 AM
L21	2764	(381/58,59,60).ccls.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:00 PM

L22	42	("5142587"|"5764781"|"8774435"|"20050078838"|"5267321"|"6021207"|"7072482"|"10860287"|"7801318"|"8971561"|"9293128"|"7477756"|"20140247948"|"4020297"|"5319163"|"7688983"|"8550206"|"20080037801"|"6567524"|"6728385"|"6754359"|"20130177179"|"5138722"|"5712453"|"8774433"|"6321070"|"8401200"|"6415034"|"7450730"|"8014553"|"20140247952"|"20060204014"|"4088849"|"7778434"|"RE38351"|"20180176681"|"9313572"|"6359993"|"9084060"|"20030165246"|"5524056"|"7209569").PN.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:01 PM
L23	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) SAME (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) SAME (microphone$2 WITH acoustic WITH barrier$2) SAME (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) SAME (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:08 PM
L24	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:09 PM

		earbud$2))) AND (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) AND (microphone$2 WITH acoustic WITH barrier$2) AND (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) AND (compar$4 WITH threshold WITH qualit$4)					
L25	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2) AND (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:10 PM
L26	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:10 PM
L27	358040	((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5) OR (headset$2 OR earphone$2 OR headphone$2 OR	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:11 PM

		earpiece$2 OR earbud$2))					
L28	1606	L27 AND (test$4 WITH seal$3)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:12 PM
L29	422	L28 AND L16	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:13 PM
L30	65	L29 AND L21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:13 PM
L31	119	L29 AND (test$3 near3 signal$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:15 PM
L32	25	L31 AND I21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:15 PM
L33	2215	L27 AND ((test$4 OR analy$3) WITH seal$3)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:16 PM
L34	706	I33 AND (test$3 near3 signal$2)	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L35	123	L34 AND L16	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L36	27	L34 AND L21	(US-PGPUB; USPAT; USOCR; EPO; DERWENT)	OR	ON	ON	2022/11/05 05:17 PM
L37	44	(Artificial Intelligence) Similar to: 16/838,277 with 0 CPC Selections and 0 Text Selections Text: (US-10986432-B2 OR EP-3646614-A1 OR WO-2019005492-A1 OR US-20190007762-A1 OR EP-2737720-A2 OR US-20220256270-A1 OR EP-2737720-B1 OR US-20200068283-A1 OR US-10972823-B2 OR WO-2013016336-A2 OR US-20210022996-A1 OR US-10462549-B2 OR US-8737669-B2 OR FR-3062014-A1 OR KR-102213388-B1 OR US-7837005-B2 OR CN-113491504-A OR JP-6037483-B2 OR KR-	(US-PGPUB; USPAT; USOCR; FIT (AP, AT, AU, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DK, EA, EE, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, RU, SE, SG, SI, SK, SU, TH, TN, TR, TW, UA, VN, WO))	OR	ON	ON	2022/11/05 05:19 PM

		102336438-B1 OR JP-2016048931-A OR US-11159876-B2 OR CN-209184754-U OR CN-209184762-U OR JP-2021150949-A OR JP-5934358-B2 OR JP-2014524219-A OR JP-2014087054-A OR US-20210297765-A1 OR US-20130230204-A1 OR EP-3852386-A1 OR JP-6306273-B2 OR JP-2018502525-A OR CN-112703747-B OR US-20200025732-A1 OR CN-107925810-B OR US-9351064-B2 OR US-20210007669-A1 OR US-20200016006-A1 OR EP-3522842-A1 OR WO-2018065255-A1 OR EP-3522842-B1 OR US-20190387305-A1 OR US-20070240931-A1 OR US-11381897-B2 OR US-20220070572-A1 OR US-20090041285-A1 OR US-20140177899-A1 OR AU-2016209621-A1 OR CN-209402688-U OR US-20170360614-A1).did.					
L38	33	(Artificial Intelligence) Similar to: 17/182,570 with 0 CPC Selections and 0 Text Selections Text: (US-20210211797-A1 OR US-5699435-A OR EP-2894879-A3 OR EP-2858383-A3 OR EP-2884769-A3 OR EP-2884768-A3 OR EP-2879406-A3 OR EP-3120578-B2 OR EP-3120578-B1 OR US-20160112814-A1 OR US-20150271607-A1 OR WO-2015143151-A1 OR EP-3120578-A1 OR CN-106465025-A OR CN-106465025-B OR	(US-PGPUB; USPAT; USOCR; FIT (AP, AT, AU, BE, BG, BR, BY, CA, CH, CN, CS, CU, CZ, DD, DK, EA, EE, ES, FI, FR, GB, HR, HU, ID, IE, IL, IS, IT, JP, KR, LT, LU, LV, MA, MC, MD, MY, NL, NO, NZ, PH, PL, PT, RO, RS, RU, SE, SG, SI, SK, SU, TH, TN, TR, TW, UA, VN, WO))	OR	ON	ON	2022/11/05 05:19 PM

		CN-106331972-A OR CN-106331972-B OR US-10448177-B2 OR EP-3113519-A1 OR US-11438716-B1 OR US-11218817-B1 OR EP-3503573-A1 OR US-20150271608-A1 OR EP-1416764-A3 OR US-20210185466-A1 OR US-9648430-B2 OR US-10945086-B2 OR CA-2215764-C OR US-20190191254-A1 OR US-20200045484-A1 OR CN-110035344-A OR US-11019434-B2 OR JP-2019146152-A OR CH-708901-A2 OR WO-2015069416-A1 OR CN-1627865-A OR US-20170195807-A1 OR WO-2012093343-A3 OR DE-4418203-A1 OR US-20030021434-A1 OR EP-3849212-A1 OR EP-2547120-A1 OR US-10645505-B2 OR EP-1416764-A2 OR US-20150124977-A1 OR EP-1416764-B1 OR CN-113099334-B OR CH-690220-A5 OR CN-113099334-A OR US-11197081-B2).did.					
L39	39	(Artificial Intelligence) More like doc: US-6567524-B1 Text: (WO-0217838-A1 OR EP-1313419-A1 OR EP-1313419-B1 OR CA-2418027-C OR CA-2418027-A1 OR WO-0217839-A1 OR EP-1322268-A1 OR EP-1322268-B1 OR CA-2418031-C OR WO-0217836-A1 OR EP-1313417-A1 OR EP-1313417-B1 OR ES-2279829-T3 OR CA-2418010-C OR WO-0217837-A1 OR EP-1313418-A1 OR WO-0217835-A1 OR US-6661901-B1 OR US-	(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	2022/11/05 05:20 PM

		7039195-B1 OR EP-1313418-B1).did.					
L40	118	("4049930" | "4489440" | "4548082" | "4985925" | "5201006" | "5276739" | "5426719" | "5680466").pn. OR ("6567524").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:21 PM
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L42	5	("2006/0188105").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/11/05 05:22 PM
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L58	38	(Artificial Intelligence) More like doc:	(US-PGPUB; USPAT; USOCR; FIT (AU, AP,	OR	ON	ON	2022/12/13 08:07 PM

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L61	167	("2001/0046304").urpn. AND (PGPB | USPT | USOC).dbnm.	(US-PGPUB; USPAT; USOCR)	OR	ON	ON	2022/12/13 08:08 PM

PE2E SEARCH - Search History (Interference)

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	British Equivalents	Time Stamp
N1	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) SAME (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) SAME (microphone$2 WITH acoustic WITH barrier$2) SAME (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) SAME (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:08 PM
N2	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2 WITH multiple$2 WITH frequenc$5) AND (microphone$2 WITH acoustic WITH barrier$2) AND (calculat$4 WITH value$2 WITH (coherence OR correlation OR difference$2 OR cross-correlation)) AND (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:09 PM

N3	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2) AND (compar$4 WITH threshold WITH qualit$4)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:10 PM
N4	0	(test$4 WITH seal$3 WITH (((hearing near1 aid$2) OR (hearing-aid) OR (hearing near2 assist$5)) OR (headset$2 OR earphone$2 OR headphone$2 OR earpiece$2 OR earbud$2))) AND (generat$4 WITH test WITH signal$2) AND (microphone$2 WITH acoustic WITH barrier$2)	(US-PGPUB; USPAT)	OR	ON	ON	2022/11/05 05:10 PM

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277

CONFIRMATION NO. 5671

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

POA ACCEPTANCE LETTER

OC00000137069763

Date Mailed: 12/28/2022

NOTICE OF ACCEPTANCE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 12/15/2022.

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

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

/qnguyen/

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

Amendment After Allowance Under 37 CFR 1.312

Commissioner of Patents
Alexandria, VA 22313-1450

DATE: 30 Dec. 2022

The Office issued a notice of allowance 23 Dec. 2022. Applicants submit the attached marked-up and clean version of the specification to address typographical errors, in particular in paragraphs [0079] and [0115].

The minor 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. However, 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.

Respectfully submitted,
TECHIYA LLC

Date: December 30, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci), J.D.
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

Electronic Acknowledgement Receipt

EFS ID:	47298370
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	11472-36-3_CONT-16838277
Receipt Date:	30-DEC-2022
Filing Date:	23-FEB-2021
Time Stamp:	12:04:32
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Applicant Arguments/Remarks Made in an Amendment	CONT-17182570-SPEC-MARKED.pdf	395404 9626b89e3b6ac48bca9d58169c780cce6d5f21be	no	25

Warnings:

Information:					
2	Specification	CONT-17182570-SPEC-CLEAN.pdf	385532 4622978cd82c041d61e9cd18d0254f81aee0f944	no	25
Warnings:					
Information:					
3	Amendment after Notice of Allowance (Rule 312)	17182570_312-amndt.pdf	115954 43b458623a1d68213553706a53451af17d21d73c	no	1
Warnings:					
Information:					
			Total Files Size (in bytes):	896890	

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.

ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of U.S. Patent Application Ser. No. 17/182,570, filed 23 Feb. 2021, which is a continuation of U.S. Patent Application Ser. No. 16/838,277, filed April 2, 2020, which is a continuation of U.S. Patent Application Ser. No. 16/414,136, filed May 16, 2019, which is a continuation of U.S. Patent Application Ser. No. 15/700,511, filed September 11, 2017, which is a continuation of U.S. Patent Application Ser. No. 14/827,332, filed August 17, 2015, now U.S. Patent No. 9,781,530, which is a continuation of U.S. Patent Application Ser. No. 14/054,015, filed October 15, 2013, now U.S. Patent No. 9,113,267, which is a Divisional Application of U.S. Application Ser. No. 12/555,864, filed September 9, 2009, now U.S. Patent No. 8,600,067 and claims the benefit of U.S. Provisional Patent Application No. 61/098,250 filed September 19, 2008. The disclosure of all the aforementioned references are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to testing the seal of an orifice-inserted device, and more particularly, though not exclusively, to a device and method for determining if an earpiece is sealed correctly in an ear canal.

BACKGROUND OF THE INVENTION

[0003] It can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

[0004] Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient

sound to provide adequate performance to the user. Additionally, user conditions may change substantially during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

[0005] Broadly stated, embodiments are directed to a device and method to determine if an earpiece is sealing within the design specification of the device.

[0006] In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume. The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

[0007] At least one exemplary embodiment is directed to a method of detecting sealing integrity of an earpiece comprising the steps of: providing a test signal; generating an acoustic signal corresponding to the test signal incident on an ear canal side of a sealing section; converting the acoustic signal incident on a first side of the sealing section to an electrical signal; and cross- correlating the test signal to the electrical signal where the earpiece is sealed correctly when a cross-correlation between the test signal and the electrical signal is above a threshold.

[0008] At least one exemplary embodiment is directed to a method of adjusting attenuation of an earpiece comprising the steps of: relating cross-correlation of a test signal and a measured acoustic signal in an ear canal of a user to an attenuation level of a sealing section of the earpiece; comparing the attenuation level of the sealing section of the earpiece to a minimum attenuation value; and adjusting a pressure of the sealing section to meet the minimum attenuation value.

[0009] At least one exemplary embodiment is directed to a device comprising: a sealing section configured to seal a user's orifice, where the sealing section is configured to produce an acoustic seal between a first side of the sealing section and a second side of the sealing section; a transducer configured to generate a first acoustic signal incident on the first side of the sealing section; and a first microphone configured to measure a second acoustic signal incident on the second side of the sealing section, where the second acoustic signal

includes at least a portion of the first acoustic signal that has passed from the first side to the second side of the sealing section where the first acoustic signal is compared to the second acoustic signal to determine if the sealing section is sealed.

[0010] Further areas of applicability of exemplary embodiments of the present invention will become apparent from the detailed description provided hereinafter. It should be understood that the detailed description and specific examples, while indicating exemplary embodiments of the invention, are intended for purposes of illustration only and are not intended to limit the scope of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0012] FIG. 1 is a diagram of an earpiece inserted in an ear canal in accordance with an exemplary embodiment;

[0013] FIG. 2 is a block diagram of optional components of an earpiece in accordance with an exemplary embodiment;

[0014] FIG. 3 is a flowchart of a method for checking an ear seal in accordance with an exemplary embodiment;

[0015] FIG. 4 is a flowchart to determine acoustic seal integrity of an earpiece in accordance with the exemplary embodiment.

[0016] FIG. 5 is a flowchart of a method to estimate the instantaneous cross-correlation between a first and second audio signal.

[0017] FIG. 6 is a flowchart to determine when to emit a test signal in accordance with an exemplary embodiment;

[0018] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention;

[0019] FIG. 8 is a block diagram for a method of adjusting the IMS system in accordance with at least one exemplary embodiment;

[0020] FIG. 9 is a block diagram for a method of adjusting IMS pressure in accordance with at least one exemplary embodiment;

[0021] FIG. 10 illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with at least one exemplary embodiment;

[0022] FIG. 11 illustrates a flowchart of an exemplary method to determine a test signal fundamental;

[0023] FIG. 12 illustrates a flowchart of an exemplary embodiment to determine tonal presence in audio content;

[0024] FIG. 13 illustrates a flowchart of a method to determine when to emit the test signal;

[0025] FIG. 14 illustrates a flowchart of an exemplary method to determine acoustic seal integrity;

[0026] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment; and

[0027] FIG. 16 illustrates the sending of a signal to an inflation controller upon detection of a seal fail to modify the seal pressure in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION

[0028] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0029] Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

[0030] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0031] Additionally exemplary embodiments are not limited to earpieces, for example some functionality can be implemented on other systems with speakers and/or microphones for example computer systems, PDAs, BlackBerry® smart phones, cell and mobile phones, and any other device that emits or measures acoustic energy. Additionally, exemplary embodiments can be used with digital and non-digital acoustic

systems. Additionally various receivers and microphones can be used, for example MEMs transducers, diaphragm transducers, for example Knowles' FG and EG series transducers.

[0032] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0033] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0034] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for sealing or partially sealing an ear. FIG. 1 is a diagram of an earpiece inserted in an ear canal 124 in accordance with at least one exemplary embodiment of the invention. FIG. 1 also illustrates portions of the ear including pinna 128, ear canal 124 and eardrum 126. As illustrated, the earpiece comprises an electronic housing unit 100 and a sealing unit 108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 124 of a user 130. The earpiece is an in-ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude the ear canal 124.

[0036] The earpiece includes an Ambient Sound Microphone (ASM) 120 to capture ambient sound, an Ear Canal Receiver (ECR) 114 to deliver audio to an ear canal124, and an Ear Canal Microphone (ECM) 106 to capture and assess a sound exposure level within the ear canal 124. The earpiece can partially or fully occlude the ear canal 124 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 124, and to form an acoustic seal with the walls of the ear canal 124 at a location between the entrance to the ear canal 124 and the tympanic membrane (or ear drum) 126. In general, such a seal is typically achieved by means of a soft and compliant housing of the sealing unit 108. Additionally, the sealing unit 108 can be a pressurized expandable element that fills a portion of the available local space.

[0037] Sealing unit 108 is an acoustic barrier having a first side corresponding to ear canal124 and a second

side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 108 includes an ear canal microphone tube 110 and an ear canal receiver tube 112. Sealing unit 108 creates a closed cavity of approximately 5 cc or less between the first side of sealing unit 108 and the tympanic membrane 126 in ear canal124. In at least one exemplary embodiment the sealing facilitates using the ECR (speaker) 114 to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 126 resulting from the sound field at the entrance to the ear canal 124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0038] In at least one exemplary embodiment and in broader context, the second side of sealing unit 108 corresponds to the side adjacent to electronic housing unit 100. Ambient sound microphone 120 is housed in electronic housing unit 100 and is exposed to the ambient environment for receiving sound from the ambient environment around the user.

[0039] The electronic housing unit 100 can include various system components such as a microprocessor 116, memory 104, battery 102, ECM 106, ASM 120, ECR, 114, and user interface 122, or these components can reside in a separate system or interface operatively connected. Microprocessor 116 (or processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 116 is operatively coupled to memory 104, ECM 106, ASM 120, ECR 114, and user interface 122. An optional wire 118 can provide an external connection to the earpiece. Battery 102 powers the circuits and transducers of the earpiece. Battery 102 can be a rechargeable or replaceable battery.

[0040] In at least one exemplary embodiment, electronic housing unit 100 is adjacent to sealing unit 108. Openings in electronic housing unit 100 receive ECM tube 110 and ECR tube 112 to respectively couple to ECM 106 and ECR 114. ECR tube 112 and ECM tube 110 acoustically couple signals to and from ear canal 124. For example, ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124 where it is received by the tympanic membrane 126 of the user of the earpiece. Conversely, ECM 106 receives an acoustic signal present in ear canal124 though ECM tube 110.

[0041] One function of ECM 106 is that of measuring the sound pressure level in the ear canal cavity 124 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece. In one arrangement, ASM 120 is used to monitor sound pressure at the entrance to the occluded or partially occluded ear canal 124. All transducers shown can receive or transmit

audio signals to a processor 116 that undertakes audio signal processing and provides a transceiver for audio via the wired (wire 118) or a wireless communication path. Note also that the acoustic signals can be stored for later retrieval.

[0042] In at least one exemplary embodiment the earpiece can be constructed to actively monitor a sound pressure level both inside and outside an ear canal 124. In at least one exemplary embodiment monitored data can be used to enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. In at least one exemplary embodiment an earpiece can facilitate at least one of conducting listening tests, filtering sounds in the environment, monitoring warning sounds in the environment, presenting notification based on identified warning sounds, maintaining constant audio content to ambient sound levels, and filtering sound in accordance with a Personalized Hearing Level (PHL).

[0043] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 124 using ECR 114 and ECM 106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 120. For instance, the ECR 114 can deliver an impulse within the ear canal124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. In at least one exemplary embodiment the earpiece can use either the ASM 120 or the ECM 106 to monitor the sound pressure level, which can then be used in a Sound Pressure Level Dosimeter calculation, to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0044] Referring to FIG. 2, a block diagram of an earpiece 201 in accordance with an exemplary embodiment is shown. A power supply 205 (e.g., USB power connection, hearing aid battery (batteries)) powers components of the earpiece 201 including microprocessor 206 (or processor 206, e.g., Texas Instruments TMS320C6713) and a data communication system 216 (e.g., RF or Bluetooth communication chip). As illustrated, the earpiece 201 includes the processor 206 operatively coupled to data communication system 216, ASM 210, ECR 212, and ECM 208. Data communication system 216 may include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 206. The processor 206 can also include a clock to record a time stamp.

[0045] In general, data communication system 216 is a communication pathway to components of the earpiece 201 and components external to the earpiece 201. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system 216 is configured to communicate with ECM 208, ASM 210, visual display 218, and user control interface 214 of the earpiece 201. As shown, user control interface 214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 216 is capable of communication to devices exterior to the earpiece 201 such as the user's mobile phone 234, a second earpiece 222, and a portable media player 228. Portable media player 228 can be controlled by a manual user control 230.

[0046] The user's mobile phone 234 includes a mobile phone communication system 224. A microprocessor 226 is operatively coupled to mobile phone communication system 224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 220 worn by another person to the user's mobile phone. Similarly, the user's mobile phone 234 can be connected to the data communication system 216 of the earpiece 201 as well as the second earpiece 222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 234 through their respective earpieces.

[0047] As illustrated, a data communication system 216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

[0048] The RAM 202 stores program instructions for execution on the processor 206 as well as captured audio processing data. For instance, memory RAM 202 and ROM 204 can be off-chip and external to the processor 206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. In at least one exemplary embodiment, the data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer is operatively connected with processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0049] Data communication system 216 includes an audio interface operatively coupled to the processor

206 and the VOX to receive audio content, for example from portable media player 228, a cell phone, or any other communication device, and deliver the audio content to the processor 206. The processor 206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 206 (or the VOX of data communication system 216) can lower a volume of the audio content responsive to detecting an event for transmitting the acute sound to the ear canal of the user. The processor 206 by way of the ECM 208 can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 216.

[0050] The earpiece 201 and data communication system 216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading over-the-air to the earpiece 201. It should be noted also that next generation access technologies can also be used in exemplary embodiments.

[0051] Data communication system 216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 201.

[0052] The power supply 205 utilizes common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 201 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 205 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0053] Microprocessor 206 is operatively connected with an EarSeal Inflation Management System 232 to control the degree to which the sealing unit 108 is inflated or deflated. In one exemplary embodiment, sealing unit 108 comprises an expandable element (e.g., inflatable balloon mechanism), whereby a cavity can be filled with air or a liquid to change the degree of acoustic isolation between the internal ear canal space 124 and the ambient environment. Alternately, a passive system for sealing ear canal 124 is used such as a flexible rubber or a silicon sealing unit or a foam plug. In one exemplary embodiment, the passive

system is a balloon mechanism that is filled with air or liquid. The balloon mechanism conforms to the shape and size of an ear canal and includes a restorative force module that applies a pressure to the balloon mechanism for sealing the ear canal cavity.

[0054] The earpiece is a single operational device or a family of devices configured in a master- slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece are reused in different form factors for the master and slave devices.

[0055] Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user (second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

[0056] The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

[0057] In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

[0058] The degree of sealing between the first and second acoustic volumes is determined in a step 310 and the process ends at step 312. The cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or "leaky") if the cross- correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly or incorrectly.

[0059] Referring to FIG. 1, the earpiece is used as an example to illustrate a test sequence as disclosed in FIG. 3. Sealing unit 108 occludes an opening of ear canal 124 creating a first volume (ear canal124) and a second volume (the ambient environment). Sealing unit 108 has a first side exposed to ear canal 124 and a second side is exposed to or corresponds to the ambient environment external to the ear.

[0060] In at least one exemplary embodiment, processor 116 is configured to receive a test signal in memory 104. Processor 116 generates the test signal and provides the test signal to Ear Canal Receiver 114 (ECR 114). ECR 114 emits the test signal into Ear Canal Receiver Tube (ECR Tube 112). The test signal propagates through ECR tube 112 and into ear canal 124. Ear Canal Microphone tube 110 (ECM tube 110) is configured to receive an acoustic signal incident on the first side of sealing unit 108. The test signal in ear canal 124 propagates through ECM tube 110 and is received by Ear Canal Microphone 106 (ECM 106). ECM 106 is configured to measure the test signal in ear canal124 and provide the measured test signal to processor 116.

[0061] As shown, electronic housing unit 100 of the earpiece is adjacent to the second side of sealing unit 108. Electronic housing unit 100 is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit 108. Electronic housing unit 100 includes Ambient Sound Microphone 120 (ASM 120), which is configured to measure sounds in the ambient environment. Thus, ASM 120 receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit 108. ASM 120 provides the measured ambient signal to processor 116.

[0062] Ideally, sealing unit 108 is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit 108 and into the ambient environment. Conversely, sealing unit 108 if improperly sealed will pass some of the test signal. Processor 116 compares the test signal to the signal

provided by ECM 106 corresponding to the acoustic signal in ear canal124. In particular, processor 116 undertakes the cross-correlation between emitted test signal and the ECM signal.

[0063] Referring to FIG. 4, a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0064] In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively--input) of the test signal data buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time- align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

[0065] A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively--input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10Hz greater than the test signal frequency.

[0066] A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5. The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

[0067] Referring to FIG. 5, a flowchart of an exemplary embodiment to determine the instantaneous cross-correlation between a first audio signal and a second audio signal is illustrated. The process begins at step 500. In at least one exemplary embodiment, the first audio signal is the test signal (i.e. a sine wave) and the second signal is the low-pass-filtered ECM signal.

[0068] The correlation between two signals x and y at time k using an exponential window is defined as:

[0069] $$p(k) = \frac{S_{xy}(k)}{\sqrt{S_{xx}(k)_{yy}(k)}} \quad (1)$$

[0070] Where

[0071] $$S_{xx}(k) = \sum_{l=0}^{\infty} ce^{-\eta^l} x_{k-l} y_{k-l} \quad (2)$$

$$c = 1 - e^{-\eta}$$

[0072] And S_{xx} and S_{yy} are defined similarly as in (2) (replacing y with x for S_{xx} etc.).

[0073] It can be shown (see Aarts et al, 2001) that (1) can be approximated with the recursion:

[0074] $$\hat{\rho}(k) = \hat{\rho}(k-1) + \gamma[\partial_k - \beta_k \hat{\rho}(k-1)] \quad (3)$$

$$\partial_k = 2x_k y_k$$

$$\beta_k = \alpha x_k^2 \alpha^{-1} y_k^2$$

[0075] Where

$$\alpha = \frac{y_{RMS}}{x_{RMS}} \quad (4)$$

$$\gamma = \frac{ce^{\eta}}{2x_{RMS} y_{RMS}}$$

[0077] The cross-correlation estimate using the above recursion is modified for block-wise processing rather than the sample-by-sample basis. This modification replaces the sample values (i.e. x(k) and y(k)) with values for the N-length block mean, i.e.

[0078]

$$x(k) = \frac{1}{N} \sum_{l=0}^{N-1} x(k-l)$$

[0079] Furthermore, the numerator for y is replaced with a small constant and so is [[a]] α (replacing [[a]] α with a constant effectively un-normalizes the correlation estimate). It is found that the modified un-normalized block-wise cross-correlation accurately estimates the cross-correlation compared with using the standard cross-correlation for two signals.

[0080] The modified block-wise fast cross-correlation algorithm, as summarized in FIG. 5, comprises the following steps:

1. A first signal buffer 502 is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated 504 (x_{RMS})
3. The mean level of the first buffer is calculated 506.
4. A second signal buffer 508 is accumulated. This signal buffer corresponds to the filtered ECM signal.
5. The RMS level of the second buffer is calculated 510 (y_{RMS}).
6. The mean level of the second buffer is calculated 512.
7. In step 514, γ (gamma) is approximated as:

$$\gamma = \frac{\Gamma}{2x_{RMS}y_{RMS}}$$

Where Γ is a small constant, e.g. 10E-3.

8. In step 516,∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.
9. In step 518, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.
10. In step 520, the new temporary estimate of the correlation newRho_temp is calculated as: newRho_temp=(delta-beta*rho_old))
11. In step 522, the new estimate of the correlation newRho is updated by summing the previous estimate of the correlation with the product of gamma and the temporary estimate of the correlation newRho_temp.
12. In step 524, the "old" value of the correlation is set to the newest correlation estimate, ready for the next iteration of the update algorithm.
13. In step 526, the current correlation estimate between the emitted test signal and the received and filtered ECM signal is set as equal to the value of newRho.

[0081] Referring to FIG. 6, a flowchart of a method to determine when to emit the test signal is shown. The test signal is emitted when the test can be performed unobtrusively to the user and also provides an accurate test. In at least one exemplary embodiment, a test event to determine if an earpiece is sealed correctly is

initiated via a timing methodology. In a first timing scenario, the test event occurs after a delay of a first predetermined time period when the RMS of the Audio Content (AC) is less than a RMS threshold. In a second timing scenario, the delay of the first predetermined time period is allowed to lapse without the test event occurring when the RMS of the audio content is greater than the RMS threshold. A second predetermined time period is started where the test event occurs when the RMS of the audio content is less than the RMS threshold. The test event is then initiated when the second predetermined time period is exceeded independent of the RMS of the audio content.

[0082] A test sequence is initiated in a step 602. The previous seal test event resets the first digital timer in a step 604. A time delay is generated by the loop comprising steps 606 and 608. The first digital timer is time incremented in the step 606. After each added time increment, the first digital timer is compared against a digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step 608 if the first digital timer is less than the digital_timer_threshold1.

[0083] A second digital timer is reset in a step 610 when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step 612. Audio content (AC) from a signal buffer is retrieved in a step 614. The audio content can be filtered through a low pass filter in an optional step 616. The RMS of the audio content is calculated in a step 618. The calculated RMS of the audio content is compared against an RMS_threshold 622 in a step 620. The second digital timer is compared against a digital_timer_threshold2 in a step 624 if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of 624.

[0084] The audio content signal is mixed with the test signal when the RMS of the audio content is less than the RMS_threshold in a step 626. Also, the audio content signal is mixed with the test signal when the second digital timer is greater than digital_timer_threshold2 in the step 624. The modified audio signal (having the test signal mixed in) is emitted by the ECR in a step 628 for testing the sealing section of the earpiece. The first digital timer is then reset in the step 604 to begin a timing sequence for another sealing section test.

[0085] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention. The estimated un-normalized cross-correlation between the ECM signal and the test signal (i.e. sine wave) is shown for different sine wave frequencies from 30-80Hz. Three different curves are provided corresponding

to a good fit (i.e. a tight optimal seal providing approximately 20-30 dB of acoustic attenuation), mid or partial seal (i.e. an ear-seal that could be characterized as "half in" providing approximately 10-15 dB of acoustic attenuation), and a poor seal (i.e. an ear-seal providing less than 10 dB of acoustic attenuation). At lower test frequencies, the change in correlation is more pronounced as the degree of ear seal fitting is changed from "good" to "mid" and "poor". From the data, the threshold used to determine whether the ear seal can be characterized as "good" is approximately -20 dB, (i.e. 0.85 of FIG. 10 which corresponds to the value for XCorr_threshold 414 in FIG. 4).

[0086] In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40Hz test tone, the cross-correlation for a "good" ear seal is -8 dB, and for a bad ear seal it is -68 dB (i.e. a 60 dB difference). At a test signal frequency of 80Hz, the cross-correlation for a "good" ear seal is -8 dB and for a bad ear seal it is -38 dB (i.e. a 30 dB difference). Thus, above 200Hz the cross- correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

[0087] Using the cross-correlation rather than a level differencing approach improves the accuracy and minimizes errors which occur due to user non-speech body noise, such as teeth chatter; sneezes, coughs, etc. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0088] FIG. 8 is a flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) in accordance with an exemplary embodiment. The IMS is adjusted depending on the degree of acoustic sealing provided by an earpiece. The method begins at step 802. The acoustic sealing is measured as disclosed in FIG. 7 and the result provided in a step 804 to determine the cross-correlation (XCorr) between a test signal and corresponding ECM signal. In general, the higher the cross-correlation, the higher the degree of acoustic sealing. An exemplary graph showing the relationship between XCorr and acoustic sealing is given in FIG. 10. The degree of acoustic sealing is determined from known XCorr using a look-up (or "hash") table or using a formula (e.g. of a polynomial form) that maps the acoustic sealing to the known XCorr value. The ambient sound level is measured in a step 806. The ambient sound level corresponds to

the noise level in proximity to the user. In general, a higher degree of attenuation is desired when the ambient sound levels are high. Conversely, at low ambient sound levels the attenuation level of the IMS may be less of an issue and comfort more of a factor. The IMS is adjusted in a step 808 to meet the attenuation needs. In general, inflating the IMS increases attenuation while deflating the IMS decreases attenuation. The method terminates at step 810.

[0089] Referring to FIG. 9, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) is shown. In general, the attenuation increases when the pressure in the IMS is raised thereby allowing a degree of control to make adjustments. For example, an adjustment is made to increase attenuation when the background noise level rises or a seal check produces a failed result. Adjustments are made until the seal check passes. The pressure level adjustments of the IMS will fall within a comfort range of a user (e.g., between 0.1 bar and 0.3 bar gauge pressure). Typically, the pressure level is set at a minimum level to achieve a predetermined attenuation level.

[0090] The method begins at step 902. The degree of acoustic sealing is determined from cross-correlation between the ECM signal and the generated test signal. The XCorr value is provided in a step 904. In step 906, the attenuation provided by the IMS is calculated (equation) or looked up (table) from data such as that shown in FIG. 10. In one exemplary embodiment, the desired attenuation value is dependant on the ambient sound level of the user. In another exemplary embodiment, the desired attenuation value is dependant on the ear-canal sound level of the user. In yet another exemplary embodiment, the desired attenuation value is dependant on the level of audio content (e.g. speech or music audio) reproduced with the earphone device. In all of the above examples, the desired attenuation value is determined by one or more of the embodiments in a step 907.

[0091] The difference between the degree of acoustic sealing determined in step 906 and the desired attenuation value determined in step 907 is calculated in step 908. The difference value in step 908 is used to determine the change in pressure of the IMS necessary to minimize the difference value in a step 910. In at least one exemplary embodiment, the difference value of the attenuation is converted into a corresponding pressure value change (e.g. in milli-Bars) using a similar look-up table or equation method as described previously. The pressure change in the IMS is then affected with step 912 to meet the desired attenuation level. For example if the desired attenuation is a decrease of 10 dB in sound across the earpiece in the ear canal, then a pressure of a variable volume inflatable system can have a gauge pressure of about 0.15 bar. If the desired attenuation is a decrease of 20 dB across the earpiece in the ear canal then the gauge pressure can be increased to about 0.25 bar, where an increased pressure is associated with an increase in attenuation. An

experimental table for each earpiece can be generated in a standard devised experimental setup (e.g. impedance tunnel) and referred to when changes are needed. The method ends at step 914.

[0092] Referring to FIG. 11, a flowchart of an exemplary method to determine a test signal fundamental is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user or made pleasant such that the user is unaware that the test signal is being played. This allows unobtrusive (periodic or non- periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0093] In at least one exemplary embodiment, an audio content 1102 is provided. A step 1104 stores audio content 1102 in a data buffer. In this example, audio content 1102 is music played from a media player and received via a wired or wireless connection to at least one earpiece in the user's ear. An alternate example would be that audio content 1102 is a speech audio signal from a portable telephone device or the like.

[0094] A step 1106 determines if the buffer of audio content 1102 comprises a strong tonal signal component. Mixing the test signal having a similar fundamental frequency as audio content 1102 will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

[0095] A step 1108 determines whether to update or generate the first fundamental tone for the test signal. The test signal is not updated or generated if buffered audio content 402 does not contain a strong tonal signal component. A return to step 1104 fills the buffer with the next audio content 402 for analysis.

[0096] A step 1110 analyzes the buffer of data of audio content 1102 when it has been determined that it contains a strong tonal signal component. Step 1110 determines the fundamental frequency of the tonal signal. The fundamental tone, often referred to as the fundamental and abbreviate f_0 is the lowest frequency in a harmonic series. The fundamental frequency (also called a natural frequency) of a periodic signal is the inverse of the pitch period length. The pitch period is the smallest repeating unit of a signal. The fundamental frequency of the tonal signal can be calculated using an autocorrelation analysis.

[0097] In one exemplary embodiment, a mathematical operation 1114 is performed where the frequency component of the test signal is limited to a frequency range below a lower minimum and upper maximum frequency range. Fund_ratio is calculated, which is defined as a ratio of the determined fundamental frequency (F_fund) of the tonal signal from step 1110 to an upper threshold value F_fund_threshold 1112, which in one exemplary embodiment, is a fixed constant equal to approximately 100Hz. In general,

F_fund_threshold 1112 is chosen to be a low frequency value which is above the lowest (or -3 dB) frequency that a transducer can reproduce, but below a predetermined frequency. In a comparison step 1116, if the estimated F-fund is higher than the F_fund_threshold 1112 (ratio > 1), then F_fund is reduced by an integer multiple to be below F_fund_threshold 1112 corresponding to the mathematical operation of step 1118. Otherwise, the test signal fundamental is equal to F_fund as shown in step 1120. Although not shown, the calculated test signal fundamental is compared and determined to be greater than a predetermined threshold.

[0098] Referring to FIG. 1, in at least one exemplary embodiment, processor 116 is configured to receive or generate audio content. As mentioned previously, the audio content can from external devices such as a portable phone or a media player. Memory 104 can be used as a buffer for the audio content. Processor 116 is configured to receive the buffer of audio content from memory 104. The steps and calculations of the block diagram of FIG. 4 are then performed by processor 116. The result being one of the identification of a strong tonal signal component in the buffer of audio content and the test signal fundamental or loading the buffer with new audio content and starting the process again.

[0099] Referring to FIG. 12, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step 1106 of FIG. 11 that analyzes audio content stored in a buffer. The method begins at step 1202. A step 1204 gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step 1206 filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20Hz to 500Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step 1208 analyzes the audio signal where a strong tonal signal component is represented by peaks in the analysis results. A step 1210 generates Absolute(Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute(Acorr) can be the square of the results from the auto-correlation.

[0100] A crest_factor_Acorr 1218 is generated from the results by calculating an RMS value 1214 (or time-averaged peak value) and peak value 1216 (or time averaged peak value). In at least one exemplary embodiment, the crest_factor_Acorr 1218 is the ratio of the peak value to the RMS value of an absolute auto-correlation sequence of the audio signal.

[0101] A comparison step 1222 is then performed. A strong tonal presence is identified when crest_factor_Acorr 1218 is greater than a threshold Crest_factor_Acorr_threshold 1220. Identification of the

strong tonal presence indicates the audio signal would facilitate masking of the test signal to determine sealing of the device (step 1226). The audio signal is not used in conjunction with the test signal if crest_factor_Acorr 1218 is less than Crest_factor_Acorr_threshold 1220 (step 1224). The process would begin again loading a next sequence of the audio signal into the buffer for review.

[0102] Referring to FIG. 1, as mentioned previously, audio content is stored in a buffer, for example memory 104. The audio content in the buffer is provided to processor 116. In at least one exemplary embodiment, processor 116, runs the analysis as described in the block diagram of FIG. 12 thereby determining if a strong tonal presence is found in the audio content in the buffer. New audio content is loaded into the buffer (memory 104) if a strong tonal presence is not found beginning the procedure again.

[0103] Referring to FIG. 13, a flowchart of a method to determine when to emit the test signal is shown. The method begins at step 1302. The test signal is emitted when the test can be performed unobtrusively to the user and also provide an accurate test. In a step 1304, an audio signal is retrieved from a buffer. In at least one exemplary embodiment, the audio signal is received from an ECM or an ASM. The audio signal is measured to determine when the sound level is low in the ear canal, the ambient environment, or both. In general, the test signal is emitted when the sound level is low.

[0104] A filter step 1306 band pass filters the audio signal. In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150Hz which corresponds to a frequency range of the test signal. In a step 1308, the RMS of the audio signal is calculated. The audio signal is analyzed to detect when the energy within an audio frequency range is below a threshold RMS_threshold 1310. The RMS of the audio signal is the signal level in the volume being measured. A comparison step 1312 compares the measured RMS level of the filtered audio signal against RMS_threshold 1310. In a step 1314, a test signal is emitted when the measured RMS value is less than RMS_threshold 1310. No test signal is emitted when the RMS of the audio signal is greater than RMS_threshold 1310. The method ends at step 1316.

[0105] Referring to FIG. 14, a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the

earpiece to ensure that the seal is within normal operating specifications.

[0106] The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

[0107] In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

[0108] A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity of the signals in the first and second volumes. In particular, the non- difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

[0109] In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

[0110] Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the

predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a second comparison step 1428, the Magnitude of Peak is compared against a Peak_threshold 1430. If the Magnitude of Peak is greater than the Peak_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

[0111] In at least one exemplary embodiment and referring briefly to FIG. 1, an earpiece is tested to determine if sealing unit 108 is sealed correctly to the ear canal of the user. Sealing unit 108 creates a first volume in ear canal 124 and a second volume outside the ear canal 124 in the ambient environment. A masking approach is used to perform seal testing unobtrusively to the user. The user is listening to music or speech (audio content) provided to ear canal124 from ECR 114.

[0112] The music or speech is buffered in memory 104 or memory in processor 116. Processor 116 analyzes the audio content in the buffer to identify a strong tonal content. A test signal can be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor 116 also analyzes the measured signals from ECM 106 and ASM 120 to determine when to emit the test signal. Processor 116 monitors and compares the sound level in the ambient environment and ear canal124. Processor 116 will provide the generated test signal to ECR 114 during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor 116 will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

[0113] Processor 116 monitors the test conditions and then provides the test signal to ECR 114 when an accurate sealing test can be performed. ECR 114 outputs an acoustic test signal which may or may not have other audio content. ECM 106 and ASM 120 respectively measure acoustic signals in ear canal 124 and the ambient environment. Processor 116 is operatively coupled to ECM 106 and ASM 120. The measured signals are buffered in memory 104.

[0114] In an exemplary embodiment, a cross-correlation is used to measure the similarity between the

signals in ear canal124 and the ambient environment. Processor 116 performs the cross-correlation calculations using the measured acoustic signals from ECM 106 and ASM 120. In particular, the cross-correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$XCorr(n,l) = \sum_{n=0}^{N} ASM(n)ECM(n-l), \quad (5)$$

Where: l=0, 1, 2, … N

[0115] Where ASM(n) is the nth sample of the ASM signal, and ECM(n-[[1]] l) is the (n-[[1]] l)th sample of the ECM signal. A peak of the absolute cross-correlation is estimated using a peak- picking function and also the lag time at which this peak occurs (i.e. the index I at which this occurs). Thus, the Lag-time of Peak and the Magnitude of Peak are known and respectively compared against a Target Lag Limit range and a Peak Threshold. The user of the earpiece is notified or warned that sealing unit 108 is improperly sealed by processor 116 if the measured Lag-time of Peak is within the Target Lag Limit range and the Magnitude of Peak is greater than the Peak Threshold.

[0116] Like Correlation and Cross-Correlation, a coherence function is also a measure of similarity between two signals. Coherence is another non-difference comparison approach that can be used for detecting acoustic seal integrity. Coherence is defined as:

$$\gamma^2{}_{xy} = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (6)$$

[0117] Where Gxy is the cross-spectrum of two signals (e.g. the ASM and ECM signals), and can be calculated by first computing the cross-correlation in equation (5), applying a window function, for example a Hanning window, and transforming to the frequency domain, for example via an FFT. Gxx or Gyy is the auto-power spectrum of either the ASM or ECM signals, and can be calculated by first computing the auto-correlation (using equation 5, but where the two input signals are both from either the ASM or ECM and transforming to the frequency domain. The coherence function gives a frequency-dependent vector between 0 and 1, where a high coherence at a particular frequency indicates a high degree of coherence at this frequency, and can therefore be used to analyze test signal frequencies in the ASM and ECM signals whereby a high coherence indicates the presence of the test signal in the ambient environment (indicating

leakage past the acoustic barrier).

[0118] Other approaches such as frequency spectrum analysis and RMS levels can also be used to determine if the earpiece is sealed correctly. Using a non-difference comparison approach such as coherence or cross-correlation between the ASM and ECM signals to determine sealing is more reliable than taking the level difference of the ASM and ECM signals. Using the cross correlation rather than a level differencing approach improves the accuracy and minimizes errors which may occur due to user non-speech body noise, such as teeth chatter, sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0119] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., 1500) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. 15A and 15B shows a slider actuator 1500, with a moving slide 1510, having attached a pumping arm 1560, with a hole 1550 covering bump 1520. Upon receipt of a signal1505 the actuator can move 1570 such that the bump 1520 covers the hole 1550 in a bellows 1530 pump. The actuation compresses the miniature bellows 1530 pushing 1590 gas through the one-way valve 1540, upon the back stroke the bump 1520 uncovers the hole 1550 and air rushes back into the bellows 1530 for the next pump. For example if the stroke length is 2 mm and the pump arm 1560 contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system 1570 needs to be inflated more (e.g., more gas to increase sealing) then each stroke can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system 1580. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal 1505 sent, which can be specifically tailored depending

upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation systems can be used and controlled by signal 1505.

[0120] FIG. 16 illustrates a block diagram of controlled sealing in response to a seal fail signal and/or a request for increased seal attenuation. For example, a command signal 1600 is received (e.g., a seal fail signal in which a default attenuation value is attached for example 15 dB, or a signal requesting an additional amount of attenuation) by a processor. The command signal specifies an additional amount of attenuation or that the seal has failed. An attenuation needed N is identified 1610. For example if the current attenuation is 5 dB loss at f=500 Hz, at a pressure of 0.1 bar, and a command signal is received requesting a 10 dB loss at 500 Hz, then an experimental table is queried to find the pressure needed P which is then subtracted from the current pressure to obtain an increase in pressure DP needed. The increase in pressure is converted into a volume of gas increase needed (e.g., again referring to experimental tables based upon the inflation system volume). The volume of gas increase needed can then be directly linked with the number of cycles of an actuator pump M, 1620. The number of requested cycles **M** can then be sent 1630 to the actuator control circuit to pump the designated number of cycles. The system can then retest the attenuation 1640 and if refinements are needed the process can start again.

[0121] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

[0122] Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of U.S. Patent Application Ser. No. 17/182,570, filed 23 Feb. 2021, which is a continuation of U.S. Patent Application Ser. No. 16/838,277, filed April 2, 2020, which is a continuation of U.S. Patent Application Ser. No. 16/414,136, filed May 16, 2019, which is a continuation of U.S. Patent Application Ser. No. 15/700,511, filed September 11, 2017, which is a continuation of U.S. Patent Application Ser. No. 14/827,332, filed August 17, 2015, now U.S. Patent No. 9,781,530, which is a continuation of U.S. Patent Application Ser. No. 14/054,015, filed October 15, 2013, now U.S. Patent No. 9,113,267, which is a Divisional Application of U.S. Application Ser. No. 12/555,864, filed September 9, 2009, now U.S. Patent No. 8,600,067 and claims the benefit of U.S. Provisional Patent Application No. 61/098,250 filed September 19, 2008. The disclosure of all the aforementioned references are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to testing the seal of an orifice-inserted device, and more particularly, though not exclusively, to a device and method for determining if an earpiece is sealed correctly in an ear canal.

BACKGROUND OF THE INVENTION

[0003] It can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

[0004] Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient

sound to provide adequate performance to the user. Additionally, user conditions may change substantially during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

[0005] Broadly stated, embodiments are directed to a device and method to determine if an earpiece is sealing within the design specification of the device.

[0006] In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume. The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

[0007] At least one exemplary embodiment is directed to a method of detecting sealing integrity of an earpiece comprising the steps of: providing a test signal; generating an acoustic signal corresponding to the test signal incident on an ear canal side of a sealing section; converting the acoustic signal incident on a first side of the sealing section to an electrical signal; and cross- correlating the test signal to the electrical signal where the earpiece is sealed correctly when a cross-correlation between the test signal and the electrical signal is above a threshold.

[0008] At least one exemplary embodiment is directed to a method of adjusting attenuation of an earpiece comprising the steps of: relating cross-correlation of a test signal and a measured acoustic signal in an ear canal of a user to an attenuation level of a sealing section of the earpiece; comparing the attenuation level of the sealing section of the earpiece to a minimum attenuation value; and adjusting a pressure of the sealing section to meet the minimum attenuation value.

[0009] At least one exemplary embodiment is directed to a device comprising: a sealing section configured to seal a user's orifice, where the sealing section is configured to produce an acoustic seal between a first side of the sealing section and a second side of the sealing section; a transducer configured to generate a first acoustic signal incident on the first side of the sealing section; and a first microphone configured to measure a second acoustic signal incident on the second side of the sealing section, where the second acoustic signal

includes at least a portion of the first acoustic signal that has passed from the first side to the second side of the sealing section where the first acoustic signal is compared to the second acoustic signal to determine if the sealing section is sealed.

[0010] Further areas of applicability of exemplary embodiments of the present invention will become apparent from the detailed description provided hereinafter. It should be understood that the detailed description and specific examples, while indicating exemplary embodiments of the invention, are intended for purposes of illustration only and are not intended to limit the scope of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0012] FIG. 1 is a diagram of an earpiece inserted in an ear canal in accordance with an exemplary embodiment;

[0013] FIG. 2 is a block diagram of optional components of an earpiece in accordance with an exemplary embodiment;

[0014] FIG. 3 is a flowchart of a method for checking an ear seal in accordance with an exemplary embodiment;

[0015] FIG. 4 is a flowchart to determine acoustic seal integrity of an earpiece in accordance with the exemplary embodiment.

[0016] FIG. 5 is a flowchart of a method to estimate the instantaneous cross-correlation between a first and second audio signal.

[0017] FIG. 6 is a flowchart to determine when to emit a test signal in accordance with an exemplary embodiment;

[0018] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention;

[0019] FIG. 8 is a block diagram for a method of adjusting the IMS system in accordance with at least one exemplary embodiment;

[0020] FIG. 9 is a block diagram for a method of adjusting IMS pressure in accordance with at least one exemplary embodiment;

[0021] FIG. 10 illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with at least one exemplary embodiment;

[0022] FIG. 11 illustrates a flowchart of an exemplary method to determine a test signal fundamental;

[0023] FIG. 12 illustrates a flowchart of an exemplary embodiment to determine tonal presence in audio content;

[0024] FIG. 13 illustrates a flowchart of a method to determine when to emit the test signal;

[0025] FIG. 14 illustrates a flowchart of an exemplary method to determine acoustic seal integrity;

[0026] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment; and

[0027] FIG. 16 illustrates the sending of a signal to an inflation controller upon detection of a seal fail to modify the seal pressure in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION

[0028] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0029] Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

[0030] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0031] Additionally exemplary embodiments are not limited to earpieces, for example some functionality can be implemented on other systems with speakers and/or microphones for example computer systems, PDAs, BlackBerry® smart phones, cell and mobile phones, and any other device that emits or measures acoustic energy. Additionally, exemplary embodiments can be used with digital and non-digital acoustic systems. Additionally various receivers and microphones can be used, for example MEMs transducers,

diaphragm transducers, for example Knowles' FG and EG series transducers.

[0032] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0033] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0034] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for sealing or partially sealing an ear. FIG. 1 is a diagram of an earpiece inserted in an ear canal 124 in accordance with at least one exemplary embodiment of the invention. FIG. 1 also illustrates portions of the ear including pinna 128, ear canal 124 and eardrum 126. As illustrated, the earpiece comprises an electronic housing unit 100 and a sealing unit 108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 124 of a user 130. The earpiece is an in-ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude the ear canal 124.

[0036] The earpiece includes an Ambient Sound Microphone (ASM) 120 to capture ambient sound, an Ear Canal Receiver (ECR) 114 to deliver audio to an ear canal124, and an Ear Canal Microphone (ECM) 106 to capture and assess a sound exposure level within the ear canal 124. The earpiece can partially or fully occlude the ear canal 124 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 124, and to form an acoustic seal with the walls of the ear canal 124 at a location between the entrance to the ear canal 124 and the tympanic membrane (or ear drum) 126. In general, such a seal is typically achieved by means of a soft and compliant housing of the sealing unit 108. Additionally, the sealing unit 108 can be a pressurized expandable element that fills a portion of the available local space.

[0037] Sealing unit 108 is an acoustic barrier having a first side corresponding to ear canal124 and a second

side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 108

includes an ear canal microphone tube 110 and an ear canal receiver tube 112. Sealing unit 108 creates a closed cavity of approximately 5 cc or less between the first side of sealing unit 108 and the tympanic membrane 126 in ear canal124. In at least one exemplary embodiment the sealing facilitates using the ECR (speaker) 114 to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 126 resulting from the sound field at the entrance to the ear canal 124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0038] In at least one exemplary embodiment and in broader context, the second side of sealing unit 108 corresponds to the side adjacent to electronic housing unit 100. Ambient sound microphone 120 is housed in electronic housing unit 100 and is exposed to the ambient environment for receiving sound from the ambient environment around the user.

[0039] The electronic housing unit 100 can include various system components such as a microprocessor 116, memory 104, battery 102, ECM 106, ASM 120, ECR, 114, and user interface 122, or these components can reside in a separate system or interface operatively connected. Microprocessor 116 (or processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 116 is operatively coupled to memory 104, ECM 106, ASM 120, ECR 114, and user interface 122. An optional wire 118 can provide an external connection to the earpiece. Battery 102 powers the circuits and transducers of the earpiece. Battery 102 can be a rechargeable or replaceable battery.

[0040] In at least one exemplary embodiment, electronic housing unit 100 is adjacent to sealing unit 108. Openings in electronic housing unit 100 receive ECM tube 110 and ECR tube 112 to respectively couple to ECM 106 and ECR 114. ECR tube 112 and ECM tube 110 acoustically couple signals to and from ear canal 124. For example, ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124 where it is received by the tympanic membrane 126 of the user of the earpiece. Conversely, ECM 106 receives an acoustic signal present in ear canal124 though ECM tube 110.

[0041] One function of ECM 106 is that of measuring the sound pressure level in the ear canal cavity 124 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece. In one arrangement, ASM 120 is used to monitor sound pressure at the entrance to the occluded or partially occluded ear canal 124. All transducers shown can receive or transmit

audio signals to a processor 116 that undertakes audio signal processing and provides a transceiver for audio

via the wired (wire 118) or a wireless communication path. Note also that the acoustic signals can be stored for later retrieval.

[0042] In at least one exemplary embodiment the earpiece can be constructed to actively monitor a sound pressure level both inside and outside an ear canal 124. In at least one exemplary embodiment monitored data can be used to enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. In at least one exemplary embodiment an earpiece can facilitate at least one of conducting listening tests, filtering sounds in the environment, monitoring warning sounds in the environment, presenting notification based on identified warning sounds, maintaining constant audio content to ambient sound levels, and filtering sound in accordance with a Personalized Hearing Level (PHL).

[0043] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 124 using ECR 114 and ECM 106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 120. For instance, the ECR 114 can deliver an impulse within the ear canal124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. In at least one exemplary embodiment the earpiece can use either the ASM 120 or the ECM 106 to monitor the sound pressure level, which can then be used in a Sound Pressure Level Dosimeter calculation, to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0044] Referring to FIG. 2, a block diagram of an earpiece 201 in accordance with an exemplary embodiment is shown. A power supply 205 (e.g., USB power connection, hearing aid battery (batteries)) powers components of the earpiece 201 including microprocessor 206 (or processor 206, e.g., Texas Instruments TMS320C6713) and a data communication system 216 (e.g., RF or Bluetooth communication chip). As illustrated, the earpiece 201 includes the processor 206 operatively coupled to data communication system 216, ASM 210, ECR 212, and ECM 208. Data communication system 216 may include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 206. The processor 206 can also include a clock to record a time stamp.

[0045] In general, data communication system 216 is a communication pathway to components of the earpiece 201 and components external to the earpiece 201. The communication link can be wired or

wireless. In at least one exemplary embodiment, data communication system 216 is configured to communicate with ECM 208, ASM 210, visual display 218, and user control interface 214 of the earpiece 201. As shown, user control interface 214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 216 is capable of communication to devices exterior to the earpiece 201 such as the user's mobile phone 234, a second earpiece 222, and a portable media player 228. Portable media player 228 can be controlled by a manual user control 230.

[0046] The user's mobile phone 234 includes a mobile phone communication system 224. A microprocessor 226 is operatively coupled to mobile phone communication system 224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 220 worn by another person to the user's mobile phone. Similarly, the user's mobile phone 234 can be connected to the data communication system 216 of the earpiece 201 as well as the second earpiece 222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 234 through their respective earpieces.

[0047] As illustrated, a data communication system 216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

[0048] The RAM 202 stores program instructions for execution on the processor 206 as well as captured audio processing data. For instance, memory RAM 202 and ROM 204 can be off-chip and external to the processor 206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. In at least one exemplary embodiment, the data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer is operatively connected with processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0049] Data communication system 216 includes an audio interface operatively coupled to the processor

206 and the VOX to receive audio content, for example from portable media player 228, a cell phone, or

any other communication device, and deliver the audio content to the processor 206. The processor 206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 206 (or the VOX of data communication system 216) can lower a volume of the audio content responsive to detecting an event for transmitting the acute sound to the ear canal of the user. The processor 206 by way of the ECM 208 can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 216.

[0050] The earpiece 201 and data communication system 216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading over-the-air to the earpiece 201. It should be noted also that next generation access technologies can also be used in exemplary embodiments.

[0051] Data communication system 216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 201.

[0052] The power supply 205 utilizes common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 201 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 205 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0053] Microprocessor 206 is operatively connected with an EarSeal Inflation Management System 232 to control the degree to which the sealing unit 108 is inflated or deflated. In one exemplary embodiment, sealing unit 108 comprises an expandable element (e.g., inflatable balloon mechanism), whereby a cavity can be filled with air or a liquid to change the degree of acoustic isolation between the internal ear canal space 124 and the ambient environment. Alternately, a passive system for sealing ear canal 124 is used such as a flexible rubber or a silicon sealing unit or a foam plug. In one exemplary embodiment, the passive

system is a balloon mechanism that is filled with air or liquid. The balloon mechanism conforms to the

shape and size of an ear canal and includes a restorative force module that applies a pressure to the balloon mechanism for sealing the ear canal cavity.

[0054] The earpiece is a single operational device or a family of devices configured in a master- slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece are reused in different form factors for the master and slave devices.

[0055] Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user (second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

[0056] The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

[0057] In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

[0058] The degree of sealing between the first and second acoustic volumes is determined in a step 310 and

the process ends at step 312. The cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or "leaky") if the cross- correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly or incorrectly.

[0059] Referring to FIG. 1, the earpiece is used as an example to illustrate a test sequence as disclosed in FIG. 3. Sealing unit 108 occludes an opening of ear canal 124 creating a first volume (ear canal124) and a second volume (the ambient environment). Sealing unit 108 has a first side exposed to ear canal 124 and a second side is exposed to or corresponds to the ambient environment external to the ear.

[0060] In at least one exemplary embodiment, processor 116 is configured to receive a test signal in memory 104. Processor 116 generates the test signal and provides the test signal to Ear Canal Receiver 114 (ECR 114). ECR 114 emits the test signal into Ear Canal Receiver Tube (ECR Tube 112). The test signal propagates through ECR tube 112 and into ear canal 124. Ear Canal Microphone tube 110 (ECM tube 110) is configured to receive an acoustic signal incident on the first side of sealing unit 108. The test signal in ear canal 124 propagates through ECM tube 110 and is received by Ear Canal Microphone 106 (ECM 106). ECM 106 is configured to measure the test signal in ear canal124 and provide the measured test signal to processor 116.

[0061] As shown, electronic housing unit 100 of the earpiece is adjacent to the second side of sealing unit 108. Electronic housing unit 100 is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit 108. Electronic housing unit 100 includes Ambient Sound Microphone 120 (ASM 120), which is configured to measure sounds in the ambient environment. Thus, ASM 120 receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit 108. ASM 120 provides the measured ambient signal to processor 116.

[0062] Ideally, sealing unit 108 is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit 108 and into the ambient environment. Conversely, sealing unit 108 if improperly sealed will pass some of the test signal. Processor 116 compares the test signal to the signal

provided by ECM 106 corresponding to the acoustic signal in ear canal124. In particular, processor 116

undertakes the cross-correlation between emitted test signal and the ECM signal.

[0063] Referring to FIG. 4, a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0064] In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively--input) of the test signal data buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time- align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

[0065] A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively--input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10Hz greater than the test signal frequency.

[0066] A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5. The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

[0067] Referring to FIG. 5, a flowchart of an exemplary embodiment to determine the instantaneous cross-correlation between a first audio signal and a second audio signal is illustrated. The process begins at step 500. In at least one exemplary embodiment, the first audio signal is the test signal (i.e. a sine wave) and the second signal is the low-pass-filtered ECM signal.

[0068] The correlation between two signals x and y at time k using an exponential window is defined as:

[0069] $$p(k) = \frac{S_{xy}(k)}{\sqrt{S_{xx}(k)_{yy}(k)}} \quad (1)$$

[0070] Where

[0071] $$S_{xx}(k) = \sum_{l=0}^{\infty} ce^{-\eta^l} x_{k-l} y_{k-l} \quad (2)$$

$$c = 1 - e^{-\eta}$$

[0072] And S_{xx} and S_{yy} are defined similarly as in (2) (replacing y with x for S_{xx} etc.).

[0073] It can be shown (see Aarts et al, 2001) that (1) can be approximated with the recursion:

[0074] $$\hat{\rho}(k) = \hat{\rho}(k-1) + \gamma[\partial_k - \beta_k \hat{\rho}(k-1)] \quad (3)$$

$$\partial_k = 2x_k y_k$$

$$\beta_k = \alpha x_k^2 \alpha^{-1} y_k^2$$

[0075] Where

$$\alpha = \frac{y_{RMS}}{x_{RMS}} \quad (4)$$

$$\gamma = \frac{ce^{\eta}}{2x_{RMS} y_{RMS}}$$

[0077] The cross-correlation estimate using the above recursion is modified for block-wise processing rather than the sample-by-sample basis. This modification replaces the sample values (i.e. x(k) and y(k)) with values for the N-length block mean, i.e.

[0078]

$$x(k) = \frac{1}{N} \sum_{l=0}^{N-1} x(k-l)$$

[0079] Furthermore, the numerator for y is replaced with a small constant and so is α (replacing α with a

constant effectively un-normalizes the correlation estimate). It is found that the modified un-normalized block-wise cross-correlation accurately estimates the cross-correlation compared with using the standard cross-correlation for two signals.

[0080] The modified block-wise fast cross-correlation algorithm, as summarized in FIG. 5, comprises the following steps:

1. A first signal buffer 502 is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated 504 (x_{RMS})
3. The mean level of the first buffer is calculated 506.
4. A second signal buffer 508 is accumulated. This signal buffer corresponds to the filtered ECM signal.
5. The RMS level of the second buffer is calculated 510 (y_{RMS}).
6. The mean level of the second buffer is calculated 512.
7. In step 514, γ (gamma) is approximated as:

$$\gamma = \frac{\Gamma}{2x_{RMS}y_{RMS}}$$

Where Γ is a small constant, e.g. 10E-3.

8. In step 516,∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.
9. In step 518, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.
10. In step 520, the new temporary estimate of the correlation newRho_temp is calculated as: newRho_temp=(delta-beta*rho_old))
11. In step 522, the new estimate of the correlation newRho is updated by summing the previous estimate of the correlation with the product of gamma and the temporary estimate of the correlation newRho_temp.
12. In step 524, the "old" value of the correlation is set to the newest correlation estimate, ready for the next iteration of the update algorithm.
13. In step 526, the current correlation estimate between the emitted test signal and the received and filtered ECM signal is set as equal to the value of newRho.

[0081] Referring to FIG. 6, a flowchart of a method to determine when to emit the test signal is shown. The test signal is emitted when the test can be performed unobtrusively to the user and also provides an accurate test. In at least one exemplary embodiment, a test event to determine if an earpiece is sealed correctly is

initiated via a timing methodology. In a first timing scenario, the test event occurs after a delay of a first

predetermined time period when the RMS of the Audio Content (AC) is less than a RMS threshold. In a second timing scenario, the delay of the first predetermined time period is allowed to lapse without the test event occurring when the RMS of the audio content is greater than the RMS threshold. A second predetermined time period is started where the test event occurs when the RMS of the audio content is less than the RMS threshold. The test event is then initiated when the second predetermined time period is exceeded independent of the RMS of the audio content.

[0082] A test sequence is initiated in a step 602. The previous seal test event resets the first digital timer in a step 604. A time delay is generated by the loop comprising steps 606 and 608. The first digital timer is time incremented in the step 606. After each added time increment, the first digital timer is compared against a digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step 608 if the first digital timer is less than the digital_timer_threshold1.

[0083] A second digital timer is reset in a step 610 when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step 612. Audio content (AC) from a signal buffer is retrieved in a step 614. The audio content can be filtered through a low pass filter in an optional step 616. The RMS of the audio content is calculated in a step 618. The calculated RMS of the audio content is compared against an RMS_threshold 622 in a step 620. The second digital timer is compared against a digital_timer_threshold2 in a step 624 if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of 624.

[0084] The audio content signal is mixed with the test signal when the RMS of the audio content is less than the RMS_threshold in a step 626. Also, the audio content signal is mixed with the test signal when the second digital timer is greater than digital_timer_threshold2 in the step 624. The modified audio signal (having the test signal mixed in) is emitted by the ECR in a step 628 for testing the sealing section of the earpiece. The first digital timer is then reset in the step 604 to begin a timing sequence for another sealing section test.

[0085] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention. The estimated un-normalized cross-correlation between the ECM signal and the test signal (i.e. sine wave) is shown for different sine wave frequencies from 30-80Hz. Three different curves are provided corresponding

to a good fit (i.e. a tight optimal seal providing approximately 20-30 dB of acoustic attenuation), mid or

partial seal (i.e. an ear-seal that could be characterized as "half in" providing approximately 10-15 dB of acoustic attenuation), and a poor seal (i.e. an ear-seal providing less than 10 dB of acoustic attenuation). At lower test frequencies, the change in correlation is more pronounced as the degree of ear seal fitting is changed from "good" to "mid" and "poor". From the data, the threshold used to determine whether the ear seal can be characterized as "good" is approximately -20 dB, (i.e. 0.85 of FIG. 10 which corresponds to the value for XCorr_threshold 414 in FIG. 4).

[0086] In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40Hz test tone, the cross-correlation for a "good" ear seal is -8 dB, and for a bad ear seal it is -68 dB (i.e. a 60 dB difference). At a test signal frequency of 80Hz, the cross-correlation for a "good" ear seal is -8 dB and for a bad ear seal it is -38 dB (i.e. a 30 dB difference). Thus, above 200Hz the cross- correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

[0087] Using the cross-correlation rather than a level differencing approach improves the accuracy and minimizes errors which occur due to user non-speech body noise, such as teeth chatter; sneezes, coughs, etc. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0088] FIG. 8 is a flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) in accordance with an exemplary embodiment. The IMS is adjusted depending on the degree of acoustic sealing provided by an earpiece. The method begins at step 802. The acoustic sealing is measured as disclosed in FIG. 7 and the result provided in a step 804 to determine the cross-correlation (XCorr) between a test signal and corresponding ECM signal. In general, the higher the cross-correlation, the higher the degree of acoustic sealing. An exemplary graph showing the relationship between XCorr and acoustic sealing is given in FIG. 10. The degree of acoustic sealing is determined from known XCorr using a look-up (or "hash") table or using a formula (e.g. of a polynomial form) that maps the acoustic sealing to the known XCorr value. The ambient sound level is measured in a step 806. The ambient sound level corresponds to

the noise level in proximity to the user. In general, a higher degree of attenuation is desired when the

ambient sound levels are high. Conversely, at low ambient sound levels the attenuation level of the IMS may be less of an issue and comfort more of a factor. The IMS is adjusted in a step 808 to meet the attenuation needs. In general, inflating the IMS increases attenuation while deflating the IMS decreases attenuation. The method terminates at step 810.

[0089] Referring to FIG. 9, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) is shown. In general, the attenuation increases when the pressure in the IMS is raised thereby allowing a degree of control to make adjustments. For example, an adjustment is made to increase attenuation when the background noise level rises or a seal check produces a failed result. Adjustments are made until the seal check passes. The pressure level adjustments of the IMS will fall within a comfort range of a user (e.g., between 0.1 bar and 0.3 bar gauge pressure). Typically, the pressure level is set at a minimum level to achieve a predetermined attenuation level.

[0090] The method begins at step 902. The degree of acoustic sealing is determined from cross-correlation between the ECM signal and the generated test signal. The XCorr value is provided in a step 904. In step 906, the attenuation provided by the IMS is calculated (equation) or looked up (table) from data such as that shown in FIG. 10. In one exemplary embodiment, the desired attenuation value is dependant on the ambient sound level of the user. In another exemplary embodiment, the desired attenuation value is dependant on the ear-canal sound level of the user. In yet another exemplary embodiment, the desired attenuation value is dependant on the level of audio content (e.g. speech or music audio) reproduced with the earphone device. In all of the above examples, the desired attenuation value is determined by one or more of the embodiments in a step 907.

[0091] The difference between the degree of acoustic sealing determined in step 906 and the desired attenuation value determined in step 907 is calculated in step 908. The difference value in step 908 is used to determine the change in pressure of the IMS necessary to minimize the difference value in a step 910. In at least one exemplary embodiment, the difference value of the attenuation is converted into a corresponding pressure value change (e.g. in milli-Bars) using a similar look-up table or equation method as described previously. The pressure change in the IMS is then affected with step 912 to meet the desired attenuation level. For example if the desired attenuation is a decrease of 10 dB in sound across the earpiece in the ear canal, then a pressure of a variable volume inflatable system can have a gauge pressure of about 0.15 bar. If the desired attenuation is a decrease of 20 dB across the earpiece in the ear canal then the gauge pressure can be increased to about 0.25 bar, where an increased pressure is associated with an increase in attenuation. An

experimental table for each earpiece can be generated in a standard devised experimental setup (e.g.

impedance tunnel) and referred to when changes are needed. The method ends at step 914.

[0092] Referring to FIG. 11, a flowchart of an exemplary method to determine a test signal fundamental is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user or made pleasant such that the user is unaware that the test signal is being played. This allows unobtrusive (periodic or non- periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0093] In at least one exemplary embodiment, an audio content 1102 is provided. A step 1104 stores audio content 1102 in a data buffer. In this example, audio content 1102 is music played from a media player and received via a wired or wireless connection to at least one earpiece in the user's ear. An alternate example would be that audio content 1102 is a speech audio signal from a portable telephone device or the like.

[0094] A step 1106 determines if the buffer of audio content 1102 comprises a strong tonal signal component. Mixing the test signal having a similar fundamental frequency as audio content 1102 will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

[0095] A step 1108 determines whether to update or generate the first fundamental tone for the test signal. The test signal is not updated or generated if buffered audio content 402 does not contain a strong tonal signal component. A return to step 1104 fills the buffer with the next audio content 402 for analysis.

[0096] A step 1110 analyzes the buffer of data of audio content 1102 when it has been determined that it contains a strong tonal signal component. Step 1110 determines the fundamental frequency of the tonal signal. The fundamental tone, often referred to as the fundamental and abbreviate f_0 is the lowest frequency in a harmonic series. The fundamental frequency (also called a natural frequency) of a periodic signal is the inverse of the pitch period length. The pitch period is the smallest repeating unit of a signal. The fundamental frequency of the tonal signal can be calculated using an autocorrelation analysis.

[0097] In one exemplary embodiment, a mathematical operation 1114 is performed where the frequency component of the test signal is limited to a frequency range below a lower minimum and upper maximum frequency range. Fund_ratio is calculated, which is defined as a ratio of the determined fundamental frequency (F_fund) of the tonal signal from step 1110 to an upper threshold value F_fund_threshold 1112, which in one exemplary embodiment, is a fixed constant equal to approximately 100Hz. In general,

F_fund_threshold 1112 is chosen to be a low frequency value which is above the lowest (or -3 dB)

frequency that a transducer can reproduce, but below a predetermined frequency. In a comparison step 1116, if the estimated F-fund is higher than the F_fund_threshold 1112 (ratio > 1), then F_fund is reduced by an integer multiple to be below F_fund_threshold 1112 corresponding to the mathematical operation of step 1118. Otherwise, the test signal fundamental is equal to F_fund as shown in step 1120. Although not shown, the calculated test signal fundamental is compared and determined to be greater than a predetermined threshold.

[0098] Referring to FIG. 1, in at least one exemplary embodiment, processor 116 is configured to receive or generate audio content. As mentioned previously, the audio content can from external devices such as a portable phone or a media player. Memory 104 can be used as a buffer for the audio content. Processor 116 is configured to receive the buffer of audio content from memory 104. The steps and calculations of the block diagram of FIG. 4 are then performed by processor 116. The result being one of the identification of a strong tonal signal component in the buffer of audio content and the test signal fundamental or loading the buffer with new audio content and starting the process again.

[0099] Referring to FIG. 12, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step 1106 of FIG. 11 that analyzes audio content stored in a buffer. The method begins at step 1202. A step
1204 gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step 1206 filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20Hz to 500Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step 1208 analyzes the audio signal where a strong tonal signal component is represented by peaks in the analysis results. A step 1210 generates Absolute(Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute(Acorr) can be the square of the results from the auto-correlation.

[0100] A crest_factor_Acorr 1218 is generated from the results by calculating an RMS value 1214 (or time-averaged peak value) and peak value 1216 (or time averaged peak value). In at least one exemplary embodiment, the crest_factor_Acorr 1218 is the ratio of the peak value to the RMS value of an absolute auto-correlation sequence of the audio signal.

[0101] A comparison step 1222 is then performed. A strong tonal presence is identified when crest_factor_Acorr 1218 is greater than a threshold Crest_factor_Acorr_threshold 1220. Identification of the

strong tonal presence indicates the audio signal would facilitate masking of the test signal to determine

sealing of the device (step 1226). The audio signal is not used in conjunction with the test signal if crest_factor_Acorr 1218 is less than Crest_factor_Acorr_threshold 1220 (step 1224). The process would begin again loading a next sequence of the audio signal into the buffer for review.

[0102] Referring to FIG. 1, as mentioned previously, audio content is stored in a buffer, for example memory 104. The audio content in the buffer is provided to processor 116. In at least one exemplary embodiment, processor 116, runs the analysis as described in the block diagram of FIG. 12 thereby determining if a strong tonal presence is found in the audio content in the buffer. New audio content is loaded into the buffer (memory 104) if a strong tonal presence is not found beginning the procedure again.

[0103] Referring to FIG. 13, a flowchart of a method to determine when to emit the test signal is shown. The method begins at step 1302. The test signal is emitted when the test can be performed unobtrusively to the user and also provide an accurate test. In a step 1304, an audio signal is retrieved from a buffer. In at least one exemplary embodiment, the audio signal is received from an ECM or an ASM. The audio signal is measured to determine when the sound level is low in the ear canal, the ambient environment, or both. In general, the test signal is emitted when the sound level is low.

[0104] A filter step 1306 band pass filters the audio signal. In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150Hz which corresponds to a frequency range of the test signal. In a step 1308, the RMS of the audio signal is calculated. The audio signal is analyzed to detect when the energy within an audio frequency range is below a threshold RMS_threshold 1310. The RMS of the audio signal is the signal level in the volume being measured. A comparison step 1312 compares the measured RMS level of the filtered audio signal against RMS_threshold 1310. In a step 1314, a test signal is emitted when the measured RMS value is less than RMS_threshold 1310. No test signal is emitted when the RMS of the audio signal is greater than RMS_threshold 1310. The method ends at step 1316.

[0105] Referring to FIG. 14, a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the

earpiece to ensure that the seal is within normal operating specifications.

[0106] The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

[0107] In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

[0108] A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity of the signals in the first and second volumes. In particular, the non- difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

[0109] In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

[0110] Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the

predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within

the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a second comparison step 1428, the Magnitude of Peak is compared against a Peak_threshold 1430. If the Magnitude of Peak is greater than the Peak_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

[0111] In at least one exemplary embodiment and referring briefly to FIG. 1, an earpiece is tested to determine if sealing unit 108 is sealed correctly to the ear canal of the user. Sealing unit 108 creates a first volume in ear canal 124 and a second volume outside the ear canal 124 in the ambient environment. A masking approach is used to perform seal testing unobtrusively to the user. The user is listening to music or speech (audio content) provided to ear canal124 from ECR 114.

[0112] The music or speech is buffered in memory 104 or memory in processor 116. Processor 116 analyzes the audio content in the buffer to identify a strong tonal content. A test signal can be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor 116 also analyzes the measured signals from ECM 106 and ASM 120 to determine when to emit the test signal. Processor 116 monitors and compares the sound level in the ambient environment and ear canal124. Processor 116 will provide the generated test signal to ECR 114 during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor 116 will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

[0113] Processor 116 monitors the test conditions and then provides the test signal to ECR 114 when an accurate sealing test can be performed. ECR 114 outputs an acoustic test signal which may or may not have other audio content. ECM 106 and ASM 120 respectively measure acoustic signals in ear canal 124 and the ambient environment. Processor 116 is operatively coupled to ECM 106 and ASM 120. The measured signals are buffered in memory 104.

[0114] In an exemplary embodiment, a cross-correlation is used to measure the similarity between the

signals in ear canal124 and the ambient environment. Processor 116 performs the cross-correlation

calculations using the measured acoustic signals from ECM 106 and ASM 120. In particular, the cross-correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$XCorr(n,l) = \sum_{n=0}^{N} ASM(n)ECM(n-l), \quad (5)$$

Where: l=0, 1, 2, … N

[0115] Where ASM(n) is the nth sample of the ASM signal, and ECM(n-l) is the (n-l)th sample of the ECM signal. A peak of the absolute cross-correlation is estimated using a peak- picking function and also the lag time at which this peak occurs (i.e. the index I at which this occurs). Thus, the Lag-time of Peak and the Magnitude of Peak are known and respectively compared against a Target Lag Limit range and a Peak Threshold. The user of the earpiece is notified or warned that sealing unit 108 is improperly sealed by processor 116 if the measured Lag-time of Peak is within the Target Lag Limit range and the Magnitude of Peak is greater than the Peak Threshold.

[0116] Like Correlation and Cross-Correlation, a coherence function is also a measure of similarity between two signals. Coherence is another non-difference comparison approach that can be used for detecting acoustic seal integrity. Coherence is defined as:

$$\gamma^2{}_{xy} = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (6)$$

[0117] Where Gxy is the cross-spectrum of two signals (e.g. the ASM and ECM signals), and can be calculated by first computing the cross-correlation in equation (5), applying a window function, for example a Hanning window, and transforming to the frequency domain, for example via an FFT. Gxx or Gyy is the auto-power spectrum of either the ASM or ECM signals, and can be calculated by first computing the auto-correlation (using equation 5, but where the two input signals are both from either the ASM or ECM and transforming to the frequency domain. The coherence function gives a frequency-dependent vector between 0 and 1, where a high coherence at a particular frequency indicates a high degree of coherence at this frequency, and can therefore be used to analyze test signal frequencies in the ASM and ECM signals whereby a high coherence indicates the presence of the test signal in the ambient environment (indicating

leakage past the acoustic barrier).

[0118] Other approaches such as frequency spectrum analysis and RMS levels can also be used to determine if the earpiece is sealed correctly. Using a non-difference comparison approach such as coherence or cross-correlation between the ASM and ECM signals to determine sealing is more reliable than taking the level difference of the ASM and ECM signals. Using the cross correlation rather than a level differencing approach improves the accuracy and minimizes errors which may occur due to user non-speech body noise, such as teeth chatter, sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0119] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., 1500) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. 15A and 15B shows a slider actuator 1500, with a moving slide 1510, having attached a pumping arm 1560, with a hole 1550 covering bump 1520. Upon receipt of a signal1505 the actuator can move 1570 such that the bump 1520 covers the hole 1550 in a bellows 1530 pump. The actuation compresses the miniature bellows 1530 pushing 1590 gas through the one-way valve 1540, upon the back stroke the bump 1520 uncovers the hole 1550 and air rushes back into the bellows 1530 for the next pump. For example if the stroke length is 2 mm and the pump arm 1560 contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system 1570 needs to be inflated more (e.g., more gas to increase sealing) then each stroke can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system 1580. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal 1505 sent, which can be specifically tailored depending upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation

systems can be used and controlled by signal 1505.

[0120] FIG. 16 illustrates a block diagram of controlled sealing in response to a seal fail signal and/or a request for increased seal attenuation. For example, a command signal 1600 is received (e.g., a seal fail signal in which a default attenuation value is attached for example 15 dB, or a signal requesting an additional amount of attenuation) by a processor. The command signal specifies an additional amount of attenuation or that the seal has failed. An attenuation needed N is identified 1610. For example if the current attenuation is 5 dB loss at f=500 Hz, at a pressure of 0.1 bar, and a command signal is received requesting a 10 dB loss at 500 Hz, then an experimental table is queried to find the pressure needed P which is then subtracted from the current pressure to obtain an increase in pressure DP needed. The increase in pressure is converted into a volume of gas increase needed (e.g., again referring to experimental tables based upon the inflation system volume). The volume of gas increase needed can then be directly linked with the number of cycles of an actuator pump M, 1620. The number of requested cycles M can then be sent 1630 to the actuator control circuit to pump the designated number of cycles. The system can then retest the attenuation 1640 and if refinements are needed the process can start again.

[0121] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

[0122] Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

2nd Amendment After Allowance Under 37 CFR 1.312

Commissioner of Patents
Alexandria, VA 22313-1450

DATE: 3 Jan. 2023

The Office issued a notice of allowance 23 Dec. 2022. Prior to final payment of the issue fee Applicant has noticed two additional grammatical issues, corrected herein. Thus, Applicants submit the second attached marked-up and clean version of the specification to address typographical errors, in particular in paragraphs [0001] and formula (2) [0071] where Sxx should be Sxy.

The minor 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. However, 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.

Respectfully submitted,
TECHIYA LLC

Date: January 3, 2023

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Marked Specification

ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application ~~is a continuation of U.S. Patent Application Ser. No. 17/182,570, filed 23 Feb. 2021, which~~ is a continuation of U.S. Patent Application Ser. No. 16/838,277, filed April 2, 2020, which is a continuation of U.S. Patent Application Ser. No. 16/414,136, filed May 16, 2019, which is a continuation of U.S. Patent Application Ser. No. 15/700,511, filed September 11, 2017, which is a continuation of U.S. Patent Application Ser. No. 14/827,332, filed August 17, 2015, now U.S. Patent No. 9,781,530, which is a continuation of U.S. Patent Application Ser. No. 14/054,015, filed October 15, 2013, now U.S. Patent No. 9,113,267, which is a Divisional Application of U.S. Application Ser. No. 12/555,864, filed September 9, 2009, now U.S. Patent No. 8,600,067 and claims the benefit of U.S. Provisional Patent Application No. 61/098,250 filed September 19, 2008. The disclosure of all the aforementioned references are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to testing the seal of an orifice-inserted device, and more particularly, though not exclusively, to a device and method for determining if an earpiece is sealed correctly in an ear canal.

BACKGROUND OF THE INVENTION

[0003] It can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

[0004] Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient

sound to provide adequate performance to the user. Additionally, user conditions may change substantially during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

[0005] Broadly stated, embodiments are directed to a device and method to determine if an earpiece is sealing within the design specification of the device.

[0006] In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume. The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

[0007] At least one exemplary embodiment is directed to a method of detecting sealing integrity of an earpiece comprising the steps of: providing a test signal; generating an acoustic signal corresponding to the test signal incident on an ear canal side of a sealing section; converting the acoustic signal incident on a first side of the sealing section to an electrical signal; and cross- correlating the test signal to the electrical signal where the earpiece is sealed correctly when a cross-correlation between the test signal and the electrical signal is above a threshold.

[0008] At least one exemplary embodiment is directed to a method of adjusting attenuation of an earpiece comprising the steps of: relating cross-correlation of a test signal and a measured acoustic signal in an ear canal of a user to an attenuation level of a sealing section of the earpiece; comparing the attenuation level of the sealing section of the earpiece to a minimum attenuation value; and adjusting a pressure of the sealing section to meet the minimum attenuation value.

[0009] At least one exemplary embodiment is directed to a device comprising: a sealing section configured to seal a user's orifice, where the sealing section is configured to produce an acoustic seal between a first side of the sealing section and a second side of the sealing section; a transducer configured to generate a first acoustic signal incident on the first side of the sealing section; and a first microphone configured to measure a second acoustic signal incident on the second side of the sealing section, where the second acoustic signal

includes at least a portion of the first acoustic signal that has passed from the first side to the second side of the sealing section where the first acoustic signal is compared to the second acoustic signal to determine if the sealing section is sealed.

[0010] Further areas of applicability of exemplary embodiments of the present invention will become apparent from the detailed description provided hereinafter. It should be understood that the detailed description and specific examples, while indicating exemplary embodiments of the invention, are intended for purposes of illustration only and are not intended to limit the scope of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0012] FIG. 1 is a diagram of an earpiece inserted in an ear canal in accordance with an exemplary embodiment;

[0013] FIG. 2 is a block diagram of optional components of an earpiece in accordance with an exemplary embodiment;

[0014] FIG. 3 is a flowchart of a method for checking an ear seal in accordance with an exemplary embodiment;

[0015] FIG. 4 is a flowchart to determine acoustic seal integrity of an earpiece in accordance with the exemplary embodiment.

[0016] FIG. 5 is a flowchart of a method to estimate the instantaneous cross-correlation between a first and second audio signal.

[0017] FIG. 6 is a flowchart to determine when to emit a test signal in accordance with an exemplary embodiment;

[0018] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention;

[0019] FIG. 8 is a block diagram for a method of adjusting the IMS system in accordance with at least one exemplary embodiment;

[0020] FIG. 9 is a block diagram for a method of adjusting IMS pressure in accordance with at least one exemplary embodiment;

[0021] FIG. 10 illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with at least one exemplary embodiment;

[0022] FIG. 11 illustrates a flowchart of an exemplary method to determine a test signal fundamental;

[0023] FIG. 12 illustrates a flowchart of an exemplary embodiment to determine tonal presence in audio content;

[0024] FIG. 13 illustrates a flowchart of a method to determine when to emit the test signal;

[0025] FIG. 14 illustrates a flowchart of an exemplary method to determine acoustic seal integrity;

[0026] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment; and

[0027] FIG. 16 illustrates the sending of a signal to an inflation controller upon detection of a seal fail to modify the seal pressure in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION

[0028] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0029] Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

[0030] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0031] Additionally exemplary embodiments are not limited to earpieces, for example some functionality can be implemented on other systems with speakers and/or microphones for example computer systems, PDAs, BlackBerry® smart phones, cell and mobile phones, and any other device that emits or measures acoustic energy. Additionally, exemplary embodiments can be used with digital and non-digital acoustic

systems. Additionally various receivers and microphones can be used, for example MEMs transducers, diaphragm transducers, for example Knowles' FG and EG series transducers.

[0032] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0033] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0034] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for sealing or partially sealing an ear. FIG. 1 is a diagram of an earpiece inserted in an ear canal 124 in accordance with at least one exemplary embodiment of the invention. FIG. 1 also illustrates portions of the ear including pinna 128, ear canal 124 and eardrum 126. As illustrated, the earpiece comprises an electronic housing unit 100 and a sealing unit 108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 124 of a user 130. The earpiece is an in-ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude the ear canal 124.

[0036] The earpiece includes an Ambient Sound Microphone (ASM) 120 to capture ambient sound, an Ear Canal Receiver (ECR) 114 to deliver audio to an ear canal124, and an Ear Canal Microphone (ECM) 106 to capture and assess a sound exposure level within the ear canal 124. The earpiece can partially or fully occlude the ear canal 124 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 124, and to form an acoustic seal with the walls of the ear canal 124 at a location between the entrance to the ear canal 124 and the tympanic membrane (or ear drum) 126. In general, such a seal is typically achieved by means of a soft and compliant housing of the sealing unit 108. Additionally, the sealing unit 108 can be a pressurized expandable element that fills a portion of the available local space.

[0037] Sealing unit 108 is an acoustic barrier having a first side corresponding to ear canal124 and a second

side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 108 includes an ear canal microphone tube 110 and an ear canal receiver tube 112. Sealing unit 108 creates a closed cavity of approximately 5 cc or less between the first side of sealing unit 108 and the tympanic membrane 126 in ear canal124. In at least one exemplary embodiment the sealing facilitates using the ECR (speaker) 114 to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 126 resulting from the sound field at the entrance to the ear canal 124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0038] In at least one exemplary embodiment and in broader context, the second side of sealing unit 108 corresponds to the side adjacent to electronic housing unit 100. Ambient sound microphone 120 is housed in electronic housing unit 100 and is exposed to the ambient environment for receiving sound from the ambient environment around the user.

[0039] The electronic housing unit 100 can include various system components such as a microprocessor 116, memory 104, battery 102, ECM 106, ASM 120, ECR, 114, and user interface 122, or these components can reside in a separate system or interface operatively connected. Microprocessor 116 (or processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 116 is operatively coupled to memory 104, ECM 106, ASM 120, ECR 114, and user interface 122. An optional wire 118 can provide an external connection to the earpiece. Battery 102 powers the circuits and transducers of the earpiece. Battery 102 can be a rechargeable or replaceable battery.

[0040] In at least one exemplary embodiment, electronic housing unit 100 is adjacent to sealing unit 108. Openings in electronic housing unit 100 receive ECM tube 110 and ECR tube 112 to respectively couple to ECM 106 and ECR 114. ECR tube 112 and ECM tube 110 acoustically couple signals to and from ear canal 124. For example, ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124 where it is received by the tympanic membrane 126 of the user of the earpiece. Conversely, ECM 106 receives an acoustic signal present in ear canal124 though ECM tube 110.

[0041] One function of ECM 106 is that of measuring the sound pressure level in the ear canal cavity 124 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece. In one arrangement, ASM 120 is used to monitor sound pressure at the entrance to the occluded or partially occluded ear canal 124. All transducers shown can receive or transmit

audio signals to a processor 116 that undertakes audio signal processing and provides a transceiver for audio via the wired (wire 118) or a wireless communication path. Note also that the acoustic signals can be stored for later retrieval.

[0042] In at least one exemplary embodiment the earpiece can be constructed to actively monitor a sound pressure level both inside and outside an ear canal 124. In at least one exemplary embodiment monitored data can be used to enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. In at least one exemplary embodiment an earpiece can facilitate at least one of conducting listening tests, filtering sounds in the environment, monitoring warning sounds in the environment, presenting notification based on identified warning sounds, maintaining constant audio content to ambient sound levels, and filtering sound in accordance with a Personalized Hearing Level (PHL).

[0043] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 124 using ECR 114 and ECM 106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 120. For instance, the ECR 114 can deliver an impulse within the ear canal124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. In at least one exemplary embodiment the earpiece can use either the ASM 120 or the ECM 106 to monitor the sound pressure level, which can then be used in a Sound Pressure Level Dosimeter calculation, to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0044] Referring to FIG. 2, a block diagram of an earpiece 201 in accordance with an exemplary embodiment is shown. A power supply 205 (e.g., USB power connection, hearing aid battery (batteries)) powers components of the earpiece 201 including microprocessor 206 (or processor 206, e.g., Texas Instruments TMS320C6713) and a data communication system 216 (e.g., RF or Bluetooth communication chip). As illustrated, the earpiece 201 includes the processor 206 operatively coupled to data communication system 216, ASM 210, ECR 212, and ECM 208. Data communication system 216 may include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 206. The processor 206 can also include a clock to record a time stamp.

[0045] In general, data communication system 216 is a communication pathway to components of the earpiece 201 and components external to the earpiece 201. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system 216 is configured to communicate with ECM 208, ASM 210, visual display 218, and user control interface 214 of the earpiece 201. As shown, user control interface 214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 216 is capable of communication to devices exterior to the earpiece 201 such as the user's mobile phone 234, a second earpiece 222, and a portable media player 228. Portable media player 228 can be controlled by a manual user control 230.

[0046] The user's mobile phone 234 includes a mobile phone communication system 224. A microprocessor 226 is operatively coupled to mobile phone communication system 224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 220 worn by another person to the user's mobile phone. Similarly, the user's mobile phone 234 can be connected to the data communication system 216 of the earpiece 201 as well as the second earpiece 222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 234 through their respective earpieces.

[0047] As illustrated, a data communication system 216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

[0048] The RAM 202 stores program instructions for execution on the processor 206 as well as captured audio processing data. For instance, memory RAM 202 and ROM 204 can be off-chip and external to the processor 206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. In at least one exemplary embodiment, the data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer is operatively connected with processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0049] Data communication system 216 includes an audio interface operatively coupled to the processor

206 and the VOX to receive audio content, for example from portable media player 228, a cell phone, or any other communication device, and deliver the audio content to the processor 206. The processor 206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 206 (or the VOX of data communication system 216) can lower a volume of the audio content responsive to detecting an event for transmitting the acute sound to the ear canal of the user. The processor 206 by way of the ECM 208 can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 216.

[0050] The earpiece 201 and data communication system 216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading over-the-air to the earpiece 201. It should be noted also that next generation access technologies can also be used in exemplary embodiments.

[0051] Data communication system 216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 201.

[0052] The power supply 205 utilizes common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 201 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 205 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0053] Microprocessor 206 is operatively connected with an EarSeal Inflation Management System 232 to control the degree to which the sealing unit 108 is inflated or deflated. In one exemplary embodiment, sealing unit 108 comprises an expandable element (e.g., inflatable balloon mechanism), whereby a cavity can be filled with air or a liquid to change the degree of acoustic isolation between the internal ear canal space 124 and the ambient environment. Alternately, a passive system for sealing ear canal 124 is used such as a flexible rubber or a silicon sealing unit or a foam plug. In one exemplary embodiment, the passive

system is a balloon mechanism that is filled with air or liquid. The balloon mechanism conforms to the shape and size of an ear canal and includes a restorative force module that applies a pressure to the balloon mechanism for sealing the ear canal cavity.

[0054] The earpiece is a single operational device or a family of devices configured in a master- slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece are reused in different form factors for the master and slave devices.

[0055] Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user (second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

[0056] The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

[0057] In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

[0058] The degree of sealing between the first and second acoustic volumes is determined in a step 310 and the process ends at step 312. The cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or "leaky") if the cross- correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly or incorrectly.

[0059] Referring to FIG. 1, the earpiece is used as an example to illustrate a test sequence as disclosed in FIG. 3. Sealing unit 108 occludes an opening of ear canal 124 creating a first volume (ear canal124) and a second volume (the ambient environment). Sealing unit 108 has a first side exposed to ear canal 124 and a second side is exposed to or corresponds to the ambient environment external to the ear.

[0060] In at least one exemplary embodiment, processor 116 is configured to receive a test signal in memory 104. Processor 116 generates the test signal and provides the test signal to Ear Canal Receiver 114 (ECR 114). ECR 114 emits the test signal into Ear Canal Receiver Tube (ECR Tube 112). The test signal propagates through ECR tube 112 and into ear canal 124. Ear Canal Microphone tube 110 (ECM tube 110) is configured to receive an acoustic signal incident on the first side of sealing unit 108. The test signal in ear canal 124 propagates through ECM tube 110 and is received by Ear Canal Microphone 106 (ECM 106). ECM 106 is configured to measure the test signal in ear canal124 and provide the measured test signal to processor 116.

[0061] As shown, electronic housing unit 100 of the earpiece is adjacent to the second side of sealing unit 108. Electronic housing unit 100 is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit 108. Electronic housing unit 100 includes Ambient Sound Microphone 120 (ASM 120), which is configured to measure sounds in the ambient environment. Thus, ASM 120 receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit 108. ASM 120 provides the measured ambient signal to processor 116.

[0062] Ideally, sealing unit 108 is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit 108 and into the ambient environment. Conversely, sealing unit 108 if improperly sealed will pass some of the test signal. Processor 116 compares the test signal to the signal

provided by ECM 106 corresponding to the acoustic signal in ear canal124. In particular, processor 116 undertakes the cross-correlation between emitted test signal and the ECM signal.

[0063] Referring to FIG. 4, a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0064] In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively--input) of the test signal data buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time- align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

[0065] A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively--input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10Hz greater than the test signal frequency.

[0066] A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5. The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

[0067] Referring to FIG. 5, a flowchart of an exemplary embodiment to determine the instantaneous cross-correlation between a first audio signal and a second audio signal is illustrated. The process begins at step 500. In at least one exemplary embodiment, the first audio signal is the test signal (i.e. a sine wave) and the second signal is the low-pass-filtered ECM signal.

[0068] The correlation between two signals x and y at time k using an exponential window is defined as:

[0069] $$p(k) = \frac{S_{xy}(k)}{\sqrt{S_{xx}(k)_{yy}(k)}} \quad (1)$$

[0070] Where

[0071] $$S_{x[[x]]\underline{\prime}}(k) = \sum_{l=0}^{\infty} ce^{-\eta^l} x_{k-l} y_{k-l} \quad (2)$$

$$c = 1 - e^{-\eta}$$

[0072] And S_{xx} and S_{yy} are defined similarly as in (2) (replacing y with x for S_{xx} etc.).

[0073] It can be shown (see Aarts et al, 2001) that (1) can be approximated with the recursion:

[0074] $$\hat{\rho}(k) = \hat{\rho}(k-1) + \gamma[\partial_k - \beta_k \hat{\rho}(k-1)] \quad (3)$$

$$\partial_k = 2x_k y_k$$

$$\beta_k = \alpha x_k^2 \alpha^{-1} y_k^2$$

[0075] Where

$$\alpha = \frac{y_{RMS}}{x_{RMS}} \quad (4)$$

$$\gamma = \frac{ce^{\eta}}{2x_{RMS} y_{RMS}}$$

[0077] The cross-correlation estimate using the above recursion is modified for block-wise processing rather than the sample-by-sample basis. This modification replaces the sample values (i.e. x(k) and y(k)) with values for the N-length block mean, i.e.

[0078]

$$x(k) = \frac{1}{N} \sum_{l=0}^{N-1} x(k-l)$$

[0079] Furthermore, the numerator for y is replaced with a small constant and so is α (replacing α with a constant effectively un-normalizes the correlation estimate). It is found that the modified un-normalized block-wise cross-correlation accurately estimates the cross-correlation compared with using the standard cross-correlation for two signals.

[0080] The modified block-wise fast cross-correlation algorithm, as summarized in FIG. 5, comprises the following steps:

1. A first signal buffer 502 is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated 504 (x_{RMS})
3. The mean level of the first buffer is calculated 506.
4. A second signal buffer 508 is accumulated. This signal buffer corresponds to the filtered ECM signal.
5. The RMS level of the second buffer is calculated 510 (y_{RMS}).
6. The mean level of the second buffer is calculated 512.
7. In step 514, γ (gamma) is approximated as:

$$\gamma = \frac{\Gamma}{2x_{RMS}y_{RMS}}$$

Where Γ is a small constant, e.g. 10E-3.

8. In step 516,∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.
9. In step 518, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.
10. In step 520, the new temporary estimate of the correlation newRho_temp is calculated as: newRho_temp=(delta-beta*rho_old))
11. In step 522, the new estimate of the correlation newRho is updated by summing the previous estimate of the correlation with the product of gamma and the temporary estimate of the correlation newRho_temp.
12. In step 524, the "old" value of the correlation is set to the newest correlation estimate, ready for the next iteration of the update algorithm.
13. In step 526, the current correlation estimate between the emitted test signal and the received and filtered ECM signal is set as equal to the value of newRho.

[0081] Referring to FIG. 6, a flowchart of a method to determine when to emit the test signal is shown. The test signal is emitted when the test can be performed unobtrusively to the user and also provides an accurate test. In at least one exemplary embodiment, a test event to determine if an earpiece is sealed correctly is

initiated via a timing methodology. In a first timing scenario, the test event occurs after a delay of a first predetermined time period when the RMS of the Audio Content (AC) is less than a RMS threshold. In a second timing scenario, the delay of the first predetermined time period is allowed to lapse without the test event occurring when the RMS of the audio content is greater than the RMS threshold. A second predetermined time period is started where the test event occurs when the RMS of the audio content is less than the RMS threshold. The test event is then initiated when the second predetermined time period is exceeded independent of the RMS of the audio content.

[0082] A test sequence is initiated in a step 602. The previous seal test event resets the first digital timer in a step 604. A time delay is generated by the loop comprising steps 606 and 608. The first digital timer is time incremented in the step 606. After each added time increment, the first digital timer is compared against a digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step 608 if the first digital timer is less than the digital_timer_threshold1.

[0083] A second digital timer is reset in a step 610 when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step 612. Audio content (AC) from a signal buffer is retrieved in a step 614. The audio content can be filtered through a low pass filter in an optional step 616. The RMS of the audio content is calculated in a step 618. The calculated RMS of the audio content is compared against an RMS_threshold 622 in a step 620. The second digital timer is compared against a digital_timer_threshold2 in a step 624 if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of 624.

[0084] The audio content signal is mixed with the test signal when the RMS of the audio content is less than the RMS_threshold in a step 626. Also, the audio content signal is mixed with the test signal when the second digital timer is greater than digital_timer_threshold2 in the step 624. The modified audio signal (having the test signal mixed in) is emitted by the ECR in a step 628 for testing the sealing section of the earpiece. The first digital timer is then reset in the step 604 to begin a timing sequence for another sealing section test.

[0085] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention. The estimated un-normalized cross-correlation between the ECM signal and the test signal (i.e. sine wave) is shown for different sine wave frequencies from 30-80Hz. Three different curves are provided corresponding

to a good fit (i.e. a tight optimal seal providing approximately 20-30 dB of acoustic attenuation), mid or partial seal (i.e. an ear-seal that could be characterized as "half in" providing approximately 10-15 dB of acoustic attenuation), and a poor seal (i.e. an ear-seal providing less than 10 dB of acoustic attenuation). At lower test frequencies, the change in correlation is more pronounced as the degree of ear seal fitting is changed from "good" to "mid" and "poor". From the data, the threshold used to determine whether the ear seal can be characterized as "good" is approximately -20 dB, (i.e. 0.85 of FIG. 10 which corresponds to the value for XCorr_threshold 414 in FIG. 4).

[0086] In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40Hz test tone, the cross-correlation for a "good" ear seal is -8 dB, and for a bad ear seal it is -68 dB (i.e. a 60 dB difference). At a test signal frequency of 80Hz, the cross-correlation for a "good" ear seal is -8 dB and for a bad ear seal it is -38 dB (i.e. a 30 dB difference). Thus, above 200Hz the cross- correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

[0087] Using the cross-correlation rather than a level differencing approach improves the accuracy and minimizes errors which occur due to user non-speech body noise, such as teeth chatter; sneezes, coughs, etc. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0088] FIG. 8 is a flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) in accordance with an exemplary embodiment. The IMS is adjusted depending on the degree of acoustic sealing provided by an earpiece. The method begins at step 802. The acoustic sealing is measured as disclosed in FIG. 7 and the result provided in a step 804 to determine the cross-correlation (XCorr) between a test signal and corresponding ECM signal. In general, the higher the cross-correlation, the higher the degree of acoustic sealing. An exemplary graph showing the relationship between XCorr and acoustic sealing is given in FIG. 10. The degree of acoustic sealing is determined from known XCorr using a look-up (or "hash") table or using a formula (e.g. of a polynomial form) that maps the acoustic sealing to the known XCorr value. The ambient sound level is measured in a step 806. The ambient sound level corresponds to

the noise level in proximity to the user. In general, a higher degree of attenuation is desired when the ambient sound levels are high. Conversely, at low ambient sound levels the attenuation level of the IMS may be less of an issue and comfort more of a factor. The IMS is adjusted in a step 808 to meet the attenuation needs. In general, inflating the IMS increases attenuation while deflating the IMS decreases attenuation. The method terminates at step 810.

[0089] Referring to FIG. 9, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) is shown. In general, the attenuation increases when the pressure in the IMS is raised thereby allowing a degree of control to make adjustments. For example, an adjustment is made to increase attenuation when the background noise level rises or a seal check produces a failed result. Adjustments are made until the seal check passes. The pressure level adjustments of the IMS will fall within a comfort range of a user (e.g., between 0.1 bar and 0.3 bar gauge pressure). Typically, the pressure level is set at a minimum level to achieve a predetermined attenuation level.

[0090] The method begins at step 902. The degree of acoustic sealing is determined from cross-correlation between the ECM signal and the generated test signal. The XCorr value is provided in a step 904. In step 906, the attenuation provided by the IMS is calculated (equation) or looked up (table) from data such as that shown in FIG. 10. In one exemplary embodiment, the desired attenuation value is dependant on the ambient sound level of the user. In another exemplary embodiment, the desired attenuation value is dependant on the ear-canal sound level of the user. In yet another exemplary embodiment, the desired attenuation value is dependant on the level of audio content (e.g. speech or music audio) reproduced with the earphone device. In all of the above examples, the desired attenuation value is determined by one or more of the embodiments in a step 907.

[0091] The difference between the degree of acoustic sealing determined in step 906 and the desired attenuation value determined in step 907 is calculated in step 908. The difference value in step 908 is used to determine the change in pressure of the IMS necessary to minimize the difference value in a step 910. In at least one exemplary embodiment, the difference value of the attenuation is converted into a corresponding pressure value change (e.g. in milli-Bars) using a similar look-up table or equation method as described previously. The pressure change in the IMS is then affected with step 912 to meet the desired attenuation level. For example if the desired attenuation is a decrease of 10 dB in sound across the earpiece in the ear canal, then a pressure of a variable volume inflatable system can have a gauge pressure of about 0.15 bar. If the desired attenuation is a decrease of 20 dB across the earpiece in the ear canal then the gauge pressure can be increased to about 0.25 bar, where an increased pressure is associated with an increase in attenuation. An

experimental table for each earpiece can be generated in a standard devised experimental setup (e.g. impedance tunnel) and referred to when changes are needed. The method ends at step 914.

[0092] Referring to FIG. 11, a flowchart of an exemplary method to determine a test signal fundamental is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user or made pleasant such that the user is unaware that the test signal is being played. This allows unobtrusive (periodic or non- periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0093] In at least one exemplary embodiment, an audio content 1102 is provided. A step 1104 stores audio content 1102 in a data buffer. In this example, audio content 1102 is music played from a media player and received via a wired or wireless connection to at least one earpiece in the user's ear. An alternate example would be that audio content 1102 is a speech audio signal from a portable telephone device or the like.

[0094] A step 1106 determines if the buffer of audio content 1102 comprises a strong tonal signal component. Mixing the test signal having a similar fundamental frequency as audio content 1102 will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

[0095] A step 1108 determines whether to update or generate the first fundamental tone for the test signal. The test signal is not updated or generated if buffered audio content 402 does not contain a strong tonal signal component. A return to step 1104 fills the buffer with the next audio content 402 for analysis.

[0096] A step 1110 analyzes the buffer of data of audio content 1102 when it has been determined that it contains a strong tonal signal component. Step 1110 determines the fundamental frequency of the tonal signal. The fundamental tone, often referred to as the fundamental and abbreviate f_0 is the lowest frequency in a harmonic series. The fundamental frequency (also called a natural frequency) of a periodic signal is the inverse of the pitch period length. The pitch period is the smallest repeating unit of a signal. The fundamental frequency of the tonal signal can be calculated using an autocorrelation analysis.

[0097] In one exemplary embodiment, a mathematical operation 1114 is performed where the frequency component of the test signal is limited to a frequency range below a lower minimum and upper maximum frequency range. Fund_ratio is calculated, which is defined as a ratio of the determined fundamental frequency (F_fund) of the tonal signal from step 1110 to an upper threshold value F_fund_threshold 1112, which in one exemplary embodiment, is a fixed constant equal to approximately 100Hz. In general,

F_fund_threshold 1112 is chosen to be a low frequency value which is above the lowest (or -3 dB) frequency that a transducer can reproduce, but below a predetermined frequency. In a comparison step 1116, if the estimated F-fund is higher than the F_fund_threshold 1112 (ratio > 1), then F_fund is reduced by an integer multiple to be below F_fund_threshold 1112 corresponding to the mathematical operation of step 1118. Otherwise, the test signal fundamental is equal to F_fund as shown in step 1120. Although not shown, the calculated test signal fundamental is compared and determined to be greater than a predetermined threshold.

[0098] Referring to FIG. 1, in at least one exemplary embodiment, processor 116 is configured to receive or generate audio content. As mentioned previously, the audio content can from external devices such as a portable phone or a media player. Memory 104 can be used as a buffer for the audio content. Processor 116 is configured to receive the buffer of audio content from memory 104. The steps and calculations of the block diagram of FIG. 4 are then performed by processor 116. The result being one of the identification of a strong tonal signal component in the buffer of audio content and the test signal fundamental or loading the buffer with new audio content and starting the process again.

[0099] Referring to FIG. 12, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step 1106 of FIG. 11 that analyzes audio content stored in a buffer. The method begins at step 1202. A step 1204 gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step 1206 filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20Hz to 500Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step 1208 analyzes the audio signal where a strong tonal signal component is represented by peaks in the analysis results. A step 1210 generates Absolute(Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute(Acorr) can be the square of the results from the auto-correlation.

[0100] A crest_factor_Acorr 1218 is generated from the results by calculating an RMS value 1214 (or time-averaged peak value) and peak value 1216 (or time averaged peak value). In at least one exemplary embodiment, the crest_factor_Acorr 1218 is the ratio of the peak value to the RMS value of an absolute auto-correlation sequence of the audio signal.

[0101] A comparison step 1222 is then performed. A strong tonal presence is identified when crest_factor_Acorr 1218 is greater than a threshold Crest_factor_Acorr_threshold 1220. Identification of the

strong tonal presence indicates the audio signal would facilitate masking of the test signal to determine sealing of the device (step 1226). The audio signal is not used in conjunction with the test signal if crest_factor_Acorr 1218 is less than Crest_factor_Acorr_threshold 1220 (step 1224). The process would begin again loading a next sequence of the audio signal into the buffer for review.

[0102] Referring to FIG. 1, as mentioned previously, audio content is stored in a buffer, for example memory 104. The audio content in the buffer is provided to processor 116. In at least one exemplary embodiment, processor 116, runs the analysis as described in the block diagram of FIG. 12 thereby determining if a strong tonal presence is found in the audio content in the buffer. New audio content is loaded into the buffer (memory 104) if a strong tonal presence is not found beginning the procedure again.

[0103] Referring to FIG. 13, a flowchart of a method to determine when to emit the test signal is shown. The method begins at step 1302. The test signal is emitted when the test can be performed unobtrusively to the user and also provide an accurate test. In a step 1304, an audio signal is retrieved from a buffer. In at least one exemplary embodiment, the audio signal is received from an ECM or an ASM. The audio signal is measured to determine when the sound level is low in the ear canal, the ambient environment, or both. In general, the test signal is emitted when the sound level is low.

[0104] A filter step 1306 band pass filters the audio signal. In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150Hz which corresponds to a frequency range of the test signal. In a step 1308, the RMS of the audio signal is calculated. The audio signal is analyzed to detect when the energy within an audio frequency range is below a threshold RMS_threshold 1310. The RMS of the audio signal is the signal level in the volume being measured. A comparison step 1312 compares the measured RMS level of the filtered audio signal against RMS_threshold 1310. In a step 1314, a test signal is emitted when the measured RMS value is less than RMS_threshold 1310. No test signal is emitted when the RMS of the audio signal is greater than RMS_threshold 1310. The method ends at step 1316.

[0105] Referring to FIG. 14, a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the

earpiece to ensure that the seal is within normal operating specifications.

[0106] The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

[0107] In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

[0108] A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity of the signals in the first and second volumes. In particular, the non- difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

[0109] In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

[0110] Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the

predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a second comparison step 1428, the Magnitude of Peak is compared against a Peak_threshold 1430. If the Magnitude of Peak is greater than the Peak_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

[0111] In at least one exemplary embodiment and referring briefly to FIG. 1, an earpiece is tested to determine if sealing unit 108 is sealed correctly to the ear canal of the user. Sealing unit 108 creates a first volume in ear canal 124 and a second volume outside the ear canal 124 in the ambient environment. A masking approach is used to perform seal testing unobtrusively to the user. The user is listening to music or speech (audio content) provided to ear canal124 from ECR 114.

[0112] The music or speech is buffered in memory 104 or memory in processor 116. Processor 116 analyzes the audio content in the buffer to identify a strong tonal content. A test signal can be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor 116 also analyzes the measured signals from ECM 106 and ASM 120 to determine when to emit the test signal. Processor 116 monitors and compares the sound level in the ambient environment and ear canal124. Processor 116 will provide the generated test signal to ECR 114 during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor 116 will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

[0113] Processor 116 monitors the test conditions and then provides the test signal to ECR 114 when an accurate sealing test can be performed. ECR 114 outputs an acoustic test signal which may or may not have other audio content. ECM 106 and ASM 120 respectively measure acoustic signals in ear canal 124 and the ambient environment. Processor 116 is operatively coupled to ECM 106 and ASM 120. The measured signals are buffered in memory 104.

[0114] In an exemplary embodiment, a cross-correlation is used to measure the similarity between the

signals in ear canal124 and the ambient environment. Processor 116 performs the cross-correlation calculations using the measured acoustic signals from ECM 106 and ASM 120. In particular, the cross-correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$XCorr(n,l) = \sum_{n=0}^{N} ASM(n)ECM(n-l), \quad (5)$$

Where: l=0, 1, 2, … N

[0115] Where ASM(n) is the nth sample of the ASM signal, and ECM(n- l) is the (n- l)th sample of the ECM signal. A peak of the absolute cross-correlation is estimated using a peak- picking function and also the lag time at which this peak occurs (i.e. the index I at which this occurs). Thus, the Lag-time of Peak and the Magnitude of Peak are known and respectively compared against a Target Lag Limit range and a Peak Threshold. The user of the earpiece is notified or warned that sealing unit 108 is improperly sealed by processor 116 if the measured Lag-time of Peak is within the Target Lag Limit range and the Magnitude of Peak is greater than the Peak Threshold.

[0116] Like Correlation and Cross-Correlation, a coherence function is also a measure of similarity between two signals. Coherence is another non-difference comparison approach that can be used for detecting acoustic seal integrity. Coherence is defined as:

$$\gamma^2{}_{xy} = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (6)$$

[0117] Where Gxy is the cross-spectrum of two signals (e.g. the ASM and ECM signals), and can be calculated by first computing the cross-correlation in equation (5), applying a window function, for example a Hanning window, and transforming to the frequency domain, for example via an FFT. Gxx or Gyy is the auto-power spectrum of either the ASM or ECM signals, and can be calculated by first computing the auto-correlation (using equation 5, but where the two input signals are both from either the ASM or ECM and transforming to the frequency domain. The coherence function gives a frequency-dependent vector between 0 and 1, where a high coherence at a particular frequency indicates a high degree of coherence at this frequency, and can therefore be used to analyze test signal frequencies in the ASM and ECM signals whereby a high coherence indicates the presence of the test signal in the ambient environment (indicating

leakage past the acoustic barrier).

[0118] Other approaches such as frequency spectrum analysis and RMS levels can also be used to determine if the earpiece is sealed correctly. Using a non-difference comparison approach such as coherence or cross-correlation between the ASM and ECM signals to determine sealing is more reliable than taking the level difference of the ASM and ECM signals. Using the cross correlation rather than a level differencing approach improves the accuracy and minimizes errors which may occur due to user non-speech body noise, such as teeth chatter, sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0119] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., 1500) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. 15A and 15B shows a slider actuator 1500, with a moving slide 1510, having attached a pumping arm 1560, with a hole 1550 covering bump 1520. Upon receipt of a signal1505 the actuator can move 1570 such that the bump 1520 covers the hole 1550 in a bellows 1530 pump. The actuation compresses the miniature bellows 1530 pushing 1590 gas through the one-way valve 1540, upon the back stroke the bump 1520 uncovers the hole 1550 and air rushes back into the bellows 1530 for the next pump. For example if the stroke length is 2 mm and the pump arm 1560 contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system 1570 needs to be inflated more (e.g., more gas to increase sealing) then each stroke can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system 1580. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal 1505 sent, which can be specifically tailored depending

upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation systems can be used and controlled by signal 1505.

[0120] FIG. 16 illustrates a block diagram of controlled sealing in response to a seal fail signal and/or a request for increased seal attenuation. For example, a command signal 1600 is received (e.g., a seal fail signal in which a default attenuation value is attached for example 15 dB, or a signal requesting an additional amount of attenuation) by a processor. The command signal specifies an additional amount of attenuation or that the seal has failed. An attenuation needed N is identified 1610. For example if the current attenuation is 5 dB loss at f=500 Hz, at a pressure of 0.1 bar, and a command signal is received requesting a 10 dB loss at 500 Hz, then an experimental table is queried to find the pressure needed P which is then subtracted from the current pressure to obtain an increase in pressure DP needed. The increase in pressure is converted into a volume of gas increase needed (e.g., again referring to experimental tables based upon the inflation system volume). The volume of gas increase needed can then be directly linked with the number of cycles of an actuator pump M, 1620. The number of requested cycles **M** can then be sent 1630 to the actuator control circuit to pump the designated number of cycles. The system can then retest the attenuation 1640 and if refinements are needed the process can start again.

[0121] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

[0122] Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of U.S. Patent Application Ser. No. 16/838,277, filed April 2, 2020, which is a continuation of U.S. Patent Application Ser. No. 16/414,136, filed May 16, 2019, which is a continuation of U.S. Patent Application Ser. No. 15/700,511, filed September 11, 2017, which is a continuation of U.S. Patent Application Ser. No. 14/827,332, filed August 17, 2015, now U.S. Patent No. 9,781,530, which is a continuation of U.S. Patent Application Ser. No. 14/054,015, filed October 15, 2013, now U.S. Patent No. 9,113,267, which is a Divisional Application of U.S. Application Ser. No. 12/555,864, filed September 9, 2009, now U.S. Patent No. 8,600,067 and claims the benefit of U.S. Provisional Patent Application No. 61/098,250 filed September 19, 2008. The disclosure of all the aforementioned references are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to testing the seal of an orifice-inserted device, and more particularly, though not exclusively, to a device and method for determining if an earpiece is sealed correctly in an ear canal.

BACKGROUND OF THE INVENTION

[0003] It can be difficult to communicate using an earpiece or earphone device in the presence of high-level background sounds. In many earpiece designs a transducer is placed near the ear canal opening. Ambient sound from the surrounding environment enters the ear canal with the audio content from the transducer. Environmental sounds such as traffic, construction, and nearby conversations can degrade the quality of the audio content.

[0004] Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient

sound to provide adequate performance to the user. Additionally, user conditions may change substantially

during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

[0005] Broadly stated, embodiments are directed to a device and method to determine if an earpiece is sealing within the design specification of the device.

[0006] In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume. The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

[0007] At least one exemplary embodiment is directed to a method of detecting sealing integrity of an earpiece comprising the steps of: providing a test signal; generating an acoustic signal corresponding to the test signal incident on an ear canal side of a sealing section; converting the acoustic signal incident on a first side of the sealing section to an electrical signal; and cross- correlating the test signal to the electrical signal where the earpiece is sealed correctly when a cross-correlation between the test signal and the electrical signal is above a threshold.

[0008] At least one exemplary embodiment is directed to a method of adjusting attenuation of an earpiece comprising the steps of: relating cross-correlation of a test signal and a measured acoustic signal in an ear canal of a user to an attenuation level of a sealing section of the earpiece; comparing the attenuation level of the sealing section of the earpiece to a minimum attenuation value; and adjusting a pressure of the sealing section to meet the minimum attenuation value.

[0009] At least one exemplary embodiment is directed to a device comprising: a sealing section configured to seal a user's orifice, where the sealing section is configured to produce an acoustic seal between a first side of the sealing section and a second side of the sealing section; a transducer configured to generate a first acoustic signal incident on the first side of the sealing section; and a first microphone configured to measure a second acoustic signal incident on the second side of the sealing section, where the second acoustic signal

includes at least a portion of the first acoustic signal that has passed from the first side to the second side of the sealing section where the first acoustic signal is compared to the second acoustic signal to determine if the sealing section is sealed.

[0010] Further areas of applicability of exemplary embodiments of the present invention will become apparent from the detailed description provided hereinafter. It should be understood that the detailed description and specific examples, while indicating exemplary embodiments of the invention, are intended for purposes of illustration only and are not intended to limit the scope of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0012] FIG. 1 is a diagram of an earpiece inserted in an ear canal in accordance with an exemplary embodiment;

[0013] FIG. 2 is a block diagram of optional components of an earpiece in accordance with an exemplary embodiment;

[0014] FIG. 3 is a flowchart of a method for checking an ear seal in accordance with an exemplary embodiment;

[0015] FIG. 4 is a flowchart to determine acoustic seal integrity of an earpiece in accordance with the exemplary embodiment.

[0016] FIG. 5 is a flowchart of a method to estimate the instantaneous cross-correlation between a first and second audio signal.

[0017] FIG. 6 is a flowchart to determine when to emit a test signal in accordance with an exemplary embodiment;

[0018] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention;

[0019] FIG. 8 is a block diagram for a method of adjusting the IMS system in accordance with at least one exemplary embodiment;

[0020] FIG. 9 is a block diagram for a method of adjusting IMS pressure in accordance with at least one exemplary embodiment;

[0021] FIG. 10 illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with at least one exemplary embodiment;

[0022] FIG. 11 illustrates a flowchart of an exemplary method to determine a test signal fundamental;

[0023] FIG. 12 illustrates a flowchart of an exemplary embodiment to determine tonal presence in audio content;

[0024] FIG. 13 illustrates a flowchart of a method to determine when to emit the test signal;

[0025] FIG. 14 illustrates a flowchart of an exemplary method to determine acoustic seal integrity;

[0026] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment; and

[0027] FIG. 16 illustrates the sending of a signal to an inflation controller upon detection of a seal fail to modify the seal pressure in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION

[0028] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0029] Exemplary embodiments are directed to or can be operatively used on various wired or wireless orifice inserted devices for example earpiece devices (e.g., earbuds, headphones, ear terminals, behind the ear devices or other acoustic devices as known by one of ordinary skill, and equivalents).

[0030] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0031] Additionally exemplary embodiments are not limited to earpieces, for example some functionality can be implemented on other systems with speakers and/or microphones for example computer systems, PDAs, BlackBerry® smart phones, cell and mobile phones, and any other device that emits or measures acoustic energy. Additionally, exemplary embodiments can be used with digital and non-digital acoustic systems. Additionally various receivers and microphones can be used, for example MEMs transducers,

diaphragm transducers, for example Knowles' FG and EG series transducers.

[0032] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0033] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0034] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for sealing or partially sealing an ear. FIG. 1 is a diagram of an earpiece inserted in an ear canal 124 in accordance with at least one exemplary embodiment of the invention. FIG. 1 also illustrates portions of the ear including pinna 128, ear canal 124 and eardrum 126. As illustrated, the earpiece comprises an electronic housing unit 100 and a sealing unit 108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 124 of a user 130. The earpiece is an in-ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude the ear canal 124.

[0036] The earpiece includes an Ambient Sound Microphone (ASM) 120 to capture ambient sound, an Ear Canal Receiver (ECR) 114 to deliver audio to an ear canal124, and an Ear Canal Microphone (ECM) 106 to capture and assess a sound exposure level within the ear canal 124. The earpiece can partially or fully occlude the ear canal 124 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 124, and to form an acoustic seal with the walls of the ear canal 124 at a location between the entrance to the ear canal 124 and the tympanic membrane (or ear drum) 126. In general, such a seal is typically achieved by means of a soft and compliant housing of the sealing unit 108. Additionally, the sealing unit 108 can be a pressurized expandable element that fills a portion of the available local space.

[0037] Sealing unit 108 is an acoustic barrier having a first side corresponding to ear canal124 and a second

side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 108

includes an ear canal microphone tube 110 and an ear canal receiver tube 112. Sealing unit 108 creates a closed cavity of approximately 5 cc or less between the first side of sealing unit 108 and the tympanic membrane 126 in ear canal124. In at least one exemplary embodiment the sealing facilitates using the ECR (speaker) 114 to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 126 resulting from the sound field at the entrance to the ear canal 124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0038] In at least one exemplary embodiment and in broader context, the second side of sealing unit 108 corresponds to the side adjacent to electronic housing unit 100. Ambient sound microphone 120 is housed in electronic housing unit 100 and is exposed to the ambient environment for receiving sound from the ambient environment around the user.

[0039] The electronic housing unit 100 can include various system components such as a microprocessor 116, memory 104, battery 102, ECM 106, ASM 120, ECR, 114, and user interface 122, or these components can reside in a separate system or interface operatively connected. Microprocessor 116 (or processor 116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 116 is operatively coupled to memory 104, ECM 106, ASM 120, ECR 114, and user interface 122. An optional wire 118 can provide an external connection to the earpiece. Battery 102 powers the circuits and transducers of the earpiece. Battery 102 can be a rechargeable or replaceable battery.

[0040] In at least one exemplary embodiment, electronic housing unit 100 is adjacent to sealing unit 108. Openings in electronic housing unit 100 receive ECM tube 110 and ECR tube 112 to respectively couple to ECM 106 and ECR 114. ECR tube 112 and ECM tube 110 acoustically couple signals to and from ear canal 124. For example, ECR 114 outputs an acoustic signal through ECR tube 112 and into ear canal 124 where it is received by the tympanic membrane 126 of the user of the earpiece. Conversely, ECM 106 receives an acoustic signal present in ear canal124 though ECM tube 110.

[0041] One function of ECM 106 is that of measuring the sound pressure level in the ear canal cavity 124 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece. In one arrangement, ASM 120 is used to monitor sound pressure at the entrance to the occluded or partially occluded ear canal 124. All transducers shown can receive or transmit

audio signals to a processor 116 that undertakes audio signal processing and provides a transceiver for audio

via the wired (wire 118) or a wireless communication path. Note also that the acoustic signals can be stored for later retrieval.

[0042] In at least one exemplary embodiment the earpiece can be constructed to actively monitor a sound pressure level both inside and outside an ear canal 124. In at least one exemplary embodiment monitored data can be used to enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. In at least one exemplary embodiment an earpiece can facilitate at least one of conducting listening tests, filtering sounds in the environment, monitoring warning sounds in the environment, presenting notification based on identified warning sounds, maintaining constant audio content to ambient sound levels, and filtering sound in accordance with a Personalized Hearing Level (PHL).

[0043] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 124 using ECR 114 and ECM 106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 120. For instance, the ECR 114 can deliver an impulse within the ear canal124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. In at least one exemplary embodiment the earpiece can use either the ASM 120 or the ECM 106 to monitor the sound pressure level, which can then be used in a Sound Pressure Level Dosimeter calculation, to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0044] Referring to FIG. 2, a block diagram of an earpiece 201 in accordance with an exemplary embodiment is shown. A power supply 205 (e.g., USB power connection, hearing aid battery (batteries)) powers components of the earpiece 201 including microprocessor 206 (or processor 206, e.g., Texas Instruments TMS320C6713) and a data communication system 216 (e.g., RF or Bluetooth communication chip). As illustrated, the earpiece 201 includes the processor 206 operatively coupled to data communication system 216, ASM 210, ECR 212, and ECM 208. Data communication system 216 may include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 202 and Read Only Memory (ROM) 204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 206. The processor 206 can also include a clock to record a time stamp.

[0045] In general, data communication system 216 is a communication pathway to components of the earpiece 201 and components external to the earpiece 201. The communication link can be wired or

wireless. In at least one exemplary embodiment, data communication system 216 is configured to communicate with ECM 208, ASM 210, visual display 218, and user control interface 214 of the earpiece 201. As shown, user control interface 214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 216 is capable of communication to devices exterior to the earpiece 201 such as the user's mobile phone 234, a second earpiece 222, and a portable media player 228. Portable media player 228 can be controlled by a manual user control 230.

[0046] The user's mobile phone 234 includes a mobile phone communication system 224. A microprocessor 226 is operatively coupled to mobile phone communication system 224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 220 worn by another person to the user's mobile phone. Similarly, the user's mobile phone 234 can be connected to the data communication system 216 of the earpiece 201 as well as the second earpiece 222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 234 through their respective earpieces.

[0047] As illustrated, a data communication system 216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

[0048] The RAM 202 stores program instructions for execution on the processor 206 as well as captured audio processing data. For instance, memory RAM 202 and ROM 204 can be off-chip and external to the processor 206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. In at least one exemplary embodiment, the data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer is operatively connected with processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0049] Data communication system 216 includes an audio interface operatively coupled to the processor

206 and the VOX to receive audio content, for example from portable media player 228, a cell phone, or

any other communication device, and deliver the audio content to the processor 206. The processor 206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 206 (or the VOX of data communication system 216) can lower a volume of the audio content responsive to detecting an event for transmitting the acute sound to the ear canal of the user. The processor 206 by way of the ECM 208 can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 216.

[0050] The earpiece 201 and data communication system 216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading over-the-air to the earpiece 201. It should be noted also that next generation access technologies can also be used in exemplary embodiments.

[0051] Data communication system 216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 201.

[0052] The power supply 205 utilizes common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 201 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 205 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0053] Microprocessor 206 is operatively connected with an EarSeal Inflation Management System 232 to control the degree to which the sealing unit 108 is inflated or deflated. In one exemplary embodiment, sealing unit 108 comprises an expandable element (e.g., inflatable balloon mechanism), whereby a cavity can be filled with air or a liquid to change the degree of acoustic isolation between the internal ear canal space 124 and the ambient environment. Alternately, a passive system for sealing ear canal 124 is used such as a flexible rubber or a silicon sealing unit or a foam plug. In one exemplary embodiment, the passive

system is a balloon mechanism that is filled with air or liquid. The balloon mechanism conforms to the

shape and size of an ear canal and includes a restorative force module that applies a pressure to the balloon mechanism for sealing the ear canal cavity.

[0054] The earpiece is a single operational device or a family of devices configured in a master- slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece are reused in different form factors for the master and slave devices.

[0055] Referring to FIG. 3, a flowchart illustrates a method for an acoustic sealing analysis system in accordance with an exemplary embodiment. In general, a first volume is acoustically isolated from a second volume. The test determines if the two volumes have sufficient acoustic isolation from one another. For example, cars are designed to have a quiet interior. Users of an automobile do not want to be subjected to the noise of the external environment. Thus, a car interior (first volume) is acoustically isolated from the external environment outside of the automobile. Similarly, an earpiece having a sealing unit such as described in FIG. 1 will create a first volume (the ear canal) that is acoustically isolated from the ambient environment of the user (second volume). In either example, the acoustic sealing analysis system determines if there is sufficient acoustic isolation for the application. In the earpiece example, random or periodic testing of the seal may be beneficial because a new seal is formed in the ear canal when the device is put in the ear or it may shift over time depending on user activity.

[0056] The method begins at step 302. A test signal is acquired in a step 304. For example, the test signal can be stored in memory or generated by a microprocessor. The test signal is provided to the acoustic transducer. The acoustic transducer or loudspeaker (such as an ECR) emits an acoustic signal corresponding to the test signal within the first volume in a step 306. The acoustic field in the first volume is detected by an Ear Canal Microphone (ECM) in a step 308. The acoustic loading on both the ECR and ECM will change depending on the degree of acoustic sealing, thereby affecting the degree magnitude of the radiated ECR signal detected by the ECM. In general, as the degree of ear seal decreases, the effect of lumped air mass coupled to the ECR and ECM will decrease thereby increasing in Thevenin capacitance, which effectively reduces the transfer of low frequency emitted sound from the ECR to the ECM.

[0057] In one exemplary embodiment, the test signal and the acoustic signal emitted by the loudspeaker into the first volume is a single frequency sine wave signal for testing leakage from one volume to another.

[0058] The degree of sealing between the first and second acoustic volumes is determined in a step 310 and

the process ends at step 312. The cross-correlation between the emitted test signal and detected ECM signal is taken. In at least one exemplary embodiment, the test signal and the measured acoustic signal emitted by the loudspeaker are conditioned using a time delay and frequency dependent filter. The ear-seal is determined to be low (or "leaky") if the cross- correlated signals are below a predetermined value. In at least one exemplary embodiment, automatic adjustments to the sealing section are made (such as deflating and re-inflating the sealing balloon to reseal the sealing section including retesting). Alternately, an audible sound, vocal response, or visual response can be provided to let the user know that the earpiece is sealed correctly or incorrectly.

[0059] Referring to FIG. 1, the earpiece is used as an example to illustrate a test sequence as disclosed in FIG. 3. Sealing unit 108 occludes an opening of ear canal 124 creating a first volume (ear canal124) and a second volume (the ambient environment). Sealing unit 108 has a first side exposed to ear canal 124 and a second side is exposed to or corresponds to the ambient environment external to the ear.

[0060] In at least one exemplary embodiment, processor 116 is configured to receive a test signal in memory 104. Processor 116 generates the test signal and provides the test signal to Ear Canal Receiver 114 (ECR 114). ECR 114 emits the test signal into Ear Canal Receiver Tube (ECR Tube 112). The test signal propagates through ECR tube 112 and into ear canal 124. Ear Canal Microphone tube 110 (ECM tube 110) is configured to receive an acoustic signal incident on the first side of sealing unit 108. The test signal in ear canal 124 propagates through ECM tube 110 and is received by Ear Canal Microphone 106 (ECM 106). ECM 106 is configured to measure the test signal in ear canal124 and provide the measured test signal to processor 116.

[0061] As shown, electronic housing unit 100 of the earpiece is adjacent to the second side of sealing unit 108. Electronic housing unit 100 is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit 108. Electronic housing unit 100 includes Ambient Sound Microphone 120 (ASM 120), which is configured to measure sounds in the ambient environment. Thus, ASM 120 receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit 108. ASM 120 provides the measured ambient signal to processor 116.

[0062] Ideally, sealing unit 108 is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit 108 and into the ambient environment. Conversely, sealing unit 108 if improperly sealed will pass some of the test signal. Processor 116 compares the test signal to the signal

provided by ECM 106 corresponding to the acoustic signal in ear canal124. In particular, processor 116

undertakes the cross-correlation between emitted test signal and the ECM signal.

[0063] Referring to FIG. 4, a flowchart of an exemplary method to determine the acoustic seal integrity of an earpiece in accordance with an exemplary embodiment is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user. This allows unobtrusive (periodic or non-periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0064] In at least one exemplary embodiment, an audio content is provided in a step 402. A step 404 stores the test signal in a test signal data buffer. For example, a single frequency sine wave is stored in the test signal data buffer. The output (or alternatively--input) of the test signal data buffer is optionally delayed by digital delay unit 406. The function of delay unit 406 is to time- align the emitted test signal with the ECM signal so the cross-correlation is sensitive to changes in ear seal.

[0065] A step 408 stores the test ECM signal in a test signal data buffer. The output (or alternatively--input) of the ECM signal buffer can be filtered with a low-pass filter 410. The low pass filter can be configured so that the pass-band covers the frequency of the test signal. In one exemplary configuration, the low-pass filter can be a cascaded bi-quad IIR type filter with the cut-off frequency equal to 10Hz greater than the test signal frequency.

[0066] A step 412 cross-correlates the optionally delayed test signal buffer with the low-pass filtered ECM signal buffer. An exemplary method for the cross-correlation algorithm is described in FIG. 5. The instantaneous cross-correlation (i.e. the cross-correlation at zero-lag) value from step 412 is compared with the cross-correlation threshold value 414 using comparator unit 416. If the instantaneous cross-correlation of the two signal buffers is less than the threshold value 414, then the seal test status is set to FAIL 418 (i.e. an ear-seal leak is detected); otherwise, if the cross-correlation is suitably high, the seal test status is set to PASS 420.

[0067] Referring to FIG. 5, a flowchart of an exemplary embodiment to determine the instantaneous cross-correlation between a first audio signal and a second audio signal is illustrated. The process begins at step 500. In at least one exemplary embodiment, the first audio signal is the test signal (i.e. a sine wave) and the second signal is the low-pass-filtered ECM signal.

[0068] The correlation between two signals x and y at time k using an exponential window is defined as:

[0069] $$p(k) = \frac{S_{xy}(k)}{\sqrt{S_{xx}(k)_{yy}(k)}} \quad (1)$$

[0070] Where

[0071] $$S_{xy}(k) = \sum_{l=0}^{\infty} ce^{-\eta^l} x_{k-l} y_{k-l} \quad (2)$$

$$c = 1 - e^{-\eta}$$

[0072] And S_{xx} and S_{yy} are defined similarly as in (2) (replacing y with x for S_{xx} etc.).

[0073] It can be shown (see Aarts et al, 2001) that (1) can be approximated with the recursion:

[0074] $$\hat{\rho}(k) = \hat{\rho}(k-1) + \gamma[\partial_k - \beta_k \hat{\rho}(k-1)] \quad (3)$$

$$\partial_k = 2x_k y_k$$

$$\beta_k = \alpha x_k^2 \alpha^{-1} y_k^2$$

[0075] Where

$$\alpha = \frac{y_{RMS}}{x_{RMS}} \quad (4)$$

$$\gamma = \frac{ce^{\eta}}{2x_{RMS} y_{RMS}}$$

[0077] The cross-correlation estimate using the above recursion is modified for block-wise processing rather than the sample-by-sample basis. This modification replaces the sample values (i.e. x(k) and y(k)) with values for the N-length block mean, i.e.

[0078]

$$x(k) = \frac{1}{N} \sum_{l=0}^{N-1} x(k-l)$$

[0079] Furthermore, the numerator for y is replaced with a small constant and so is α (replacing α with a constant effectively un-normalizes the correlation estimate). It is found that the modified un-normalized

block-wise cross-correlation accurately estimates the cross-correlation compared with using the standard cross-correlation for two signals.

[0080] The modified block-wise fast cross-correlation algorithm, as summarized in FIG. 5, comprises the following steps:

1. A first signal buffer 502 is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated 504 (x_{RMS})
3. The mean level of the first buffer is calculated 506.
4. A second signal buffer 508 is accumulated. This signal buffer corresponds to the filtered ECM signal.
5. The RMS level of the second buffer is calculated 510 (y_{RMS}).
6. The mean level of the second buffer is calculated 512.
7. In step 514, γ (gamma) is approximated as:

$$\gamma = \frac{\Gamma}{2x_{RMS}y_{RMS}}$$

Where Γ is a small constant, e.g. 10E-3.

8. In step 516,∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.
9. In step 518, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.
10. In step 520, the new temporary estimate of the correlation newRho_temp is calculated as: newRho_temp=(delta-beta*rho_old))
11. In step 522, the new estimate of the correlation newRho is updated by summing the previous estimate of the correlation with the product of gamma and the temporary estimate of the correlation newRho_temp.
12. In step 524, the "old" value of the correlation is set to the newest correlation estimate, ready for the next iteration of the update algorithm.
13. In step 526, the current correlation estimate between the emitted test signal and the received and filtered ECM signal is set as equal to the value of newRho.

[0081] Referring to FIG. 6, a flowchart of a method to determine when to emit the test signal is shown. The test signal is emitted when the test can be performed unobtrusively to the user and also provides an accurate test. In at least one exemplary embodiment, a test event to determine if an earpiece is sealed correctly is

initiated via a timing methodology. In a first timing scenario, the test event occurs after a delay of a first predetermined time period when the RMS of the Audio Content (AC) is less than a RMS threshold. In a

second timing scenario, the delay of the first predetermined time period is allowed to lapse without the test event occurring when the RMS of the audio content is greater than the RMS threshold. A second predetermined time period is started where the test event occurs when the RMS of the audio content is less than the RMS threshold. The test event is then initiated when the second predetermined time period is exceeded independent of the RMS of the audio content.

[0082] A test sequence is initiated in a step 602. The previous seal test event resets the first digital timer in a step 604. A time delay is generated by the loop comprising steps 606 and 608. The first digital timer is time incremented in the step 606. After each added time increment, the first digital timer is compared against a digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step 608 if the first digital timer is less than the digital_timer_threshold1.

[0083] A second digital timer is reset in a step 610 when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step 612. Audio content (AC) from a signal buffer is retrieved in a step 614. The audio content can be filtered through a low pass filter in an optional step 616. The RMS of the audio content is calculated in a step 618. The calculated RMS of the audio content is compared against an RMS_threshold 622 in a step 620. The second digital timer is compared against a digital_timer_threshold2 in a step 624 if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of 624.

[0084] The audio content signal is mixed with the test signal when the RMS of the audio content is less than the RMS_threshold in a step 626. Also, the audio content signal is mixed with the test signal when the second digital timer is greater than digital_timer_threshold2 in the step 624. The modified audio signal (having the test signal mixed in) is emitted by the ECR in a step 628 for testing the sealing section of the earpiece. The first digital timer is then reset in the step 604 to begin a timing sequence for another sealing section test.

[0085] FIG. 7 is a graph illustrating different seal measurements in accordance with the present invention. The estimated un-normalized cross-correlation between the ECM signal and the test signal (i.e. sine wave) is shown for different sine wave frequencies from 30-80Hz. Three different curves are provided corresponding

to a good fit (i.e. a tight optimal seal providing approximately 20-30 dB of acoustic attenuation), mid or partial seal (i.e. an ear-seal that could be characterized as "half in" providing approximately 10-15 dB of

acoustic attenuation), and a poor seal (i.e. an ear-seal providing less than 10 dB of acoustic attenuation). At lower test frequencies, the change in correlation is more pronounced as the degree of ear seal fitting is changed from "good" to "mid" and "poor". From the data, the threshold used to determine whether the ear seal can be characterized as "good" is approximately -20 dB, (i.e. 0.85 of FIG. 10 which corresponds to the value for XCorr_threshold 414 in FIG. 4).

[0086] In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40Hz test tone, the cross-correlation for a "good" ear seal is -8 dB, and for a bad ear seal it is -68 dB (i.e. a 60 dB difference). At a test signal frequency of 80Hz, the cross-correlation for a "good" ear seal is -8 dB and for a bad ear seal it is -38 dB (i.e. a 30 dB difference). Thus, above 200Hz the cross- correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

[0087] Using the cross-correlation rather than a level differencing approach improves the accuracy and minimizes errors which occur due to user non-speech body noise, such as teeth chatter; sneezes, coughs, etc. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0088] FIG. 8 is a flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) in accordance with an exemplary embodiment. The IMS is adjusted depending on the degree of acoustic sealing provided by an earpiece. The method begins at step 802. The acoustic sealing is measured as disclosed in FIG. 7 and the result provided in a step 804 to determine the cross-correlation (XCorr) between a test signal and corresponding ECM signal. In general, the higher the cross-correlation, the higher the degree of acoustic sealing. An exemplary graph showing the relationship between XCorr and acoustic sealing is given in FIG. 10. The degree of acoustic sealing is determined from known XCorr using a look-up (or "hash") table or using a formula (e.g. of a polynomial form) that maps the acoustic sealing to the known XCorr value. The ambient sound level is measured in a step 806. The ambient sound level corresponds to

the noise level in proximity to the user. In general, a higher degree of attenuation is desired when the ambient sound levels are high. Conversely, at low ambient sound levels the attenuation level of the IMS

may be less of an issue and comfort more of a factor. The IMS is adjusted in a step 808 to meet the attenuation needs. In general, inflating the IMS increases attenuation while deflating the IMS decreases attenuation. The method terminates at step 810.

[0089] Referring to FIG. 9, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management System (IMS) is shown. In general, the attenuation increases when the pressure in the IMS is raised thereby allowing a degree of control to make adjustments. For example, an adjustment is made to increase attenuation when the background noise level rises or a seal check produces a failed result. Adjustments are made until the seal check passes. The pressure level adjustments of the IMS will fall within a comfort range of a user (e.g., between 0.1 bar and 0.3 bar gauge pressure). Typically, the pressure level is set at a minimum level to achieve a predetermined attenuation level.

[0090] The method begins at step 902. The degree of acoustic sealing is determined from cross-correlation between the ECM signal and the generated test signal. The XCorr value is provided in a step 904. In step 906, the attenuation provided by the IMS is calculated (equation) or looked up (table) from data such as that shown in FIG. 10. In one exemplary embodiment, the desired attenuation value is dependant on the ambient sound level of the user. In another exemplary embodiment, the desired attenuation value is dependant on the ear-canal sound level of the user. In yet another exemplary embodiment, the desired attenuation value is dependant on the level of audio content (e.g. speech or music audio) reproduced with the earphone device. In all of the above examples, the desired attenuation value is determined by one or more of the embodiments in a step 907.

[0091] The difference between the degree of acoustic sealing determined in step 906 and the desired attenuation value determined in step 907 is calculated in step 908. The difference value in step 908 is used to determine the change in pressure of the IMS necessary to minimize the difference value in a step 910. In at least one exemplary embodiment, the difference value of the attenuation is converted into a corresponding pressure value change (e.g. in milli-Bars) using a similar look-up table or equation method as described previously. The pressure change in the IMS is then affected with step 912 to meet the desired attenuation level. For example if the desired attenuation is a decrease of 10 dB in sound across the earpiece in the ear canal, then a pressure of a variable volume inflatable system can have a gauge pressure of about 0.15 bar. If the desired attenuation is a decrease of 20 dB across the earpiece in the ear canal then the gauge pressure can be increased to about 0.25 bar, where an increased pressure is associated with an increase in attenuation. An

experimental table for each earpiece can be generated in a standard devised experimental setup (e.g. impedance tunnel) and referred to when changes are needed. The method ends at step 914.

[0092] Referring to FIG. 11, a flowchart of an exemplary method to determine a test signal fundamental is illustrated. In at least one embodiment of an acoustic sealing analysis system, the test signal is masked or used in a manner undetectable by the user or made pleasant such that the user is unaware that the test signal is being played. This allows unobtrusive (periodic or non- periodic) testing to determine if a device is sealed correctly ensuring optimum system performance and more importantly user safety.

[0093] In at least one exemplary embodiment, an audio content 1102 is provided. A step 1104 stores audio content 1102 in a data buffer. In this example, audio content 1102 is music played from a media player and received via a wired or wireless connection to at least one earpiece in the user's ear. An alternate example would be that audio content 1102 is a speech audio signal from a portable telephone device or the like.

[0094] A step 1106 determines if the buffer of audio content 1102 comprises a strong tonal signal component. Mixing the test signal having a similar fundamental frequency as audio content 1102 will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

[0095] A step 1108 determines whether to update or generate the first fundamental tone for the test signal. The test signal is not updated or generated if buffered audio content 402 does not contain a strong tonal signal component. A return to step 1104 fills the buffer with the next audio content 402 for analysis.

[0096] A step 1110 analyzes the buffer of data of audio content 1102 when it has been determined that it contains a strong tonal signal component. Step 1110 determines the fundamental frequency of the tonal signal. The fundamental tone, often referred to as the fundamental and abbreviate f_0 is the lowest frequency in a harmonic series. The fundamental frequency (also called a natural frequency) of a periodic signal is the inverse of the pitch period length. The pitch period is the smallest repeating unit of a signal. The fundamental frequency of the tonal signal can be calculated using an autocorrelation analysis.

[0097] In one exemplary embodiment, a mathematical operation 1114 is performed where the frequency component of the test signal is limited to a frequency range below a lower minimum and upper maximum frequency range. Fund_ratio is calculated, which is defined as a ratio of the determined fundamental frequency (F_fund) of the tonal signal from step 1110 to an upper threshold value F_fund_threshold 1112, which in one exemplary embodiment, is a fixed constant equal to approximately 100Hz. In general,

F_fund_threshold 1112 is chosen to be a low frequency value which is above the lowest (or -3 dB) frequency that a transducer can reproduce, but below a predetermined frequency. In a comparison step 1116,

if the estimated F-fund is higher than the F_fund_threshold 1112 (ratio > 1), then F_fund is reduced by an integer multiple to be below F_fund_threshold 1112 corresponding to the mathematical operation of step 1118. Otherwise, the test signal fundamental is equal to F_fund as shown in step 1120. Although not shown, the calculated test signal fundamental is compared and determined to be greater than a predetermined threshold.

[0098] Referring to FIG. 1, in at least one exemplary embodiment, processor 116 is configured to receive or generate audio content. As mentioned previously, the audio content can from external devices such as a portable phone or a media player. Memory 104 can be used as a buffer for the audio content. Processor 116 is configured to receive the buffer of audio content from memory 104. The steps and calculations of the block diagram of FIG. 4 are then performed by processor 116. The result being one of the identification of a strong tonal signal component in the buffer of audio content and the test signal fundamental or loading the buffer with new audio content and starting the process again.

[0099] Referring to FIG. 12, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step 1106 of FIG. 11 that analyzes audio content stored in a buffer. The method begins at step 1202. A step 1204 gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step 1206 filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20Hz to 500Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step 1208 analyzes the audio signal where a strong tonal signal component is represented by peaks in the analysis results. A step 1210 generates Absolute(Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute(Acorr) can be the square of the results from the auto-correlation.

[0100] A crest_factor_Acorr 1218 is generated from the results by calculating an RMS value 1214 (or time-averaged peak value) and peak value 1216 (or time averaged peak value). In at least one exemplary embodiment, the crest_factor_Acorr 1218 is the ratio of the peak value to the RMS value of an absolute auto-correlation sequence of the audio signal.

[0101] A comparison step 1222 is then performed. A strong tonal presence is identified when crest_factor_Acorr 1218 is greater than a threshold Crest_factor_Acorr_threshold 1220. Identification of the strong tonal presence indicates the audio signal would facilitate masking of the test signal to determine sealing of the device (step 1226). The audio signal is not used in conjunction with the test signal if

crest_factor_Acorr 1218 is less than Crest_factor_Acorr_threshold 1220 (step 1224). The process would begin again loading a next sequence of the audio signal into the buffer for review.

[0102] Referring to FIG. 1, as mentioned previously, audio content is stored in a buffer, for example memory 104. The audio content in the buffer is provided to processor 116. In at least one exemplary embodiment, processor 116, runs the analysis as described in the block diagram of FIG. 12 thereby determining if a strong tonal presence is found in the audio content in the buffer. New audio content is loaded into the buffer (memory 104) if a strong tonal presence is not found beginning the procedure again.

[0103] Referring to FIG. 13, a flowchart of a method to determine when to emit the test signal is shown. The method begins at step 1302. The test signal is emitted when the test can be performed unobtrusively to the user and also provide an accurate test. In a step 1304, an audio signal is retrieved from a buffer. In at least one exemplary embodiment, the audio signal is received from an ECM or an ASM. The audio signal is measured to determine when the sound level is low in the ear canal, the ambient environment, or both. In general, the test signal is emitted when the sound level is low.

[0104] A filter step 1306 band pass filters the audio signal. In one exemplary embodiment, filter step 1306 filters the audio signal from 50 Hz to 150Hz which corresponds to a frequency range of the test signal. In a step 1308, the RMS of the audio signal is calculated. The audio signal is analyzed to detect when the energy within an audio frequency range is below a threshold RMS_threshold 1310. The RMS of the audio signal is the signal level in the volume being measured. A comparison step 1312 compares the measured RMS level of the filtered audio signal against RMS_threshold 1310. In a step 1314, a test signal is emitted when the measured RMS value is less than RMS_threshold 1310. No test signal is emitted when the RMS of the audio signal is greater than RMS_threshold 1310. The method ends at step 1316.

[0105] Referring to FIG. 14, a flowchart of an exemplary method to determine acoustic seal integrity is illustrated. For example, an earpiece seal integrity corresponds to a full or partial acoustic barrier between a first volume (ear canal) and a second volume (ambient environment). In one exemplary embodiment, the degree of acoustic seal integrity is expressed as either a PASS or FAIL status, where FAIL indicates that the acoustic seal is compromised relative to a normal operating acoustic seal. For example, an earpiece that has performed the seal test and determined that the sealing unit is not sealed correctly in the ear canal of the user can provide a signal or message indicating the problem. The user can then remove, reinsert, and retest the

earpiece to ensure that the seal is within normal operating specifications.

[0106] The method begins at step 1402. An acoustic test signal is provided in a first volume. In a step 1404 a transducer measures the acoustic test signal and stores it in a signal buffer. In a step 1406, a second transducer in a second volume isolated from the first volume by an acoustic barrier measures a second acoustic signal in the second volume. A portion of the acoustic test signal passes the acoustic barrier into the second volume. The amount of the acoustic test signal passing the acoustic barrier is a measure of the seal provided by the acoustic barrier.

[0107] In a filter step 1408, the measured acoustic test signal in the first volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals that are not part of the test. The measured signal from the first volume is heretofore called the first volume signal. Similarly, in a step 1410, the measured signal in the second volume is filtered in a frequency range corresponding to the acoustic test signal to remove signals not related to the test (outside the frequency range) in the second volume. The measured signal from the second volume is heretofore called the second volume signal.

[0108] A correlation, cross-correlation, or coherence analysis is performed on the first volume signal and the second volume signal. The correlation, cross-correlation, or coherence analysis is a measure of the similarity of the signals in the first and second volumes. In particular, the non- difference analysis measures the acoustic test signal leaking past the acoustic barrier by identifying the portion of the second volume signal that is similar to the acoustic test signal in the first volume.

[0109] In at least one exemplary embodiment, a correlation step 1412 is performed comprising a cross-correlation of the first volume signal and the second volume signal. In a step 1414, the peak of the cross-correlation is identified. The peak of the cross-correlation is Absolute(XCorr). In a mathematical step 1418, the Lag-time of Peak 1420 and the Magnitude of Peak 1422 is calculated. The Lag-time of Peak 1420 is a measure of the time delay between receiving the signals in the first and second volumes. In particular, the first volume signal should be received before the second volume signal. The Magnitude of Peak 1422 corresponds to the similarity between the signals in the first and second volumes. Thus, a larger number for Magnitude of Peak relates to more leakage of the acoustic test signal getting past the acoustic barrier.

[0110] Two comparisons are performed that determine if the acoustic barrier is sealed correctly based on the measured and calculated data from the first and second volumes. In a comparison step 1426, the measured Lag-time of Peak is compared against Target Lag Limits 1424. The measured lag-time should fall within the

predetermined range (Target Lag Limits 1424) for the seal test to be valid. If the Lag-time of Peak is within the appropriate range then a logic 1 is provided to AND function 1432, otherwise a logic 0 is provided. In a

second comparison step 1428, the Magnitude of Peak is compared against a Peak_threshold 1430. If the Magnitude of Peak is greater than the Peak_threshold 1430 a logic 1 is provided to AND function 1432. This indicates that a significant portion of the acoustic test signal is present in the second volume measurement, otherwise a logic 0 is provided. A FAIL output 1434 corresponds to a logic 1 at the output of AND function 1432. The FAIL occurs when the Lag-time of Peak is within the predetermined range and the Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output 1434 and the acoustic barrier is sealed correctly.

[0111] In at least one exemplary embodiment and referring briefly to FIG. 1, an earpiece is tested to determine if sealing unit 108 is sealed correctly to the ear canal of the user. Sealing unit 108 creates a first volume in ear canal 124 and a second volume outside the ear canal 124 in the ambient environment. A masking approach is used to perform seal testing unobtrusively to the user. The user is listening to music or speech (audio content) provided to ear canal124 from ECR 114.

[0112] The music or speech is buffered in memory 104 or memory in processor 116. Processor 116 analyzes the audio content in the buffer to identify a strong tonal content. A test signal can be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor 116 also analyzes the measured signals from ECM 106 and ASM 120 to determine when to emit the test signal. Processor 116 monitors and compares the sound level in the ambient environment and ear canal124. Processor 116 will provide the generated test signal to ECR 114 during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor 116 will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

[0113] Processor 116 monitors the test conditions and then provides the test signal to ECR 114 when an accurate sealing test can be performed. ECR 114 outputs an acoustic test signal which may or may not have other audio content. ECM 106 and ASM 120 respectively measure acoustic signals in ear canal 124 and the ambient environment. Processor 116 is operatively coupled to ECM 106 and ASM 120. The measured signals are buffered in memory 104.

[0114] In an exemplary embodiment, a cross-correlation is used to measure the similarity between the

signals in ear canal124 and the ambient environment. Processor 116 performs the cross-correlation calculations using the measured acoustic signals from ECM 106 and ASM 120. In particular, the cross-

correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$XCorr(n,l) = \sum_{n=0}^{N} ASM(n)ECM(n-l), \quad (5)$$

Where: l=0, 1, 2, … N

[0115] Where ASM(n) is the nth sample of the ASM signal, and ECM(n- l) is the (n- l)th sample of the ECM signal. A peak of the absolute cross-correlation is estimated using a peak- picking function and also the lag time at which this peak occurs (i.e. the index I at which this occurs). Thus, the Lag-time of Peak and the Magnitude of Peak are known and respectively compared against a Target Lag Limit range and a Peak Threshold. The user of the earpiece is notified or warned that sealing unit 108 is improperly sealed by processor 116 if the measured Lag-time of Peak is within the Target Lag Limit range and the Magnitude of Peak is greater than the Peak Threshold.

[0116] Like Correlation and Cross-Correlation, a coherence function is also a measure of similarity between two signals. Coherence is another non-difference comparison approach that can be used for detecting acoustic seal integrity. Coherence is defined as:

$$\gamma^2{}_{xy} = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (6)$$

[0117] Where Gxy is the cross-spectrum of two signals (e.g. the ASM and ECM signals), and can be calculated by first computing the cross-correlation in equation (5), applying a window function, for example a Hanning window, and transforming to the frequency domain, for example via an FFT. Gxx or Gyy is the auto-power spectrum of either the ASM or ECM signals, and can be calculated by first computing the auto-correlation (using equation 5, but where the two input signals are both from either the ASM or ECM and transforming to the frequency domain. The coherence function gives a frequency-dependent vector between 0 and 1, where a high coherence at a particular frequency indicates a high degree of coherence at this frequency, and can therefore be used to analyze test signal frequencies in the ASM and ECM signals whereby a high coherence indicates the presence of the test signal in the ambient environment (indicating

leakage past the acoustic barrier).

[0118] Other approaches such as frequency spectrum analysis and RMS levels can also be used to determine if the earpiece is sealed correctly. Using a non-difference comparison approach such as coherence or cross-correlation between the ASM and ECM signals to determine sealing is more reliable than taking the level difference of the ASM and ECM signals. Using the cross correlation rather than a level differencing approach improves the accuracy and minimizes errors which may occur due to user non-speech body noise, such as teeth chatter, sneezes, coughs, etcetera. Furthermore, such non-speech user generated noise would generate a larger sound level in the ear canal than on the outside of the same ear canal producing inaccurate results.

[0119] FIGS. 15A and 15B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., 1500) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. 15A and 15B shows a slider actuator 1500, with a moving slide 1510, having attached a pumping arm 1560, with a hole 1550 covering bump 1520. Upon receipt of a signal1505 the actuator can move 1570 such that the bump 1520 covers the hole 1550 in a bellows 1530 pump. The actuation compresses the miniature bellows 1530 pushing 1590 gas through the one-way valve 1540, upon the back stroke the bump 1520 uncovers the hole 1550 and air rushes back into the bellows 1530 for the next pump. For example if the stroke length is 2 mm and the pump arm 1560 contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system 1570 needs to be inflated more (e.g., more gas to increase sealing) then each stroke can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system 1580. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal 1505 sent, which can be specifically tailored depending upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation systems can be used and controlled by signal 1505.

[0120] FIG. 16 illustrates a block diagram of controlled sealing in response to a seal fail signal and/or a request for increased seal attenuation. For example, a command signal 1600 is received (e.g., a seal fail signal in which a default attenuation value is attached for example 15 dB, or a signal requesting an additional amount of attenuation) by a processor. The command signal specifies an additional amount of attenuation or that the seal has failed. An attenuation needed N is identified 1610. For example if the current attenuation is 5 dB loss at f=500 Hz, at a pressure of 0.1 bar, and a command signal is received requesting a 10 dB loss at 500 Hz, then an experimental table is queried to find the pressure needed P which is then subtracted from the current pressure to obtain an increase in pressure DP needed. The increase in pressure is converted into a volume of gas increase needed (e.g., again referring to experimental tables based upon the inflation system volume). The volume of gas increase needed can then be directly linked with the number of cycles of an actuator pump M, 1620. The number of requested cycles **M** can then be sent 1630 to the actuator control circuit to pump the designated number of cycles. The system can then retest the attenuation 1640 and if refinements are needed the process can start again.

[0121] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

[0122] Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

Electronic Acknowledgement Receipt

EFS ID:	47304647
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	11472-36-3_CONT-16838277
Receipt Date:	03-JAN-2023
Filing Date:	23-FEB-2021
Time Stamp:	09:32:11
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Amendment after Notice of Allowance (Rule 312)	17182570_312-amndt-2nd.pdf	115762 abe1f862d42246edcea9de80537858da276bdcff	no	1

Warnings:

Information:					
2	Applicant Arguments/Remarks Made in an Amendment	17182570-SPEC-Marked-2nd312.pdf	395823 af7a110c27c61b612462d4549cf629342f2e07b1	no	25
Warnings:					
Information:					
3	Specification	17182570-SPEC-Clean-2nd312.pdf	381315 231a7036406b7c3f90d4f1f9c77b0050905d11c7	no	25
Warnings:					
Information:					
			Total Files Size (in bytes):	892900	

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.

<u>New Applications Under 35 U.S.C. 111</u>
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.
<u>National Stage of an International Application under 35 U.S.C. 371</u>
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.
<u>New International Application Filed with the USPTO as a Receiving Office</u>
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: MFEE.C.AD PTO/SB/47-EFS

Document Description: Maintenance Fee Address Change

FEE ADDRESS INDICATION FORM			
Application Number	17182570	Art Unit	2654
Filing Date	23-Feb-2021	Examiner Name	PAUL KIM
First Name Inventor	John Usher	Attorney Doc. Number	11472-36-3_CONT-16838277

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**	01-05-2023
Name	Peter Allen Chiabotti	**Registration Number**	54603

Electronic Patent Application Fee Transmittal

Application Number:	17182570
Filing Date:	23-Feb-2021
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Filer:	Peter Allen Chiabotti/Yoshi Hassan
Attorney Docket Number:	11472-36-3_CONT-16838277

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	600	600
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($)
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**600**

Electronic Acknowledgement Receipt	
EFS ID:	47320887
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	Peter Allen Chiabotti/Yoshi Hassan
Filer Authorized By:	Peter Allen Chiabotti
Attorney Docket Number:	11472-36-3_CONT-16838277
Receipt Date:	05-JAN-2023
Filing Date:	23-FEB-2021
Time Stamp:	13:23:36
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	DA
Payment was successfully received in RAM	$600
RAM confirmation Number	E202315D23354901
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	62593 b1461ee9dbecf0cf5883669f3afa9000a7d9c859	no	2
Warnings:					
Information:					
2	Maintenance Fee Address Change	web85feeaddress.pdf	42303 2fbe7932683c7fb3c237322dfd7fb37ce5b4d5b0	no	1
Warnings:					
Information:					
3	Fee Worksheet (SB06)	fee-info.pdf	40003 3a9154ed826a6156d63c9b5db5fa80e8969fec51	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	144899		

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.

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

Issue Fee Transmittal Form

Application Number	Filing Date	First Named Inventor	Atty. Docket No.	Confirmation No.
17182570	23-Feb-2021	John Usher	11472-36-3_CONT-1683827	5671

TITLE OF INVENTION :

ACOUSTIC SEALING ANALYSIS SYSTEM

Entity Status	Application Type	Art Unit	Class - Subclass	EXAMINER
Small	Utility under 35 USC 111(a)	2654	059000	PAUL KIM

Issue Fee Due	Publication Due	Total Fee(s) Due	Date Due	Prev. Paid Fee
$600	$0	$600	23-Mar-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 _____
☐ Advance Order - # of copies _____	☐ 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 _____ . 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**	01-05-2023
Name	Peter Allen Chiabotti	**Registration Number**	54603

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
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Alexandria, Virginia 22313-1450
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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277	5671

150139 7590 01/09/2023

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

EXAMINER	
KIM, PAUL	

ART UNIT	PAPER NUMBER
2654	

NOTIFICATION DATE	DELIVERY MODE
01/09/2023	ELECTRONIC

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

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

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

ip@akerman.com
jp.keady@techiya.co
personics_docketing@cardinal-ip.com

Response to Rule 312 Communication	Application No.	Applicant(s)	
	17/182,570	Usher et al.	
	Examiner	Art Unit	AIA (FITF) Status
	PAUL KIM	2654	Yes

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

1. ☑ The amendment filed on 03 January 2023 under 37 CFR 1.312 has been considered, and has been:
   a) ☑ entered.
   b) ☐ entered as directed to matters of form not affecting the scope of the invention.
   c) ☐ disapproved because the amendment was filed after the payment of the issue fee.
      Any amendment filed after the date the issue fee is paid must be accompanied by a petition under 37 CFR 1.313(c)(1) and the required fee to withdraw the application from issue.
   d) ☐ disapproved. See explanation below.
   e) ☐ entered in part. See explanation below.
   f) ☐ not entered because the supplemental or corrected Application Data sheet (ADS)
      ☐ was not accompanied by a petition to accept an unintentionally delayed claim under 37 CFR 1.55 or 27 CFR 1.78;
      ☐ did not identify the information being changed in accordance with 37 CFR 1.76(c)(2);
      ☐ was not properly signed in accordance with 37 CFR 1.76(e) (or 37 CFR 1.33(b) for applications filed prior to September 16, 2012).

/PAUL KIM/
Primary Examiner, Art Unit 2654

O.K. TO ENTER – 312
/P.K./ AU 2654
January 3, 2023

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

2nd Amendment After Allowance Under 37 CFR 1.312

Commissioner of Patents
Alexandria, VA 22313-1450

DATE: 3 Jan. 2023

The Office issued a notice of allowance 23 Dec. 2022. Prior to final payment of the issue fee Applicant has noticed two additional grammatical issues, corrected herein. Thus, Applicants submit the second attached marked-up and clean version of the specification to address typographical errors, in particular in paragraphs [0001] and formula (2) [0071] where Sxx should be Sxy.

The minor 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. However, 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.

Respectfully submitted,
TECHIYA LLC

Date: January 3, 2023

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci), J.D.
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

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.
17/182,570	02/23/2021	John Usher	11472-36-3_CONT-16838277	5671

150139 7590 02/10/2023

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

EXAMINER	
KIM, PAUL	

ART UNIT	PAPER NUMBER
2654	

NOTIFICATION DATE	DELIVERY MODE
02/10/2023	ELECTRONIC

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

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

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

ip@akerman.com
jp.keady@techiya.co
personics_docketing@cardinal-ip.com

Response to Rule 312 Communication	Application No.	Applicant(s)	
	17/182,570	Usher et al.	
	Examiner	Art Unit	AIA (FITF) Status
	PAUL KIM	2654	Yes

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

1. ☑ The amendment filed on 30 December 2022 under 37 CFR 1.312 has been considered, and has been:
   a) ☑ entered.
   b) ☐ entered as directed to matters of form not affecting the scope of the invention.
   c) ☐ disapproved because the amendment was filed after the payment of the issue fee.
      Any amendment filed after the date the issue fee is paid must be accompanied by a petition under 37 CFR 1.313(c)(1) and the required fee to withdraw the application from issue.
   d) ☐ disapproved. See explanation below.
   e) ☐ entered in part. See explanation below.
   f) ☐ not entered because the supplemental or corrected Application Data sheet (ADS)
      ☐ was not accompanied by a petition to accept an unintentionally delayed claim under 37 CFR 1.55 or 27 CFR 1.78;
      ☐ did not identify the information being changed in accordance with 37 CFR 1.76(c)(2);
      ☐ was not properly signed in accordance with 37 CFR 1.76(e) (or 37 CFR 1.33(b) for applications filed prior to September 16, 2012).

/PAUL KIM/
Primary Examiner, Art Unit 2654

O.K. TO ENTER - 312
/P.K./ AU 2654
December 30, 2022

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

Amendment After Allowance Under 37 CFR 1.312

Commissioner of Patents
Alexandria, VA 22313-1450

DATE: 30 Dec. 2022

The Office issued a notice of allowance 23 Dec. 2022. Applicants submit the attached marked-up and clean version of the specification to address typographical errors, in particular in paragraphs [0079] and [0115].

The minor 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. However, 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.

Respectfully submitted,
TECHIYA LLC

Date: December 30, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci), J.D.
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17182570
	Filing Date	2021-02-23
	First Named Inventor	John Usher
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-16838277

	24	20050123146		2005-06-09	Voix et al.	
	25	20040125965		2004-07-01	Alberth Jr., et al.	
	26	20060083395		2006-04-20	Allen et al.	
	27	20110264447		2011-10-27	Visser et al.	
	28	20100296668		2010-11-25	Lee et al.	
	29	20110293103		2011-12-01	Park et al.	
	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]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]
Change(s) applied to document, /Q.N./ 12/29/2022	1	EP1519625	EP	A2	03/2005	Victorian		
	2	2006037156	WO	A1	04/2006	Mejia et al.		

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ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

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

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	
Filing Date	
First Named Inventor	John Usher
Art Unit	
Examiner Name	Unknown
Attorney Docket Number	CONT-16838277

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	7801318		2010-09-21	Barthel	
	2	8774433		2014-07-08	Goldstein	
	3	7801318		2010-09-21	Barthel, Roland	
Change(s) applied to document, /C.S./ 12/29/2022	4	8401200		03/2013 ~~2009-11-19~~	Tiscareno	
	5	5138722		08/1992 ~~2010-09-21~~	Urella	
	6	5764781		1998-06-09	Ding	
	7	6321070		2001-11-20	Clark	
	8	9293128		2016-03-22	Goldstein	

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /P.K/

Electronic Acknowledgement Receipt

EFS ID:	47961646
Application Number:	17182570
International Application Number:	
Confirmation Number:	5671
Title of Invention:	ACOUSTIC SEALING ANALYSIS SYSTEM
First Named Inventor/Applicant Name:	John Usher
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	11472-36-3_CONT-16838277
Receipt Date:	06-MAY-2023
Filing Date:	23-FEB-2021
Time Stamp:	21:42:10
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Amendment after Notice of Allowance (Rule 312)	17182570_312.pdf	166885 e8dd6cc39ee2601564768c928e988d0dfc183a66	no	5

Warnings:

Information:	
Total Files Size (in bytes):	166885

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

New Applications Under 35 U.S.C. 111

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

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

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

New International Application Filed with the USPTO as a Receiving Office

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

IN THE U.S. PATENT AND TRADEMARK OFFICE

First Named Inventor:	John Usher	Conf.: 5671
Appl. No.:	17/182,570	Art Unit: 2654
Filed:	02/23/2021	Examiner: Kim, Paul
For:	ACOUSTIC SEALING ANALYSIS SYSTEM	

Amendment After Allowance Under 37 CFR 1.312

Commissioner of Patents
Alexandria, VA 22313-1450

DATE: 30 Dec. 2022

Sir:

Applicant submits herein a clarifying amendment to claim 1, where Applicant clarifies that the processor executes the instructions rather than "is configured to" execute the instructions. No new matter has been added. Applicant respectfully requests that the clarifying amendment be accepted.

Amendment(s) to the Claims begin on page 2 of this paper.

If the Examiner has any questions concerning this application, the Examiner is requested to contact the undersigned at 571-606-1291 in the Washington D.C. Metro area.

Respectfully submitted,
TECHIYA LLC

Date: May 6, 2023

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci), J.D.
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389

Amendment(s) to the Claim(s):

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

Listing of Claims:

1.(currently amended) An audio device configured to test a seal of an earphone comprising:

a memory that stores instructions; and

a processor, wherein the processor ~~is configured to~~ executes the instructions to perform operations, the operations comprising:

generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one component lies between a range of 30Hz and 200Hz;

sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user;

receiving a first microphone signal from the first microphone;

calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination;

comparing the value to a threshold to determine a seal quality between the first side and the second side;

generating a message to convey the seal quality; and

sending the message to the user.

2.(previously presented) The audio device according to claim 1, wherein the message is an acoustic message.

3(previously presented). The audio device according to claim 1, wherein the message is a visual message sent to a display of the audio device.

4.(previously presented) The audio device according to claim 1, wherein the speaker plays the test signal from the first side.

5.(currently amended) The audio device according to claim 4, wherein the first microphone measures sound from the first side. ~~signal.~~

6.(previously presented) The audio device according to claim 4, wherein the first microphone measures sound from the second side.

7.(previously presented) The audio device according to claim 1, wherein the first signal is the test signal and the second signal is the first microphone signal.

8.(previously presented) The audio device of claim 1, wherein the first signal is the test signal and the second signal is a modified first microphone signal.

9.(previously presented) The audio device of claim 1, wherein the first signal is a modified test signal and the second signal is the first microphone signal.

10.(previously presented) The audio device according to claim 1, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.

11.(previously presented) The audio device according to claim 8, wherein the modified first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

12.(previously presented) The audio device according to claim 10, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200Hz.

13.(previously presented) The audio device according to claim 1, wherein the earphone includes a second microphone.

14.(previously presented) The audio device according to claim 13, wherein the operations further comprise:

receiving a second microphone signal from the second microphone.

15.(currently amended) The audio device according to claim 14, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.

16.(previously presented) The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.

17.(previously presented) The audio device of claim 14, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal.

18.(previously presented) The audio device of claim 1, wherein the seal quality is good if the value is at or above the threshold value.

19.(previously presented) The audio device of claim 1, wherein the sealing section is composed of at least one of rubber, silicone, a flexible polymer, or a combination thereof.

20.(previously presented) The audio device of claim 1, wherein the sealing section is an eartip.

UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/182,570	05/30/2023	11665493	11472-36-3_CONT-16838277	5671

150139 7590 05/10/2023

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 223 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):

John Usher, Devon, UNITED KINGDOM;
John P. Keady, Fairfax Station, VA;

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

Staton Techiya LLC, Lighthouse Point, FL

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UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NO.	ISSUE DATE	PATENT NO.
17/182,570	30-MAY-23	11665493

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

Doc code: RCEX
Doc description: Request for Continued Examination (RCE)

PTO/SB/30EFS (07-09)
Approved for use through 07/31/2012. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

REQUEST FOR CONTINUED EXAMINATION(RCE)TRANSMITTAL
(Submitted Only via EFS-Web)

Application Number	18092645	Filing Date	2023-01-03	Docket Number (if applicable)	11472-36-4 CONT-17182570	Art Unit	2654
First Named Inventor	John Usher, et. al.			Examiner Name	Paul Kim		

This is a Request for Continued Examination (RCE) under 37 CFR 1.114 of the above-identified application.
Request for Continued Examination (RCE) practice under 37 CFR 1.114 does not apply to any utility or plant application filed prior to June 8, 1995, or to any design application. The Instruction Sheet for this form is located at WWW.USPTO.GOV

SUBMISSION REQUIRED UNDER 37 CFR 1.114

Note: If the RCE is proper, any previously filed unentered amendments and amendments enclosed with the RCE will be entered in the order in which they were filed unless applicant instructs otherwise. If applicant does not wish to have any previously filed unentered amendment(s) entered, applicant must request non-entry of such amendment(s).

☐ Previously submitted. If a final Office action is outstanding, any amendments filed after the final Office action may be considered as a submission even if this box is not checked.

- ☐ Consider the arguments in the Appeal Brief or Reply Brief previously filed on _____
- ☐ Other _____

☒ Enclosed

- ☐ Amendment/Reply
- ☒ Information Disclosure Statement (IDS)
- ☐ Affidavit(s)/ Declaration(s)
- ☐ Other _____

MISCELLANEOUS

☐ Suspension of action on the above-identified application is requested under 37 CFR 1.103(c) for a period of months _____
(Period of suspension shall not exceed 3 months; Fee under 37 CFR 1.17(i) required)

☐ Other _____

FEES

☒ **The RCE fee under 37 CFR 1.17(e) is required by 37 CFR 1.114 when the RCE is filed.**
The Director is hereby authorized to charge any underpayment of fees, or credit any overpayments, to Deposit Account No 500951

SIGNATURE OF APPLICANT, ATTORNEY, OR AGENT REQUIRED

☒ Patent Practitioner Signature

☐ Applicant Signature

PTO/SB/30EFS (07-09)
Approved for use through 07/31/2012. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

Signature of Registered U.S. Patent Practitioner			
Signature	/Peter A. Chiabotti/	Date (YYYY-MM-DD)	2023-08-24
Name	Peter A. Chiabotti	Registration Number	54603

This collection of information is required by 37 CFR 1.114. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450.

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

Privacy Act Statement

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

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

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these records.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent 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 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.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Application No.	:	18092645	Confirmation No. 1770
Application of	:	John Usher, et. al.	
Date Filed	:	January 3, 2023	
Examiner	:	Paul Kim	
TC/A.U.	:	2654	
Docket	:	11472-36-4 (CONT-17182570)	
Customer No.	:	30448	
Title	:	Acoustic Sealing Analysis System	

Information Disclosure Statement

Pursuant to the Duty to Disclose under 37 C.F.R. §1.56, the references cited on the accompanying form PTO/SB/08a are hereby brought to the attention of the Examiner. Because the form PTO/SB/08a does not include a certification statement, the following is provided.

This Information Disclosure Statement is being filed before the mailing date of a first Office Action on the merits, or before the mailing of a first Office Action after the filing of a Request for Continued Examination.

Although no fees are believed to be due, authorization is hereby given to charge any deficiencies and credit any overpayment to Deposit Account No. 50-0951.

Applicant respectfully requests the Examiner's consideration of the references cited therein and entry thereof into the record of this application.

Respectfully submitted,

AKERMAN LLP

Date: August 24, 2023

/Peter A. Chiabotti/

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Information Disclosure Statement List		
	Serial No.	18092645
	Filing Date	January 3, 2023
	Inventor	John Usher, et. al.
	Art Unit	2654
	Examiner	Paul Kim
	Docket No.	11472-36-4 (CONT-17182570)

	PATENT AND PATENT PUBLICATION DOCUMENTS					
Examiner's Initials	**Cite No.**	**Country Code**	**Doc. No.**	**Date**	**Inventor(s)/Applicant(s)**	**Translation?**
	1.	WO	WO2006037156	2006-04-13	Hear Works Pty Ltd	No
	2.	WO	WO2008050583	2008-05-02	Matsushita Electric Works Ltd	No
	3.	WO	WO2007092660	2007-08-16	Koninkl Philips Electronics Nv	No
	4.	WO	WO2004114722	2004-12-29	Gn Resound As	No
	5.	WO	WO9326085	1993-12-23	Noise Cancellation Tech	No
	6.	US	US8045840	2011-10-25	Murata	No
	7.	US	US7929713	2011-04-19	Victorian Thomas A	No
	8.	US	US8116489	2012-02-14	Mejia Jorge Patricio	No
	9.	US	US9497423	2016-11-15	Moberly David S	No
	10.	US	US7773763	2010-08-10	Pedersen Brian Dam	No
	11.	JP	JP3353701	2002-12-03	Yamaha Cor	Yes
	12.	EP	EP1519625	2005-03-30	Starkey Lab Inc	No
	13.	EP	EP1401240	2004-03-24	Siemens AG	No
	14.	EP	EP1385324	2004-01-28	Siemens AG	No
	15.	EP	EP1640972	2006-03-29	Phonak AG	No

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 401 240 A1

(12) **EUROPEAN PATENT APPLICATION**

(43) Date of publication:
24.03.2004 Bulletin 2004/13

(51) Int Cl.[7]: **H04R 25/00**

(21) Application number: **02020319.6**

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(84) Designated Contracting States:
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Designated Extension States:
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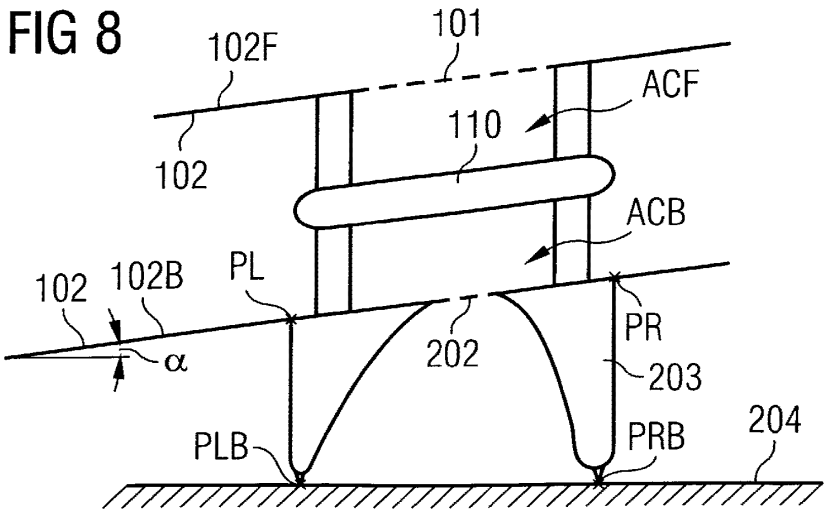
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(72) Inventor: **Jensen, Rene Eske**
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(54) **A dual directional mode mobile terminal and a method for manufacturing of the same**

(57) A mobile terminal is equipped with an acousto-electrical transducer arranged in a housing of the mobile terminal, the housing comprising i) a user sided first inlet aperture in communication with the acoustoelectrical transducer, ii) a second inlet aperture also in communication with the acoustoelectrical transducer, and iii) sealing means for sealing the second inlet aperture when the mobile terminal is placed on an external surface or object. As a result, the mobile terminal has selectable directional characteristics.

FIG 8



EP 1 401 240 A1

Printed by Jouve, 75001 PARIS (FR)

Description

Field of the invention

[0001] The invention relates generally to enhancing speech quality in a telecommunications system, and, more specifically, to enhancing directional properties of a microphone in a mobile terminal.

Background of the invention

[0002] The mobile device boom of the recent years has resulted in different new uses for the mobile terminals widely used in telecommunications. The users expect their new devices to be usable in many different situations. The expectations regarding to the acoustical quality of the device are high, and because the device may be used for various purposes, this sets very high requirements for the design of the terminals.

[0003] As an example of the dual use of the mobile handset, it is very common that the terminal, in addition to its normal use as a handset, is used as a hands-free device. This may be the case when the user is willing to use the mobile terminal in a vehicle, for example, or use it for conference calling. Typically, in the latter case, the mobile terminal is set on a table, the persons willing to speak then usually grouping around the terminal.

[0004] From the acoustical design point of view, the latter kind of use requires quite different acoustical and directional properties of the microphone than the first use mentioned. In conference calling, it is very critical that the microphone shows omni-directional characteristics so that the position of the person speaking does not influence too radically on the sound quality, and when one person is speaking, also comments from the background should be carefully detected.

[0005] On the other hand, when a mobile terminal is used as a handset, it is preferable that the background noise is not detected by the microphone. Normally, the mouth and other sound producing organs of the mobile user are within a predetermined distance from the microphone when he or she is speaking to his or her mobile handset. If human anatomy factors are taken into account, it is easier to design a phone in such a way that the mouth of an average male/female user is quite close to the centre of inlet of the microphone. Thus the microphone should preferably show unidirectional characteristics in order to receive the speech of the user but not noise from his or her background.

[0006] Figure 1 illustrates a prior art mobile terminal 100. The terminal 100 includes housing 102, wherein a microphone 110 is assembled. For the microphone 110 an aperture 101 is important so that the energy of acoustic signals is not attenuated too much which would be the case if the acoustic signals had to traverse through the housing 102.

[0007] Having both omni- and unidirectional characteristics in the acoustical parts of one mobile terminal is not a straightforward task. Some prior art solutions that might be applicable to at least some extent can be found, but they present drawbacks which both an ordinary user and an acoustics designer would find annoying.

[0008] International patent publication WO 01/69967 A2 shows one example of implementing a selectivity switch between a directional mode and an omni-directional mode in a hearing aid device. The switch includes a switch body having a port aperture therein in communication with one of the ports of the microphone, and a switch actuator moveably connected to the switch body and moveable to a first position wherein the port aperture is generally unobstructed by the actuator and a second position wherein the actuator covers the port aperture.

[0009] In the modern mobile lifestyle, a mobile terminal sometimes presents some values of the user of the terminal. Thus, the aesthetic design is very important, and the switch does not support the purpose of creating an aesthetically enjoyable device. Secondly, the shape and form of a hearing aid device is quite different to a mobile phone, basically because the hearing aid device is located in the ear of the user, so the mechanics of the device can be designed in a completely different manner. Therefore the '967 solution cannot be without problems used for ordinary mobile terminals.

[0010] Another solution from U.S. patent 5,933,510, despite of its many advantages, fits only poorly for the acoustical design of a mobile terminal. In '510, a microphone housing includes apertures and a switch; the user may select with the switch, whether or not he or she is willing to use the microphone in omni- or unidirectional mode. Firstly, it is difficult to make the switch and sealing elements to comply with the decorative design desired for the terminal. Secondly, the design of a rotating sleeve for the microphone housing in the manner described in '510 is also extremely challenging.

[0011] It is an object of the invention to bring about a mobile terminal equipped with a sound receiving system having selectable directional characteristics.

Summary of the invention

[0012] This and other objectives of the invention are accomplished in accordance with the principles of the present invention by providing in accordance with any one of the independent patent claims.

[0013] A dual directional mode mobile terminal includes a microphone having been assembled in a housing. The housing includes a first and a second aperture, of which the first aperture is on the user side when the terminal is used as a handset, the first and second apertures forming acoustical conduits to the both sides of the microphone. The housing further includes sealing means for sealing the second aperture together with an external surface, when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a

IPR2024-01015 Page 00394

second acoustical directivity mode.

[0014] A method for manufacturing a dual directional mode mobile terminal, includes the steps of i) assembling a microphone into a housing, wherein the housing includes at least a first and a second apertures, of which the first aperture is designed to be on the user side when the terminal is used as a handset, the first and second apertures thus forming acoustical conduits to both sides of the microphone, and ii) assembling the housing with sealing means for sealing at least one of the apertures together with an external surface, when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a second acoustical directivity mode.

[0015] A mobile terminal which is equipped with an acoustoelectrical transducer arranged in a housing of the mobile terminal, wherein the housing comprises i) a user sided first aperture in communication with the acoustoelectrical transducer, ii) a second aperture also in communication with the acoustoelectrical transducer, and iii) sealing means for sealing the second aperture when the mobile terminal is placed on an external surface or object.

Brief Description of the Drawings

[0016] In the following, the invention and its preferred embodiments are described more closely referring to the examples shown in Figure 2-11 in the appended drawings, wherein:

Figure 1	shows a conventional microphone arrangement in a mobile terminal,
Figure 2	shows a dual-mode mobile terminal according to one aspect of the invention,
Figure 3	illustrates one embodiment of the present invention where acoustic sealing is enhanced by a second support member,
Figure 4	depicts the traverse design aspects in accordance with the invention,
Figure 5	further illustrates one embodiment of the invention,
Figure 6	illustrates a cut of a sealing member 203 from Figure 2,
Figure 7	illustrates the sealing member 203 seen from above,
Figure 8	shows the second acoustical mode closer,
Figure 9A and 9B	show how the directivity mode of the mobile terminal is changed according to one aspect of the present invention,
Figure 10	shows a first measurement result of the mobile terminal in the first directional mode, and
Figure 11	shows a second measurement result of the mobile terminal in the second directional mode.

[0017] Same reference numbers denote similar elements throughout the figures.

Detailed description of the invention

[0018] Figure 2 illustrates one embodiment of the present invention. The mobile terminal has a housing 200, and, as shown in Figure 1 described above, there is also an aperture 101 provided for the speech. The aperture 101 leads sound waves to a microphone 110 that is assembled inside the housing 200. In the housing 200 there is also a second aperture 202. The second aperture 202 is preferably located on the opposite side of the housing 200 so that the apertures 101 and 202 provide acoustic conduits leading to both sides of the microphone.

[0019] For sake of clarity, there are three points marked in the housing. PL and PR are the points where sealing means 203 are attached to the housing 200, and the point PA illustrates the point of the housing that supports the lower end of the housing against a flat surface 204 when the mobile terminal is tilted on said sealing means 203. The sealing means 203 are not drawn to scale; according to one aspect of the present invention their height is some millimetres. Depending on the aesthetic goals of the designer, sealing means 203 that are higher or lower can be used.

[0020] According to one embodiment of the present invention the mobile terminal can be placed on a table 205, for example, or other flat surface 204, so that the sealing means 203 together with the flat surface 204 seal the second aperture 202. The sealing of the second aperture 202 has the effect that the mobile terminal is in omni-directional mode where the microphone 110 receives voice signals mostly through the aperture 101 thus losing some of its directional sensitivity.

[0021] Normally, the microphone 110 is directionally sensitive when both apertures 101 and 202 are open. When both apertures 101, 102 are open, the sound waves coming from the background noise can be understood to cancel each other, i.e. the microphone shows unidirectional characteristics and is well suitable for to be used as a handset of a mobile terminal.

The uni- and omni-directional characteristics of the microphone 110 are herewith referred with the term acous-

IPR2024-01015 Page 00395

tical modes of the microphone.

[0022] Figure 3 shows another example of implementing the present invention. Instead of only having sealing means 303 such as a sealing member, the mobile terminal now also has a support member 304. In addition, the marked point PB illustrates the effective support point of the support member 304 that is comparable to point PA with reference to Figure 2.

[0023] One function of the support member 304 is to increase the pressure between the sealing means 303 and the flat surface 204. The sealing means 303 are located close to the second aperture 202 which is further located at the spot of the microphone. As the mobile terminal of Figure 2, the mobile terminal can as well be placed on a flat surface 204 thus giving omni-directional characteristics for the mobile terminal. The position PB which is also the position of the second support member 304 may be located based on the weight distribution of the mobile terminal. To the weight distribution influences the weight of the contents of the mobile terminal, i.e. the battery, different electrical parts, the display, and so forth. The PB can be positioned so that the torque caused by the weight of the mobile terminal causes a stronger or greater force on the sealing means 303 and so enhances the sealing made by the sealing means 303 and the flat surface 204. If the sealing means 303 are made of a soft material showing high elastic properties, a good sealing can be provided.

[0024] Figure 4 depicts a cross sectional cut of a mobile terminal. The terminal has a microphone 110 which is positioned inside the housing 200. In the housing 200 there are apertures 101 and 202. The apertures 101, 202 act as inlet holes which form acoustical conduits to both sides of the microphone 110. It is to be noted that the apertures 101, 202 need not to be on opposite sides of the mobile terminal; the only limiting condition is that the acoustical conduits lead to both sides of the microphone. Further, there are support and sealing means 203 at the bottom side of the housing. The design of the sealing means 203 may involve taking into account the curvature of the housing 200. In some mobile terminal models, the housing 200 is strongly curved and, therefore, it is impossible to provide a good enough sealing with sealing means 203 that do not take into account the shape of the bottom of the housing 200.

[0025] Further, Figure 5 illustrates the mobile terminal viewed from below. In the sealing means 203 that is upper part of the mobile terminal housing 200 the aperture 202 and the sealing means 203 are shown.

[0026] Figure 6 illustrates closer the sealing means 203. First, the sealing means 203 have a hole 600 for the acoustical inlet. The contact part 601 of the sealing means 203 has been only marked with a box on the right-hand side, but it preferably extends through the sealing means 203. For example, if the sealing means 203 have the form of a ring like in Figure 5, the part 601 extends around the bottom of the ring.

[0027] Further, aperture part 603 has similarly been marked with a box. The acoustical signals coming from outside the mobile phone in order to reach the inlet hole 600 and to further proceed into the microphone 110 traverse beyond this part. In its design one must take into account the acoustical considerations: Special curvature may be needed in order to provide a proper distortion free acoustical transmission characteristics.

[0028] The sealing means 203 are connected to the mobile terminal using the aperture part 605 of the sealing means 203. On the left-hand side the height of the sealing means 203 is denoted with h' and on the right-hand side with h. The difference (h'- h) between h' and h is denoted with Δ. If the microphone is located as usually at the bottom part of the mobile terminal and if the tilting of the housing because of the height difference of the sealing means and the bottom part of the housing is like in Figure 2, h is larger than h'. The tilting may therefore depend on the distance between PA and PL, and partially as well on the distance between PL and PR. In the case of a mobile terminal with a support element like in Figure 3 there is no need for this kind of asymmetry and h' can equal h. The size of Δ may be adapted to make the sealing proper taking into account the tilting angle of the between the bottom part of the housing and the surface where the mobile terminal is lying.

[0029] Another possibility to achieve an asymmetric elastic behaviour is to use elastic materials asymmetrically, i.e. a more elastic material on the h' side and a stiffer material on the opposite side. This can in principle be achieved with e.g. block copolymers where the ratios of different polymer materials exhibit small changes along the bottom part of the sealing means. In this way it is also possible to ensure a good enough sealing. Another option, for example, is to use some other material that has spatially non uniform elasticity constant.

[0030] Figure 7 shows the sealing means from bottom. 600 is the inlet and 601 is the contact part. The profile of the aperture part 603 with its curvature is not visible in the figure.

[0031] Figure 8 presents the mobile terminal when the second acoustical directivity mode is activated. The mobile terminal 102 is placed on a flat surface 204. In the figure only the portion of the mobile terminal which illustrates the housing means and their vicinity is shown. The sealing means 203 have a contact with the flat surface 204 at aperture points PLB and PRB. Preferably, the sealing means extend from the housing of the mobile phone in such a way that in the figure left-hand side contact point PL acts as the outermost aperture point for the sealing means at part PLB and, respectively, the point PR at as the contact point of the sealing means for the point PRB.

[0032] The housing 200 is usually constructed from two parts; 102F is the front part and 102B is the back part. Between the front and back part 102F, 102B of the housing 200 there is a microphone 110. In the front part 102F at the spot of the microphone 110 there is at least one opening for the aperture 101 and on the back side

IPR2024-01015 Page 00396

part of the housing there is correspondingly at least one aperture 202 leading to the bottom side of the microphone.

[0033] Between the housing and the microphone there is sealing on both sides of the microphone 110. Preferably, the sealing is obtained with the help of an elastic gasket, such as a sealing ring. The first purpose of the sealing is to enable the propagation of sound from the exterior of the housing to the microphone, disabling propagation of the sound inside the housing, whereas the second purpose is to reduce the distortions caused by the loudspeaker of the mobile terminal.

[0034] Below the microphone 110 there is a resonance cavity ACB which is consists of smaller cavities formed by i) space limited by the microphone 110, possible sealing between the microphone, and bottom part 102 of the housing 200, and ii) acoustically well conductive part of the housing 200 including at least one aperture 202, sealing means 203, and the flat surface S. Respectively, another resonance cavity ACF is formed in between the microphone and the front part of the housing.

[0035] As already discussed with reference to Figures 2 and 6, the mobile phone may be tilted. To serve this purpose the sealing means are designed in the manner described with reference to Figure 6, the distance between PL and PLB may be shorter than the distance between PR and PRB. This means that the PL, PLB part could be more elastic than the PR and PRB so that the PL, PLB part of the sealing means compresses more. The aperture between the flat surface 204 and the sealing means 203 shall be preferably good in order to enhance the omni-directional mode. It is possible that if the sealing is not proper the direction and sensitivity of microphone is reduced to correspond the unidirectional microphone.

[0036] The tilting angle α corresponds to the tilting of the mobile terminal with respect to the flat surface 204. In practice, it may get almost any target value, the target value being thus between a very small angle and a substantially large angle. The magnitude of α has an impact on the aesthetic design of the mobile terminal, and, secondly, it may influence on the contact force between the contact part 601 and the flat surface 204 thus leading to further elasticity considerations. The more the mobile terminal is tilted, the smaller the contact force in the support means is, so requiring more elastic material.

[0037] Figure 9A illustrates another way to realise the invention. The mobile terminal has a housing 200 which includes a battery cover 901. The battery cover may be removable; usually the mobile terminal battery is stored under the cover. For some manufacturers the cover and the battery are combined to a single element. The aperture 202 is marked with a dot in the figure. When the inlet hole is open the mobile phone is in unidirectional directivity mode.

[0038] Figure 9B shows the aperture 202 covered with removable sealing means 1001 according to one aspect of the present invention.

[0039] The removable sealing means 1001 can be stored under the battery cover or back cover 901. Preferably, the aperture 202 and the sealing means 1001 are designed in such a way that the sealing means are tightly matched with the aperture 202. This may be realised, for example, by supplying the sealing means with attaching means and sealing means, so that the sealing means 1001 correspond partially to a flat surface 204. As shown in Figure 8, this mode of the invention is particularly usable when the table 205 or other surface is not essentially flat so that it would not provide a uniform enough flat surface 204 for the sealing means 203 to match at points PLB and PRB shown in Figure 8. Further, the external surface can be used for causing this kind of extra pressure for pushing the sealing means 203 stronger against the housing 200 of the mobile terminal, because the sealing means 1001 would not necessarily be able to cause the sealing to be tight enough hence the weight of the mobile terminal can be used for this purpose to increase the sealing.

[0040] Figure 10 illustrates a measurement result of the spectral energy density versus the frequency of a signal output at a reference point close to the telephone. The measurement has been performed for the inventive mobile phone assembly in unidirectional mode, and in Loudness Rating Guarding Position LRGP in accordance with the standard 3G TS 26.132 (TSG SA; Codec Working Group; Terminal Acoustic Characteristics for Telephony).

[0041] The energy spectrum measured is presented as observed by the codec of the mobile terminal. The upper piecewise linear curve corresponds to the maximal allowed spectral density values, whereas the lower piecewise linear curve illustrates the minimum acceptable spectral energies. As can be seen from the measurement results, the observed spectra lies clearly between the allowed values.

[0042] Figure 11 shows the acoustic behaviour of the same mobile terminal when it is in the second acoustical directivity mode. The spectral energy density shows again that the values are allowed, i.e. that the spectra lies clearly between the agreed limits.

[0043] Both measurements were conducted with a the mobile terminal lying head up on the table, positioned 50 cm from the reference mouth and with 30° azimuth. For the measurements the mobile terminal input gain was not adjusted, though this is not visible in the figure. In practice, because of the different resonance cavities caused by the differences in the acoustical apertures when the mobile terminal is in different acoustical modes, the gain must be adjusted between the modes. The adjustment may be done automatically by detecting the measured signal strength in the codec.

[0044] To summarise, according to one aspect of the present invention, the mobile phone can be made to show dual mode characteristics. First, a microphone is assembled into a housing that includes at least two inlet

IPR2024-01015 Page 00397

holes. At least one of the apertures is designed to be on the user side when the terminal is used as a handset, and the apertures together form acoustical conduits to the both sides of the microphone. The housing is further assembled to include sealing means; with the sealing means at least one of the inlet holes may be sealed together with an external surface, such as a table or any other flat surface. When the mobile terminal is placed on said flat surface, one acoustical aperture is thus closed and the mobile terminal shows acoustical omni-directional characteristics as a second acoustical directivity mode.

[0045] According to one further aspect of the present invention, the assembling step may include the assembling of an elastic gasket. Preferably, said elastic gasket is asymmetric, and the height of the lower part of the gasket being smaller than the height of the higher part of the gasket, the difference being adapted to make the sealing proper taking into account the elevation angle between the bottom of housing and the surface.

[0046] According to still another aspect of the present invention, in addition to the sealing means, the housing may further include a support member to increase the pressure between the sealing means and the surface.

[0047] Although the invention was described above with reference to the examples shown in the appended drawings, it is obvious that the invention is not limited to these, but it may be modified by those skilled in the art without departing from the scope and the spirit of the invention. For example, instead of a microphone, any other transducer or sound detector may be used in connection with the embodiments of the inventive sealing mechanism.

Reference numbers used in the description:

[0048]

100	mobile terminal
101	aperture
102	housing
110	microphone
200	housing
202	second aperture
203	sealing means
204	flat surface
205	table
303	sealing means
304	support member
600	hole for the acoustic inlet
601	contact part
603	aperture part
901	battery cover
1001	removable sealing means

Claims

1. A dual directional mode mobile terminal including a microphone (110), said microphone (110) having been assembled in a housing (200),
characterised in that

   - the housing (200) includes at least a first and a second aperture (101, 202), of which the first aperture (101) is on the user side when the terminal is used as a handset, the first and second apertures (101, 202) forming acoustical conduits (ACF, ACB) to the both sides of the microphone (110); and
   - the housing (200) further includes sealing means (203, 303, 1001) for sealing the second aperture (202) together with an external surface (204), when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a second acoustical directivity mode.

2. A mobile terminal according to claim 1,
characterised in that
the sealing means (203) include an elastic gasket.

3. A mobile terminal according to claim 2,
characterised in that
the elastic gasket is asymmetric, the height h' of a lower part of the gasket being smaller than the height h of a higher part of the gasket, the difference $\Delta = h' - h$ being adapted to make the sealing proper taking into account an elevation angle (α) between a backside surface (102B) of the housing (200) and the external surface (204).

4. A mobile terminal according to claim 1,
characterised in that
in addition to the sealing means (303), the housing (200) further includes a support member (304) to increase the pressure exerted on the sealing means (303) by the external surface (204).

5. A mobile terminal according to claim 1,
characterised in that
the sealing means (1001) further have means adapted for connecting the sealing means to the inlet hole (202) in an acoustically proper way.

6. A mobile terminal according to claim 5,
characterised in that
the mobile terminal has means (200) for storing the sealing means (1001).

7. A method for manufacturing a dual directional mode mobile terminal, the method
characterised in that
it includes the steps of:

IPR2024-01015 Page 00398

- assembling a microphone (110) into a housing (200), wherein the housing (200) includes at least first and second apertures (101, 202), of which a first aperture (101) is designed to be on the user side when the terminal is used as a handset, the first and second apertures thus forming acoustical conduits to both sides of the microphone (110); and
- assembling the housing (200) with sealing means (203) for sealing the second aperture (202) holes together with an external surface (204), when the mobile terminal is placed on said external surface (204), thereby enabling in the mobile terminal a second acoustical directivity mode.

8. A method according to claim 7, **characterised in that** the assembling step includes the assembling of an elastic gasket.

9. A method according to claim 8, **characterised in that** said elastic gasket is asymmetric, the height h' of the lower part of the gasket being smaller than the height h of the higher part of the gasket, the difference Δ = h'- h being adapted to make the sealing proper taking into account the elevation angle (α) between the bottom (102B) of housing and the external surface (204).

10. A method according to claim 7, **characterised in that** in addition to the sealing means (303), the housing further includes a support member (304) to increase the pressure between the sealing means (303) and the surface (204).

11. A method according to claim 7, wherein the sealing means (1001) are further adapted to be connected to the inlet hole (202) in an acoustically proper way.

12. A method according to claim 11, wherein the sealing means (1001) are stored in the mobile phone, and when used, manually placed into the aperture (202).

13. A mobile terminal equipped with an acoustoelectrical transducer (110) arranged in a housing (200) of the mobile terminal, **characterised in that** the housing (200) comprises:

- a user sided first aperture (101) in communication with the acoustoelectrical transducer (110);
- a second aperture (202) also in communication with the acoustoelectrical transducer (110); and
- sealing means (203) for sealing the second aperture (202) when the mobile terminal is placed on an external surface (204) or an object.

14. A mobile terminal according to claim 13, wherein the sealing means (203) are responsive to an external force or pressure.

IPR2024-01015 Page 00399

FIG 1 Prior Art

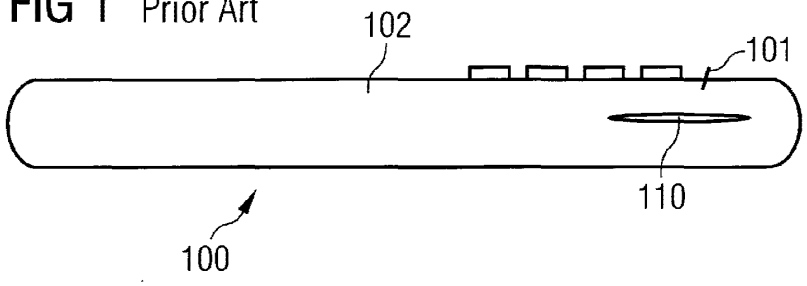


FIG 2

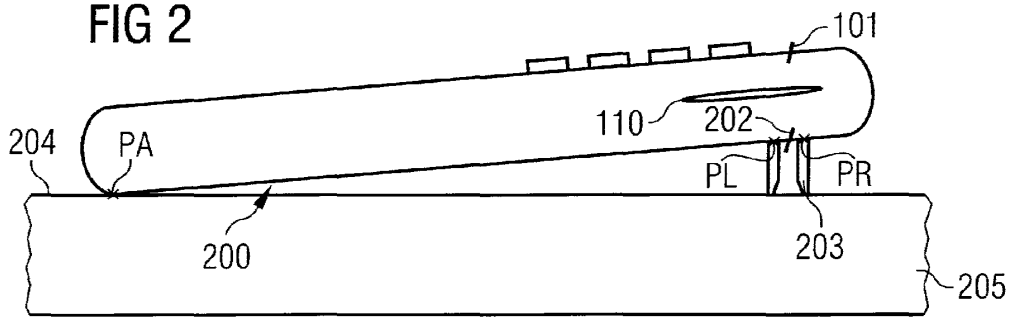


FIG 3

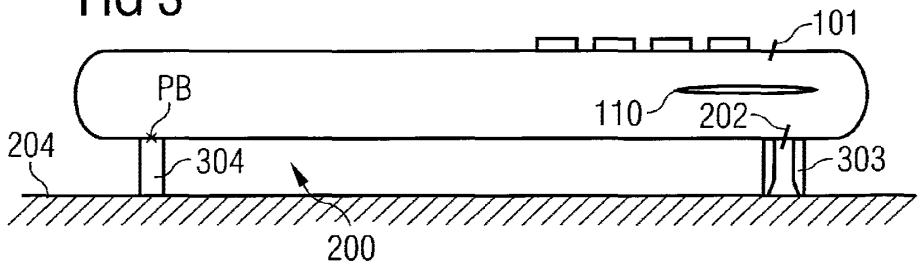
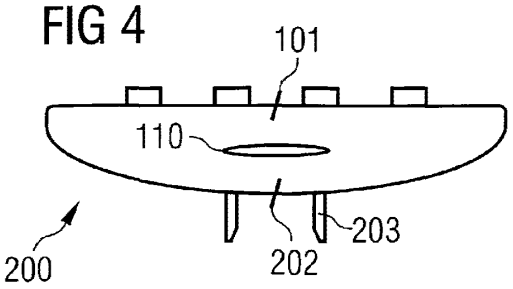


FIG 4



IPR2024-01015 Page 00400

FIG 5

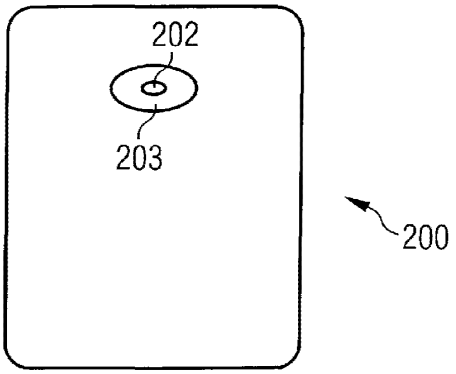


FIG 6

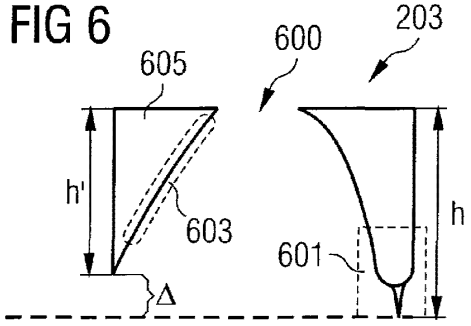


FIG 7

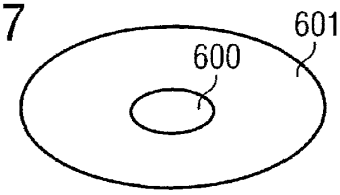
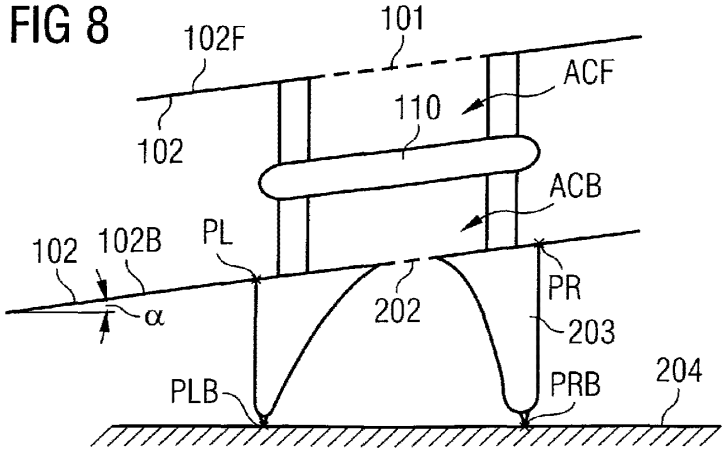


FIG 8



IPR2024-01015 Page 00401

FIG 9A

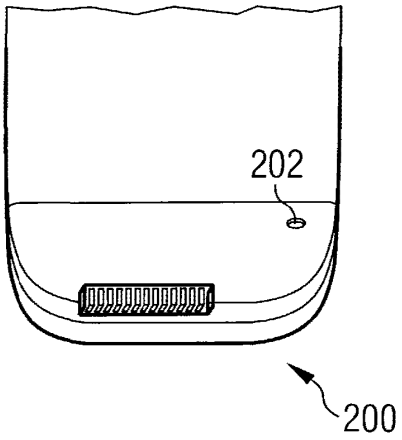
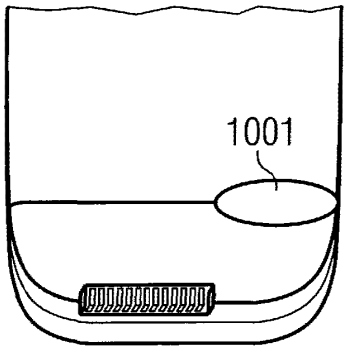
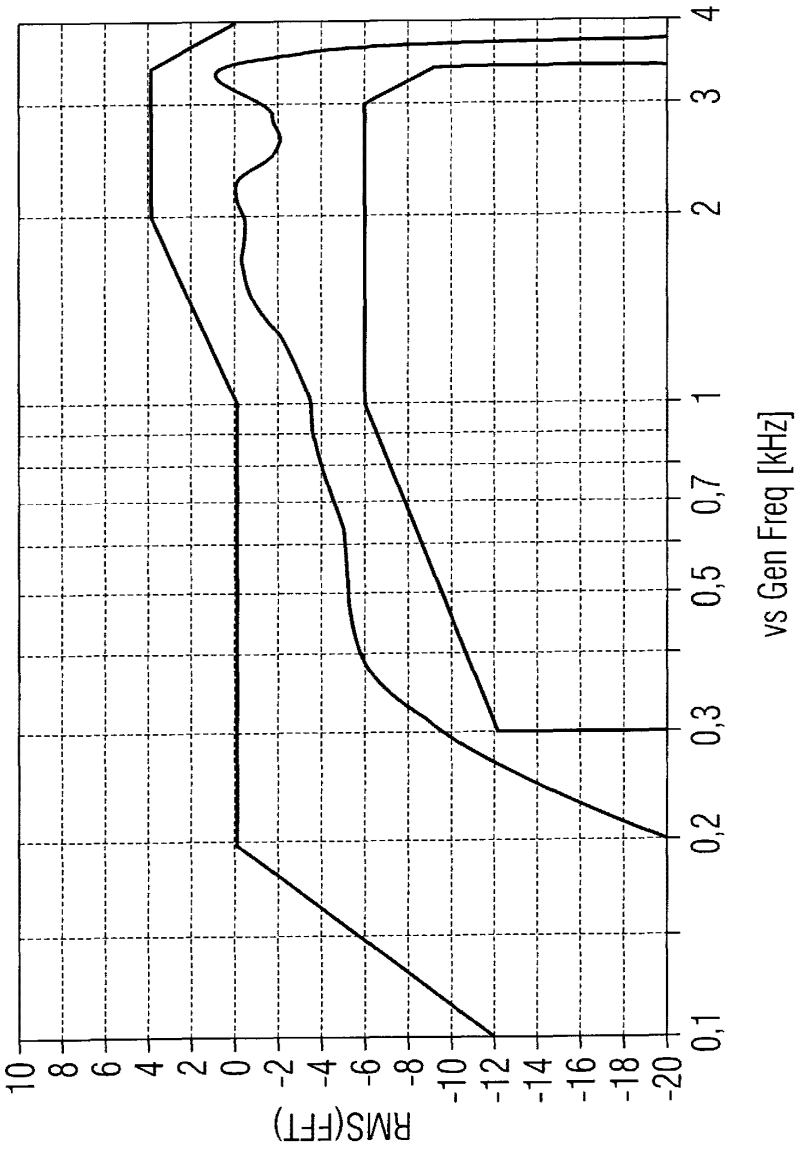


FIG 9B



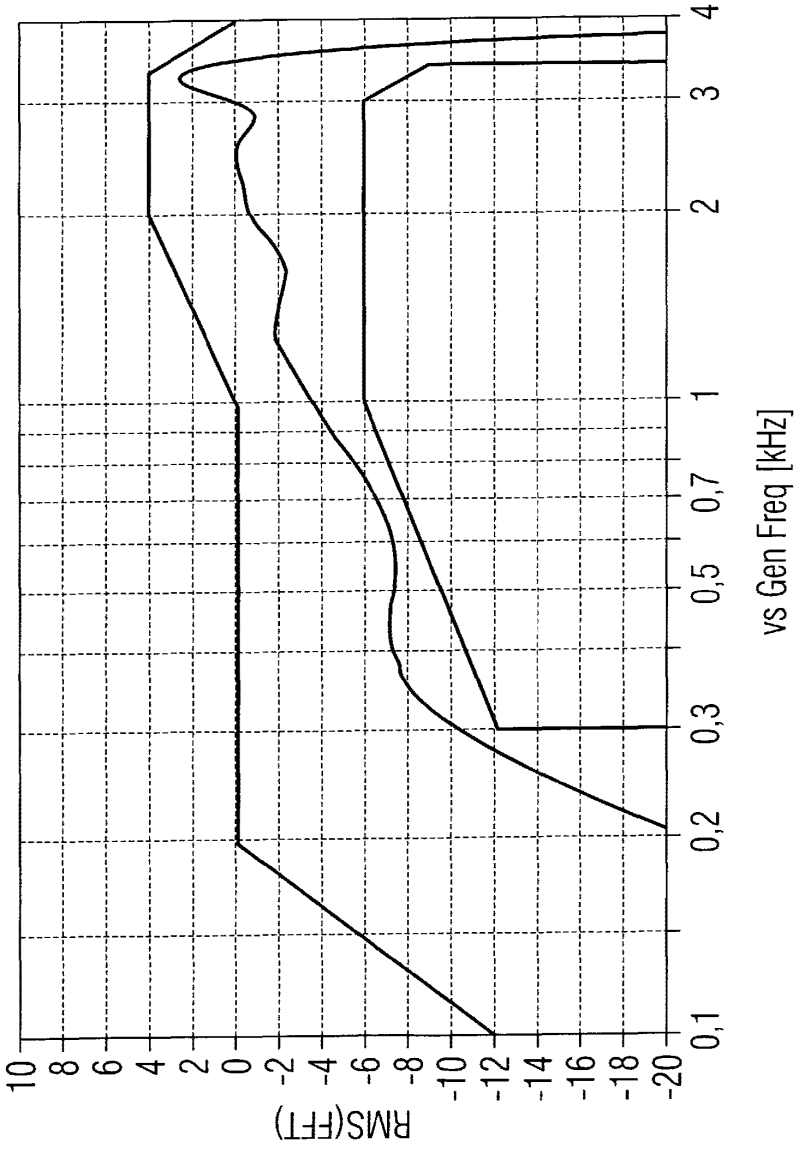
IPR2024-01015 Page 00402

FIG 10



IPR2024-01015 Page 00403

FIG 11



IPR2024-01015 Page 00404

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 02 0319

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Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
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			H04R

The present search report has been drawn up for all claims

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THE HAGUE	31 March 2003	Dionisi, M

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IPR2024-01015 Page 00405

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IPR2024-01015 Page 00406

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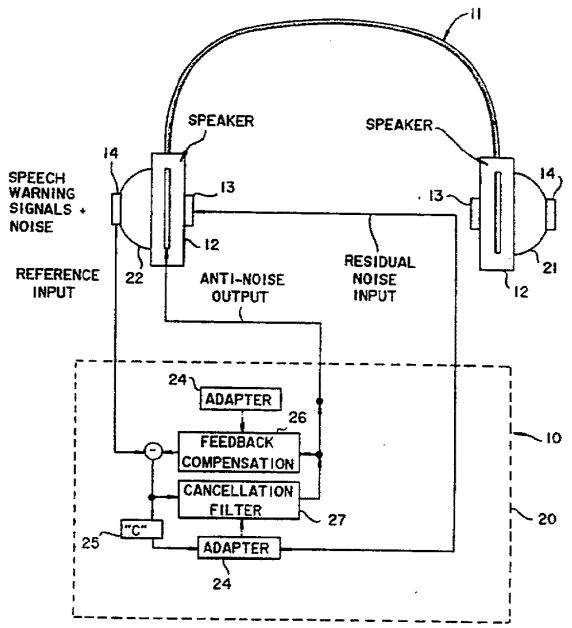
(81) Designated States: CA, JP, UA, US, European patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IT, LU, MC, NL, SE).

Published
With international search report.

(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

$$\underline{x}_k = \underline{r}_k - \underline{y}_{k-1} \cdot \underline{C}_k$$

$$Y_k = -\underline{A}_k r_k$$

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.

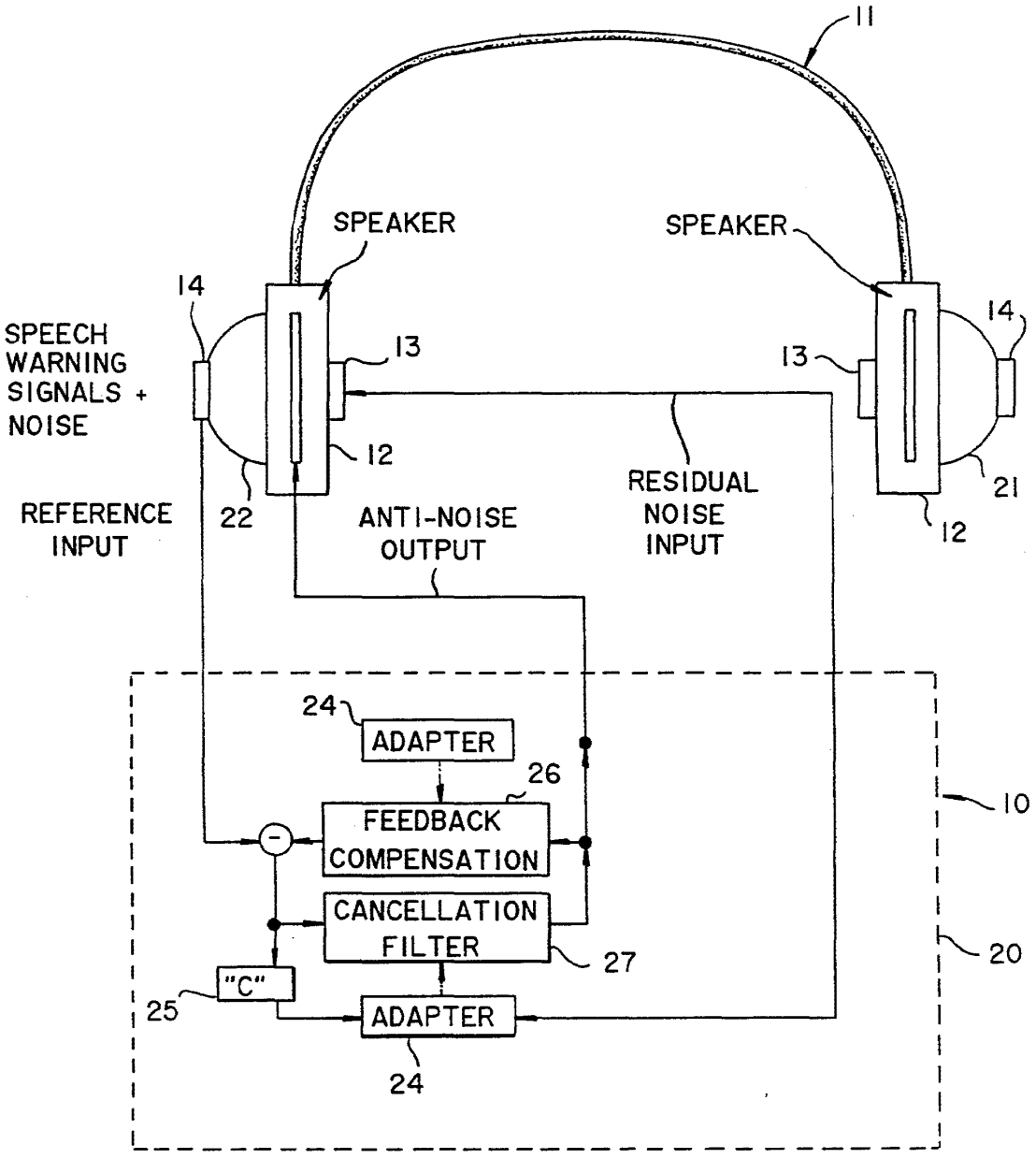

11
SPEAKER
SPEAKER
14
14
SPEECH
WARNING
SIGNALS +
NOISE
13
13
12
21
REFERENCE
INPUT
22
ANTI-NOISE
OUTPUT
RESIDUAL
NOISE
INPUT
12
24
ADAPTER
26
10
FEEDBACK
COMPENSATION
CANCELLATION
FILTER
27
20
"C"
25
ADAPTER
24

FIG.1

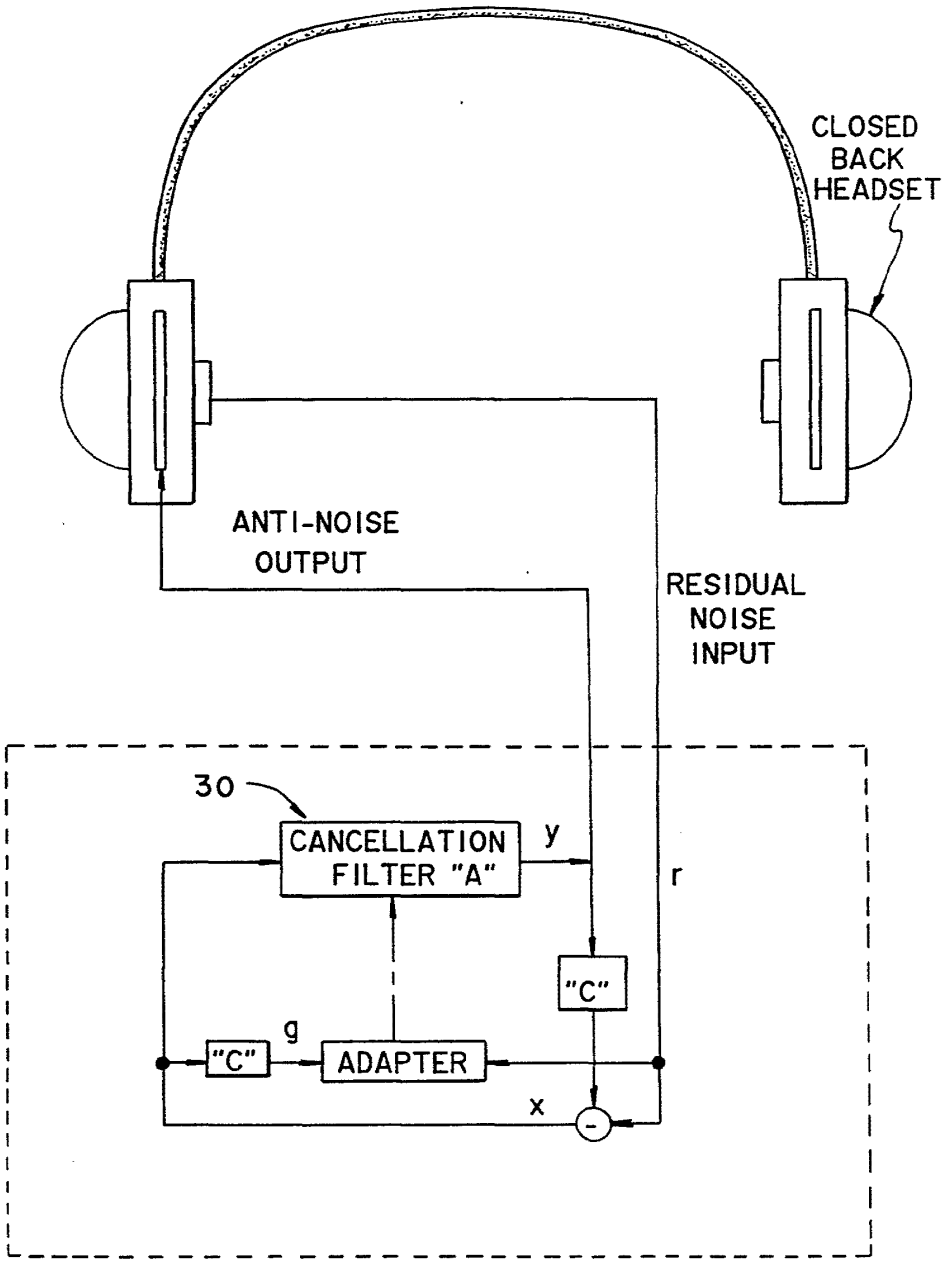

CLOSED
BACK
HEADSET
ANTI-NOISE
OUTPUT
RESIDUAL
NOISE
INPUT
30
CANCELLATION
FILTER "A"
y
r
"C"
g
"C"
ADAPTER
x
-

FIG.2

SUBSTITUTE SHEET

FIG.3

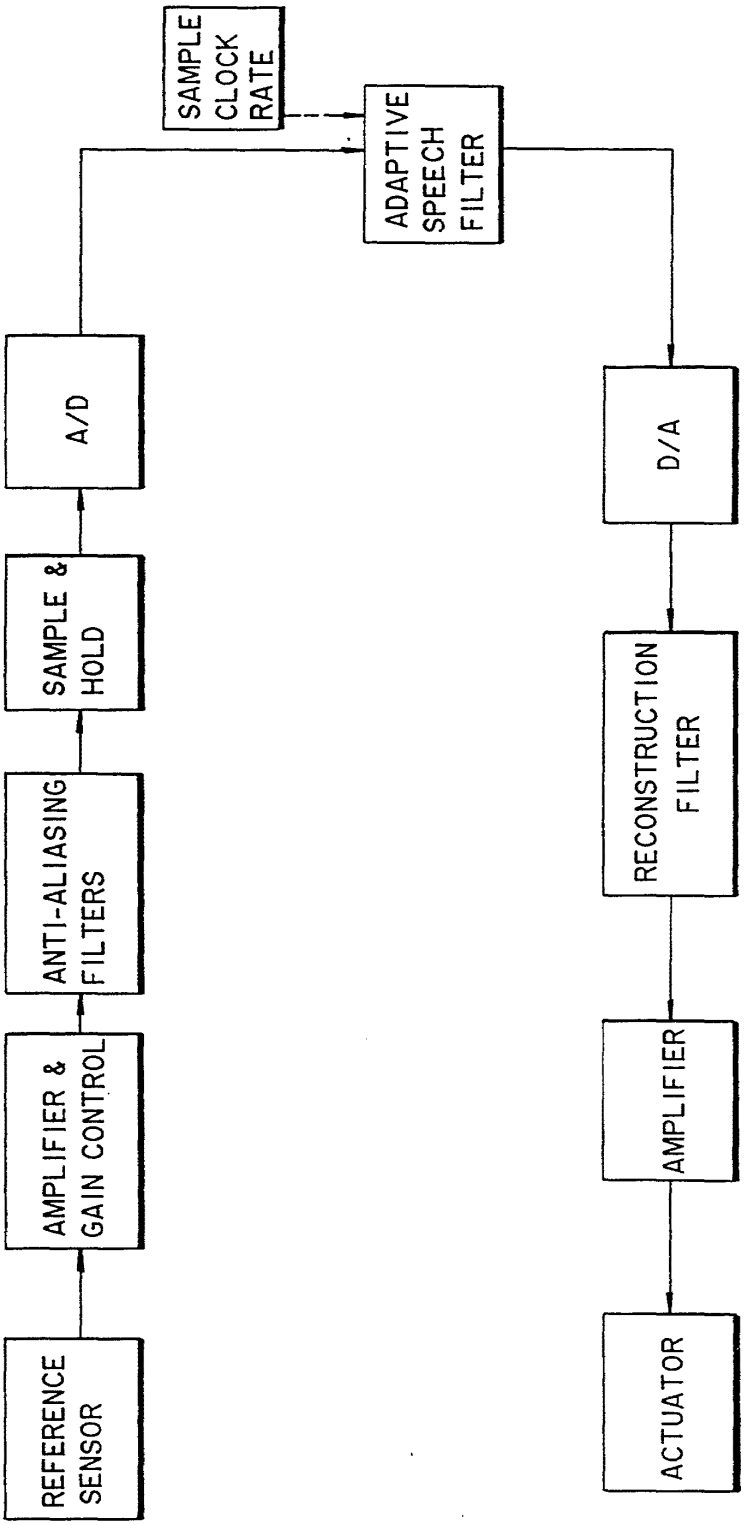

SAMPLE CLOCK RATE
ADAPTIVE SPEECH FILTER
A/D
SAMPLE & HOLD
ANTI-ALIASING FILTERS
AMPLIFIER & GAIN CONTROL
REFERENCE SENSOR
D/A
RECONSTRUCTION FILTER
AMPLIFIER
ACTUATOR

FIG.4

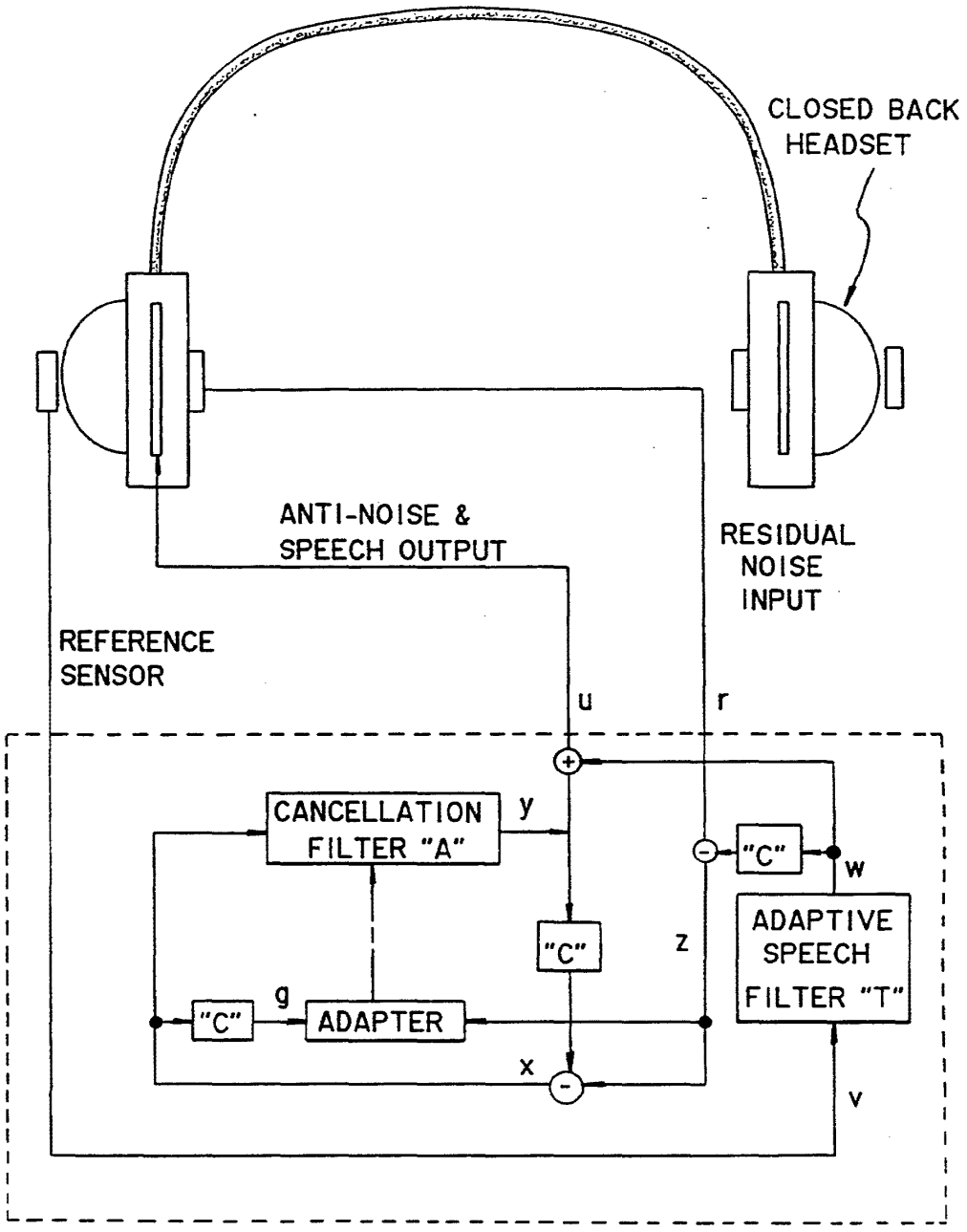

CLOSED BACK HEADSET
ANTI-NOISE & SPEECH OUTPUT
RESIDUAL NOISE INPUT
REFERENCE SENSOR
u
r
CANCELLATION FILTER "A"
y
"C"
w
ADAPTIVE SPEECH FILTER "T"
"C"
z
g
"C"
ADAPTER
x
v

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	Date of mailing of the international search report
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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

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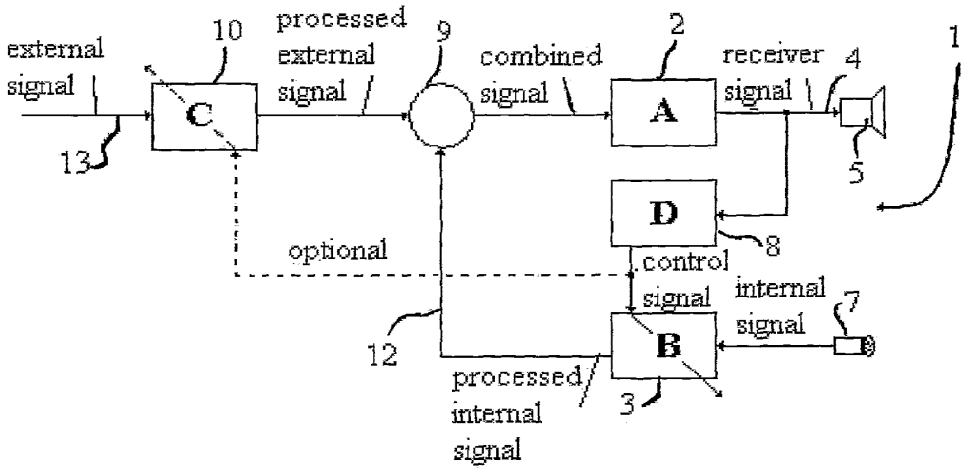
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Declaration under Rule 4.17:
— *of inventorship (Rule 4.17(iv))*

Published:
— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) **Title:** ACOUSTICALLY TRANSPARENT OCCLUSION REDUCTION SYSTEM AND METHOD



(57) **Abstract:** A system and method that reduces the perceptual effect resulting from ear occlusion is disclosed. It comprises of an electro-acoustic feedback network that produces phase cancelling sounds in the ear, where a receiver and a microphone are located. A novel control mechanism is disclosed that controls the response of the feedback network to minimise distortion in the ear while maintaining a desired frequency response for external signals. Devices producing the external signal include hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protectors. The integration of the present invention with these devices improves the user's perception of their own voice.

WO 2006/037156 A1

Acoustically Transparent Occlusion Reduction System and Method

Field of Invention

The present invention relates to an improved occlusion reduction system for applications such as hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protection devices.

Description of Background Art

An electro-acoustic negative feedback scheme was originally presented by H.F Olson, in 1961 in U.S Patent No 2,983,790. A more comprehensive implementation was later proposed by Bose et al, in 1982 in U.S Patent No 4,494,074 under the title "Feedback Control". Using headphones, Bose proposed the idea of an electro-acoustic feedback in the proximity of the ear canal. The concept was later used by Langberg et al, in 1988 in U.S. Patent No 4,985,925, describing a system functioning as a bilateral transducer drive with a shunt feedback correction network. Later on in 1991, Langberg et al, in U.S Patent No. 5,267,321 entitled "Active Sound Absorption" describes an electro-acoustical feedback system, with the receiver acting as both a diaphragm actuator and motion sensor. In 1996, U.S Patent No. 5,774,565 to Benning et al describes an electro-acoustic feedback subsystem with oscillation prevention circuit in the forward path of the loop. In 2002, U.S Patent Application No. 2003/0012391 A1 to Armstrong et al, entitled "Digital Hearing Aids System" discloses a hearing aid including an occlusion processing subsystem.

The occlusion effect is commonly described as a hollow or echoing like sound of a person's own voice. The occlusion effect results from acoustically sealing or partially sealing the ear, or to a greater effect sealing or partially sealing the ear canal from the external acoustic environment. As a result, the occlusion effect creates discomfort and/or an unnatural sound sensation. This problem is commonly reported to clinicians in the

hearing aid industry as it affects a large number of hearing aid wearers (those with mild low-frequency hearing loss). Until now, there were at least two common schemes to decrease the occlusion effect in hearing aids, either using a vent or by increasing the insertion depth of the earmold into the ear canal. To restore naturalness of a hearing aid wearer's voice, vents of up to 3.5 mm in diameter may be employed. These vents need to be sufficiently large so that the residual sound pressure in the canal due to the occlusion effect is not significant. On the other hand, a sufficiently large vent limits the hearing aid amplification due to oscillations created by positive feedback occurring around the loop defined by an external microphone, amplifier, receiver, and path through the vent back to the external microphone.

The soft tissue in the ear canal is excited by vibration energy propagated by the skull and jaw due to the wearer's voice and this results in an increased sound pressure within the occluded of partially occluded ear relative to an open ear. Another solution is to insert the earmold further into the ear canal to fill the cartilaginous portion of the canal and hence reduce the occlusion effect at its source. However there are a number of practical problems relating to the deep insertion of an earmold, for instance physical discomfort.

While some of the aforementioned patents include methods to alter the response of the electro-acoustic feedback system, none of these patents incorporate methods to compensate for the effect that response alteration within the feedback system has on external signals such as from an external microphone or hearing aid processor. In addition, the aforementioned patents do not provide a mechanism to prevent the receiver from overloading by sensing the level of signal at the optimum point, being the input to the receiver.

Summary of Invention

It is an object of the present invention to provide an improved occlusion reduction system for applications such as hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protection devices.

In accordance with a first aspect of the present invention, there is provided an electro-acoustic system comprising of: an electro-acoustic circuit including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit, (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A), for modifying the combined signal to produce an receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal such as from an external microphone or hearing aid processor to produce an external processed signal; a fourth electronic circuit (D), for estimating the level of the receiver signal optimally from the receiver signal and producing a control signal; wherein the processing performed by the second electronic circuit (B) preferably is controlled by the control signal.

The negative-feedback loop preferably can include a filter that alters the open loop response so that the real component of the response is large and negative at frequencies where the occlusion effect can be typically the greatest. Consequently reducing all signals introduced into the closed loop within this frequency range.

The third electronic circuit (C) preferably can include filtering that provides compensation for the closed loop response of the negative-feedback loop.

Optionally the filtering provided by the third electronic circuit (C) adapts to compensate for changes in the closed response of the negative-feedback loop.

Optionally the control signal can control the response of the first electronic circuit (A) while controlling the response of the third electronic circuit (C).

The microphone for generating the internal signal optionally can be coupled to the occluded or partially occluded ear by a tube. The receiver optionally can be coupled to the occluded or partially occluded ear by a tube. The microphone and receiver optionally can be combined in a single unit or jointly coupled to the occluded or partially occluded ear by a common tube. At least one of the electronic circuits can be implemented digitally. At least one of the digital electronic circuits can performed signal processing at a sampling rate at least four times that of the bandwidth of the signal.

In accordance with a further aspect of the present invention, there is provided an electro-acoustic system including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A) for modifying the combined signal to produce a receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal such as from external microphone to produce a processed external signal; a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate; wherein the control signal controls the processing performed by the second

electronic circuit (B) and controls the processing performed by the third electronic circuit (C).

In accordance with a further aspect of the present invention, there is provided an electro-acoustic system including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A) for modifying the combined signal to produce a receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal to produce a processed external signal; a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate; wherein the control signal controls the processing performed by first electronic circuit (A) and controls the processing performed by the third electronic circuit (C).

In accordance with a further aspect of the present invention, there is provided a method of providing a negative feedback loop for an electro-acoustic system, the method including the steps of: (a) generating an internal signal representing an acoustical signal located within or closely coupled to an occluded or partially occluded ear; (b) modifying the internal signal to produce a processed internal signal; (c) combining the processed internal signal with a processed external signal to produce a combined signal; (d) modifying the combined signal to produce a receiver signal; (e) generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; (f) modifying an external signal to produce a processed external

signal; (g) estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate; wherein the degree of modification in step (b) is controlled by the control signal. The control signal can be utilised to control the amount of modification occurring in step (f).

In any of the aforementioned aspects of the present invention the fourth electronic circuit (D) optionally produces a control signal using estimates of the signal levels from the internal signal or the processed internal signal and the receiver signal or the combined signal.

Brief Description of the Drawings

Preferred embodiments of the present invention will now be described with reference to the accompanying drawings in which:

Fig. 1 is a block diagram of the occlusion reduction scheme of the preferred embodiment;

Fig. 2 is a schematic diagram of an analog electronic circuit for the electro-acoustic negative feedback loop;

Fig. 3 is a schematic diagram of an analog adaptive pre-compensation electronic circuit;

Fig. 4 is a schematic diagram of an analog level estimator and control signal generation electronic circuit;

Fig. 5 is a schematic diagram of a digital implementation of the occlusion reduction scheme;

Fig. 6 is a schematic diagram of an adaptive IIR filter;

Fig. 7 is a schematic diagram of a level estimator and control signal generator ;

Fig. 8 is a graph of the simulated closed loop response in a Zwislocki coupler;

Fig. 9 is a graph of the measured response of the effective gain reduction inside a real ear;

Fig. 10 is a graph of the measured responses from an external signal to a Zwislocki coupler microphone; and

Fig. 11 is an illustration of the occlusion reduction scheme implemented as an in-the-canal hearing aid device.

Detail Description of the Preferred and Other Embodiments

The preferred embodiment operates to reduce the level of signals generated within an electro-acoustic negative feedback loop, such as signals produced by vibration in the ear canal walls due to bone conduction of a user's voice. The reduction occurs in the low to mid audible frequencies, typically ranging from 80 Hz up to 1 kHz, where the occlusion effect is more predominant and perceptually apparent.

A negative feedback scheme is provided which combines a processed externally generated signal such as from an external microphone or a sound system with a processed internal signal such as from a microphone located within or closely coupled to the occluded or partially occluded ear. The combined signal after optional further processing is applied to a receiver located within or closely coupled to the occluded or partially occluded ear. The level of the signal to be applied to the receiver is optimally estimated either from the signal applied to the receiver or from the combined processed external and processed internal signals. Optionally this signal level can be estimated from signals at other points within the scheme. High signal levels applied to the receiver may produce a distorted output from the receiver. This distortion is reduced by applying active gain reduction in the feedback path in response to estimated high signal levels being present. Optionally low signal levels can be detected so that noise inherent in the negative feedback components such as the internal microphone can be minimised by

applying active gain reduction in the feedback path. Thus, the high and the low signal level thresholds for gain reduction in the loop can depend on the dynamic operational range of the discrete components within the system. In addition, an adaptive equalisation filter is applied to the external signal to compensate for variations of the transfer response of the closed loop.

The preferred embodiment includes of a microphone to sense the sound pressure in the ear. The preferred embodiment also includes a novel design of estimating the level of the signal to be applied to the receiver and reducing the gain in the feedback path of the loop when this level is high. This mechanism effectively improves the robustness of the closed loop system by limiting excessive driving levels being applied to the receiver. A filter within the feedback path of the loop yields the necessary phase and gain around the loop to generate a phase cancelling sound in the ear without creating acoustic feedback. This negative feedback response also causes a sound pressure reduction for external signals thus affecting the response from the external processed signal to the receiver signal. As a result, an adaptive pre-compensation filter is provided. The adaptive pre-compensation filter performs adaptive equalisation to maintain a constant frequency response between the external signal and the receiver signal in response to changes in loop response.

In addition, the negative feedback response also causes a reduction in external sounds transmitted through a vent or leakage, thus minimising both effects.

Referring initially to **Fig. 1**, a schematic diagram 1 illustrates the occlusion reduction scheme of the preferred embodiment. This electronic circuit can be encapsulated in an earmold 100 as discussed hereinafter with reference to **Fig. 11**. The earmold optionally contains tubing for coupling the receiver to the ear. The earmold optionally contains tubing for coupling the internal microphone to the ear. The

microphone and receiver optionally can be combined into a single unit or jointly coupled to the occluded or partially occluded ear by a common tube. Optionally these tubes can be protected from wax blockage using wax guards. The lengths of these tubes are preferably as short as possible to minimise delays around the feedback loop, but can be any length. The earmold can optionally contain an open vent to depressurise the ear thus reducing the sensation of stuffiness in the ear. Optionally, the vent can be fitted with an acoustic damper for compensating for the vent resonance that may affect the closed loop response.

The internal signal from an internal microphone 7 is proportional to the ear canal sound pressure. This internal signal is filtered in a feedback loop shown as first electronic circuit **(A)** 2 and second electronic circuit **(B) 3** to produce the receiver signal 4 output to the receiver 5. The aim is to produce cancellation around the loop, limited to a given low to mid frequency band.

An analog implementation of the occlusion reduction scheme is depicted in **Figs. 2-4** and a digital implementation is shown in **Figs. 5-7**.

In addition, **Fig. 1** shows fourth electronic circuit (D) 8 that estimates the level of the receiver signal to produce a control signal. Optionally the input to the fourth electronic circuit (D) 8 can be obtained from the output of a combiner 9 with appropriate compensation for the effects of the first electronic circuit (A) 2. The response of the negative feedback loop is controlled by the control signal. The estimated level of the receive signal 4 can be compared to a reference level which is not shown in this figure. The control signal reduces the gain in the loop as the level of the receive signal increases above the reference level. The reference level is set to a level to minimise distortion occurring within the loop.

The combined signal within the loop results from a combination 9 of processed internal signal 12 and processed external signal. The processed external signal results from a filtered external signal 13. The pre-compensation filter 10 depicted as the third electronic circuit **(C)** 10 in **Fig. 1**, equalises the magnitude of the transfer function from the external signal 13 to the receiver signal 4 so that it is approximately constant across frequency, assuming a fixed closed loop response. Optionally, the filter in the third electronic circuit (C) 10 is adaptively controlled by fourth electronic circuit (D) 8 so that the magnitude of the transfer function from the external signal 13 to the receiver signal 4 is approximately constant across frequency regardless of changes in the closed loop response.

Optionally the fourth electronic circuit (D) uses estimates of signal levels from the receiver signal or combined signal and from the internal signal or processed internal signal to produce a control signal to control the loop response. This control signal can control the loop response by directly applying gain reduction to the first electronic circuit (A) or to the second electronic circuit (B). The control signal produce from this arrangement can be used in combination with the fixed or adaptive third electronic circuit (C).

Analog Implementation

Fig. 2, depicts an analog implementation 20 of the negative feedback loop of **Fig. 1**. Firstly, the internal signal is applied to a high pass filter 21 and lead-lag filter 22 in order to reduce effects from jaw movements and very low frequency instability in the loop. The equations for the transfer function in relation to the corresponding discrete components are shown below:

$$H_{HP}(j\omega) = \frac{j\omega \cdot R_1 C_2}{1 + j\omega \cdot R_1 \times C_1}$$

if $R_4 = R_5, C_2 = C_3$ then the lead lag transfer function equation is shown below

$$H_{Lead-Lag}(j\omega) = \frac{(R_3 + R_6) + j\omega \bullet C_2 \bullet (2R_3R_6 + R_6R_4 + R_3R_4)}{R_6 + j\omega \bullet C_2 \bullet (R_3R_6 + R_6R_4)}$$

The next filtering stage reduces the dominant transducer resonance within the loop and provides greater open loop gain at frequencies at which the occlusion effect is greatest. This is achieved by using a bi-quadratic filter 24, and placing a complex pair of zeros at the dominant transducer resonance frequency followed by a pair of real poles to provide low frequency emphasis in the open loop response.

In the arrangement of Fig. 2, if we allow $R_8 = R_{10}, C_4 = C_6$, $R_9 = R_{11}, C_5 = C_7$ and $R_{12} = R_{13}, C_8 = C_9$,then the biquadratic transfer function equation can be computed as follows

$$H_{bi-quad}(j\omega) = \frac{(j\omega)^2 C_4R_9C_5 + j\omega\left(C_4 + \frac{C_5R_9}{R_8} - C_5\right) + \frac{1}{R_8}}{(j\omega)^2 C_5R_9C_8 + j\omega\left(\frac{C_5R_9}{R_{12}} + C_8\right) + \frac{1}{R_{12}}}$$

To determine the pole zero placement, the biquadratic transfer function equation can be directly related to a well known second order transfer function equation.

$$K(j\omega) = \frac{(j\omega)^2 + 2\xi_z\omega_z j\omega + \omega_z^2}{(j\omega)^2 + 2\zeta_p\omega_p j\omega + \omega_p^2}$$

Where the ω_p and ω_z are the location of the pole and zero frequencies. Similarly ξ_z and ζ_p are the damping factors for the poles and zeros respectively. Thus, this relationship can be used to position the poles and zeros at the desired frequencies with the desired damping.

With appropriate phase compensation, amplification is added to the loop. The gain amplification is chosen in accordance to a well-known gain and phase margin criteria (e.g. Linear Control Systems Analysis and Designs, John J.D'Azzo, Constantine H.

Houpis, 2nd Edition, McGraw-Hill, 1981). A loop gain of less than or equal to –3 dB is chosen at frequencies likely to produce positive feedback. The filtering arrangement of **Fig. 1** shown in **Fig 2**, produces a gain greater than unity, for an open loop response between 80 Hz and 1 kHz and less than unity at other frequencies, where positive feedback may occur. The frequency band ranging from 80 Hz up to 1 kHz is an appropriated choice as the occlusion effect is subjectively more apparent at these frequencies, as described in literature (e.g. Hearing Aids, Harvey Dillon, Boomerang-Press, 2001).

The bi-quadratic filter 24 is followed up by a voltage controlled variable resistance 25, referred to as **VVR**. This control produces up to – 20 dB of gain around the loop, by controlling the voltage at the gate of the JFET. Thus, the gain across the VVR network is found by observing that the JFET is series with R_7 forms a voltage divider, with its gain given by:

$$Gain_{VVR} = \frac{R_{Q1}}{R_7 + R_{Q1}}$$

Where R_{Q1}, is the variable resistance across drain to source junction.

The VVR is followed by a bufferstage 26, that is subsequently followed by a variable resistor, R_{14}. 27. This latter resistor is used to fined tune the gain around the loop manually. The variable resistor 27 is followed by an amplifier and a combiner 9, combining a processed external signal 28 with the processed internal signal 12. Finally, this buffer functions as a voltage controlled voltage source to the receiver, optionally a **class D** amplifier may be used.

Referring now to **Fig. 3**, there is illustrated the precompensation circuit 10 of **Fig. 1** in more detail. The circuit may be used to pre-compensate an external signal 13 to produce a processed external signal 28. The external signal is pre-compensated with a

fixed high pass filter 32, a variable notch filter 33, and an adjustable gain control 34. Note that the notch and the gain are also controlled with voltage variable resistances, using JFET transistors.

If $C_{13} = C_{14}$,then the transfer function equation for the notch filter, positioned between U6 and U7 is shown below

$$H_{Notch}(j\omega) = \frac{1 + j\omega(2R_x C_{13}) + \left(R_x R_{19} C_{13}{}^2\right)\cdot(j\omega)^2}{1 + j\omega(R_{19} + 2R_x)C_{13} + \left(R_x R_{19} C_{13}{}^2\right)\cdot(j\omega)^2}$$

Where $R_x = R_{20} + R_{Q3} + R_{Q4}$ and $R_{Q3} + R_{Q4}$ is the combined resistance across drain to source junction of the two JFET transistors. Also note that the amplification at U7 is controlled by the source to drain resistance at Q5, and the low pass filter between U5 and U6 is essentially switched on and off by Q2.

The voltage variable resistances shown in **Fig. 2** and **3** are driven by the circuit 40 shown in **Fig. 4**. This circuit functions as a signal level estimator and threshold detector and produces a control voltage 41. In this circuit the receiver signal 4 is compared to a reference voltage 43, shown as v-Ref in the figure, at the negative input of U8 (44). If this voltage does not exceed the reference level, the current source, made up by BJ3 and BJ4 transistor pair pulls down the voltage across the RC tank towards the negative supply voltage, with a time constant equal to the release time. Note that the voltage across the RC tank feeds directly into the gate of the JFET control transistor in **Fig. 2** from U9 buffer (1), resulting in a VVR signal 25. Similarly from U10 and U11 buffer (2) and buffer (3), results is an inverted VVR signal 35, that is directly applied to the gates of all the JFET transistors shown in **Fig. 3**. As the voltage exceeds the threshold reference level, the current source, shown as BJ6 and BJ5 transistor pair, charges the RC tank, towards the base voltage at BJ6, with a time constant set by the attack time. The

attack and release time constants can be approximated by finding the appropriate ratio between resistors R_{31} and R_{30}.

The voltage across the capacitor C_{15} ranges from a voltage at the based of BJ7 , say 0.7V or fully charged to the voltage at the collector of BJ3, say -2.8V or fully discharged, then the change in voltage $\Delta V \approx 3.5V$, from which the current needed to produce the discharging rate can be evaluated as follows:

$$\text{Discharging } I = C_{15} \times \frac{\Delta V}{\text{Re}lease\_Time} = 7uA$$

thus effectively $I_{charging} \approx 11 \times I_{discharging}$

Then $R_{31} = \frac{0.7}{77u} \approx 8.2k$ $\qquad R_{30} = 11 \times R_{31} \approx 91k$

Digital Implementation

A digital implementation of the arrangement of **Fig. 1** typically requires the signal within the loop to be oversampled to reduce the delay introduced into the loop by sampling. The signal processing performed by electronic circuits A, B, C and D and the combiner 9 may be performed digitally. The digital processing optionally is performed by a digital signal processor which is instructed to perform the signal processing. Various techniques are available to those skilled in the art to perform these processes. Examples of these are performing the filtering operations using FIR or IIR filters or modifying the signals in the frequency domain using the short-time Fourier transform techniques. One implementation of the scheme described for **Fig 1** is shown in **Figs. 5-7**.

In **Fig. 5,** after A/D conversion 50, the internal signal is filtered with an IIR filter 51 in order to provide the required gain and phase compensation, with the same criteria applied in the description of **Fig. 2**.

In this digital implementation the level of the receiver signal 4 is directly estimated and the control signal *r(n)* is produced by the level estimator and control signal generator

8 which is shown in greater detailed in **Fig. 7.**. Referring now to **Fig. 7,** the receive signal level is estimated with a digital 1st order switchable time-constant envelope detector. The envelope signal is compared 71 to a threshold value and the maximum of either the ratio of its exceedance of this threshold or unity is produced. This maximum is inverted to produce the control signal. As the level of the envelope signal exceeds the threshold, the control signal shown as *r(n)* decreases at a rate determined by the attack time constant λ_{attack}. As the envelope signal decreases in level, the control signal increases at a rate determined by the release time constant $\lambda_{release}$. Alternative signal level compression strategies employing different envelope detector designs well known to those skilled in the art may be used.

The internal signal, after being filtered by the IIR filter 51, is scaled by the control signal *r(n)* to produce the processed internal signal. The processed external signal is added to the processed internal signal by the combiner 9 to produce the combined signal. The combined signal is scaled 2 to produce the receiver signal 4 which after D/A conversion is applied to the receiver 5.

As in the analog description, the negative feedback loop requires pre-compensation so that the gain response measured from the external signal to the receiver signal is not altered by the closed loop response. In **Fig. 5,** the pre-compensation filter 10 is realised as a cascaded IIR filter, with a structural realisation shown in **Fig. 6** . The first filter stage shown in **Fig. 6** is a high pass filter, comprising of a zero at 0 Hz and a pole at f_{HP}, the coefficient being a function of the sampling rate, Fs. The second stage of the filter shown in **Fig. 6** is adaptive, the locations of poles f_p and zeros f_z of the digital filter are varied in accordance with the output of a mapping function *p(n)* 61. Alternatively the damping factors for the poles and zeros can be controlled by two independent mapping functions derived from the control signal *r(n)*. **Fig. 5** shows *p(n)*

as a single output from a polynomial mapping function with input *r(n)*. The polynomial's coefficients are found using an autoregressive analysis of the required compensating response as a function of the control signal *r(n)*.

Any combination of analog and digital electronic circuits is possible in addition to the all analogue and the all digital implementations described with the appropriate conversions between analogue and digital formats. In particular, electronic circuits B, C and D may be implemented digitally with the combiner and electronic circuit A implemented in analog circuitry. Furthermore, the processed internal and processed external signals may be combined while both in a 1-bit format and applied directly to the receiver without electronic circuit A performing any function or with it simply providing the 1-bit drive current for the receiver.

Referring now to **Fig. 8** the simulated closed loop response in a Zwislocki coupler is shown. The scheme yields approximately 16 dB of occlusion reduction at 300 Hz, and 18 dB at 100 Hz. The occlusion reduction response is a measure of the level difference between the sound level in the coupler with the occlusion reduction system active and the sound level in the coupler without any occlusion reduction. Referring to **Fig. 9** occlusion reduction in a real ear is shown. This was calculated by taking the linear average of the energy in 1/12 octave bands recorded using an in-the-ear-canal microphone while a subject talked for 2 minutes. This signal was recorded while the feedback loop gain is maximum, and then again while the loop was open.

Referring now to **Fig. 10**, three responses measured in a Zwislocki coupler are shown. These are the transfer functions from an external signal to the coupler microphone for the compensated 93 and un-compensated responses 92. Also shown is the natural receiver response 94 employing a voltage controlled voltage source to the receiver. Note that the compensated response is not adversely affected by the occlusion

reduction scheme compared to that of the natural receiver response 94. Also similar responses are derived for minimally open vented cavities with the appropriate changes to the mappings, filtering and gain equalisation in electronic circuits A, B, and C.

Referring now to **Fig. 11**, there is shown one form of actual implementation of the present invention as an in-the-canal hearing aid device 100 placed in a user's ear 101 that substantially eliminates the discomfort associated with the occlusion effect. The device 100 includes a sealed or optionally vented 102 earmold that occludes or partially occludes the ear canal. The earmold encapsulates a hearing aid electronic circuit comprising of: a hearing aid processor 103 directly connected to an external microphone 104 and powered by small battery 105. The hearing aid processor includes occlusion reduction electronic circuits A, B, C, D and the combiner. A separate receiver 5 and an internal microphone 7 are also interconnected. These transducer elements are acoustically coupled to the inside of the ear canal with short tubes 108, 109; and protected from ear wax with wax-guarding devices 110, 111.

The forgoing describes preferred embodiments of the present invention. Modifications, obvious to those skilled in the art, can be made thereto without departing from the scope of the invention.

CLAIMS

1. An electro-acoustic system including:

an electro-acoustic circuit including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate;

wherein the processing performed by the second electronic circuit (B) is controlled by the control signal.

2. A system as claimed in claim 1 wherein the negative-feedback loop includes filtering that alters the open loop response so that the real component of the response is large and negative at frequencies where the occlusion effect is most apparent.

3. A system as claimed in claim 2 wherein the third electronic circuit (C) includes filtering that provides compensation for the closed response of the negative-feedback loop.

4. A system as claimed in claim 3 wherein the filtering provided by the third electronic circuit (C) is adaptive and controlled by the control signal.

5. A system as claimed in any previous claim wherein the fourth electronic circuit (D) estimates the level of the internal signal or the level of the processed internal signal and uses this estimate with the estimate of the receiver signal to produce a control signal.

6. A system as claimed in any previous claim wherein the gain of the second electronic circuit (B) is controlled by the control signal.

7. A system as claimed in any previous claim wherein the ear is occluded or partially occluded by an earmold.

8. A system as claimed in claim 7 wherein the earmold contains a vent.

9. A system as claimed in claim 8 wherein the vent contains an acoustic damper

10. A system as claimed in any previous claim wherein the microphone for generating the internal signal is coupled to the occluded or partially occluded ear by a tube.

11. A system as claimed in any previous claim wherein the receiver is coupled to the occluded or partially occluded ear by a tube.

12. A system as claimed in any previous claim wherein the microphone and the receiver are jointly coupled to the occluded or partially occluded ear by a common tube.

13. A system as claimed in any previous claim wherein at least one of the electronic circuits is digital.

14. A system as claimed in any previous claim wherein at least one of the digital electronic circuits uses a sampling rate at least four times that of the bandwidth of the signal.

15. A system as claimed in claims 13 or claim 14 wherein the first electronic circuit (A) includes a digital to analogue converter.

16. A system as claimed in claim 13, claim 14 or claim 15 wherein the second electronic circuit (B) includes an analogue to digital converter.

17. A system as claimed in claim 16 wherein the second electronic circuit (B) includes a digital to analogue converter.

18. A system as claimed in claim 14 wherein at least one of the input signals to the combiner is a binary signal.

19. An electro-acoustic system including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal such as from external microphone to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate;

wherein the control signal controls the processing performed by the second electronic circuit (B) and controls the processing performed by the third electronic circuit (C).

20. An electro-acoustic system including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate;

wherein the control signal controls the processing performed by first electronic circuit (A) and controls the processing performed by the third electronic circuit (C).

21. A method of providing a negative feedback loop for an electro-acoustic system, the method including the steps of:

(a) generating an internal signal representing an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

(b) modifying the internal signal to produce a processed internal signal;

(c) combining the processed internal signal with a processed external signal to produce a combined signal;

(d) modifying the combined signal to produce a receiver signal;

(e) generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

(f) modifying an external signal to produce a processed external signal;

(g) estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate;

wherein the degree of modification in step (b) is controlled by the control signal.

22. A method as claimed in claim 21 wherein the control signal is utilised to control the amount of modification occurring in step (f).

23. A method substantially as herein described with reference to any one of the embodiments of the invention illustrated in the accompanying drawings and/or examples.

24. A system substantially as herein described with reference to any one of the embodiments of the invention illustrated in the accompanying drawings and/or examples.

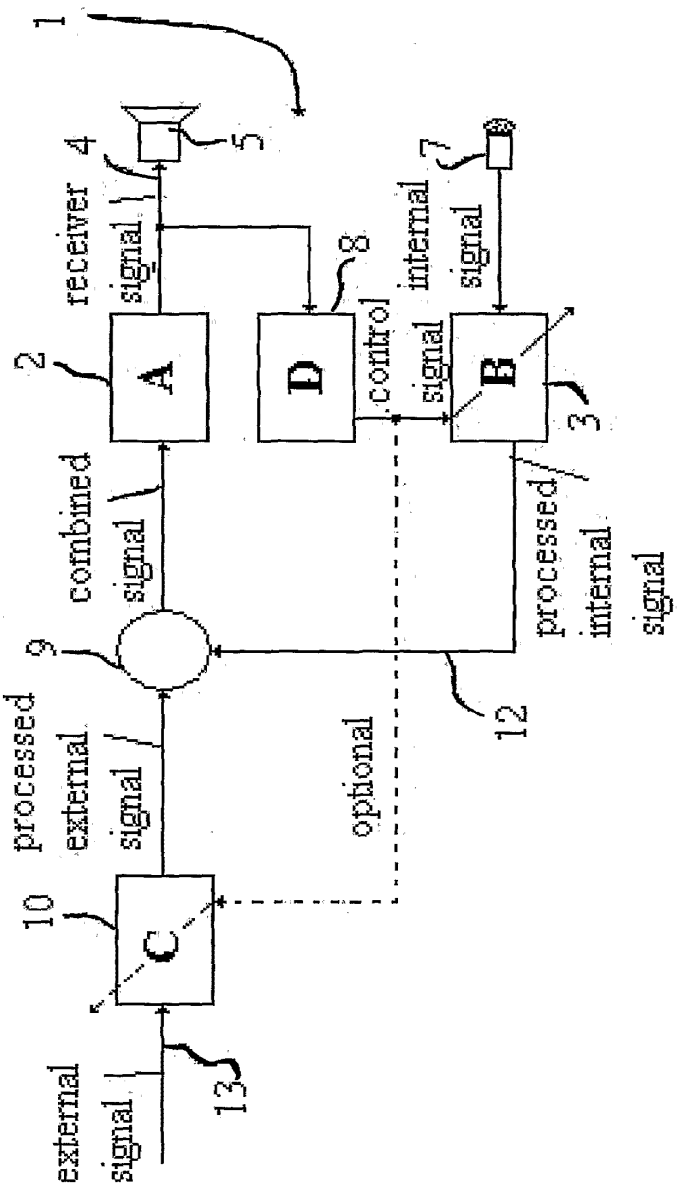

1
5
4
receiver
signal
7
internal
signal
8
2
A
D
control
signal
B
3
combined
signal
9
processed
internal
signal
12
processed
external
signal
optional
10
C
external
signal
13

Fig. 1

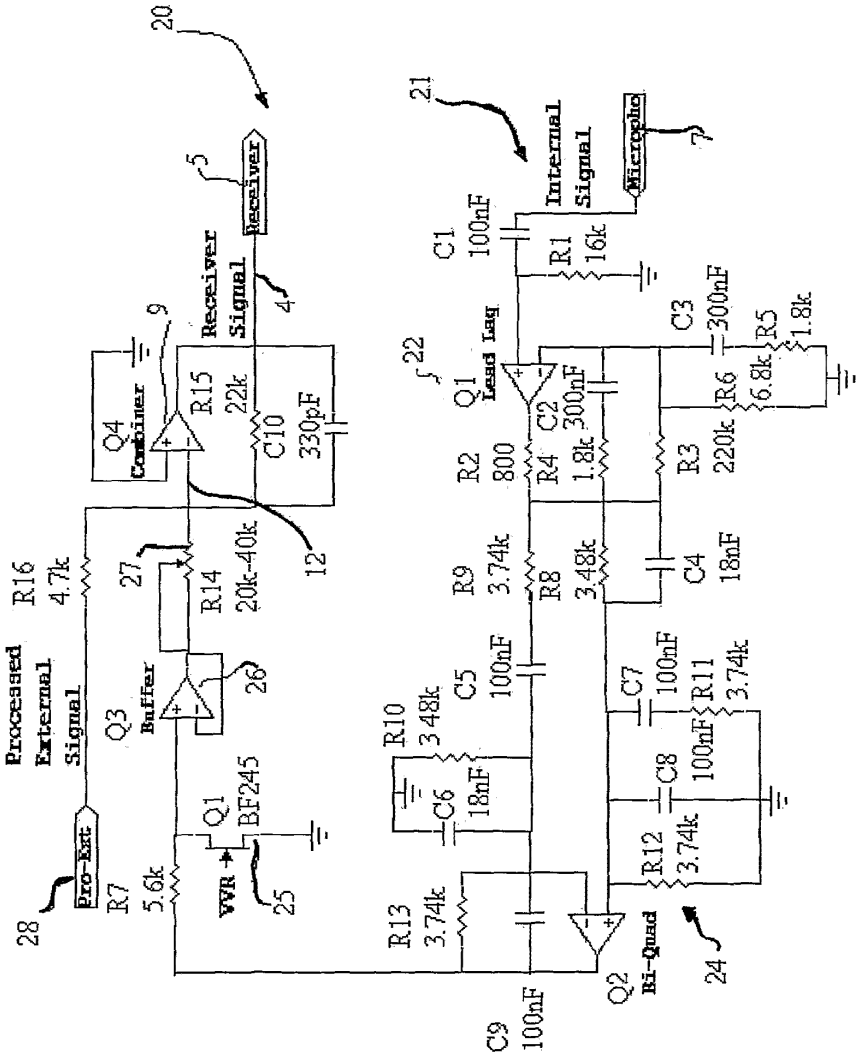

20
21
22
Receiver
Receiver Signal
Internal Signal
Microphone
Processed External Signal
Pre-Ext
Buffer
Combiner
Lead Lag
Bi-Quad
VVR
BF245
R1 16k
R2 800
R3 220k
R4 1.8k
R5 1.8k
R6 6.8k
R7 5.6k
R8 3.48k
R9 3.74k
R10 3.48k
R11 3.74k
R12 3.74k
R13 3.74k
R14 20k-40k
R15 22k
R16 4.7k
C1 100nF
C2 300nF
C3 300nF
C4 18nF
C5 100nF
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C8 100nF
C9 100nF
C10 330pF
Q1
Q2
Q3
Q4

Fig. 2

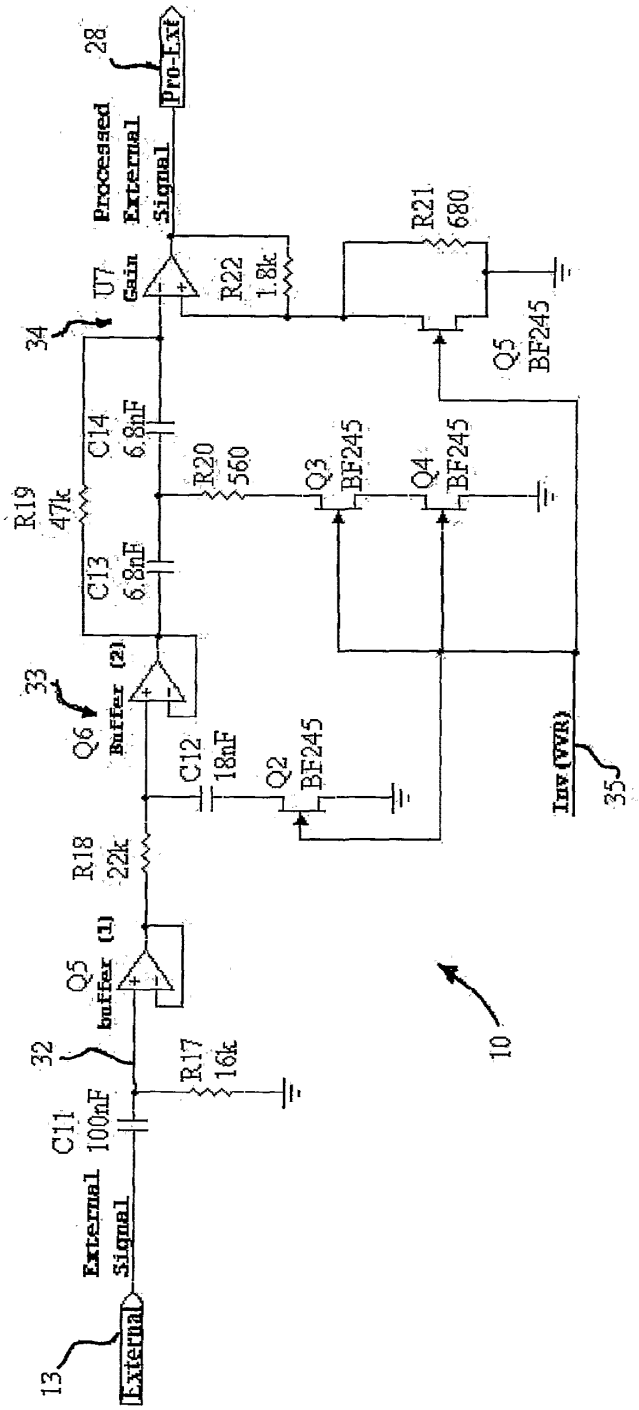

28
Pro-Ext
Processed External Signal
U7 Gain
34
R22 1.8k
R21 680
Q5 BF245
C14 6.8nF
R20 560
Q3 BF245
Q4 BF245
R19 47k
C13 6.8nF
33
Q6 Buffer (2)
C12 18nF
Q2 BF245
Inv (WR)
35
R18 22k
Q5 buffer (1)
32
R17 16k
C11 100nF
External Signal
13
External
10

Fig. 3

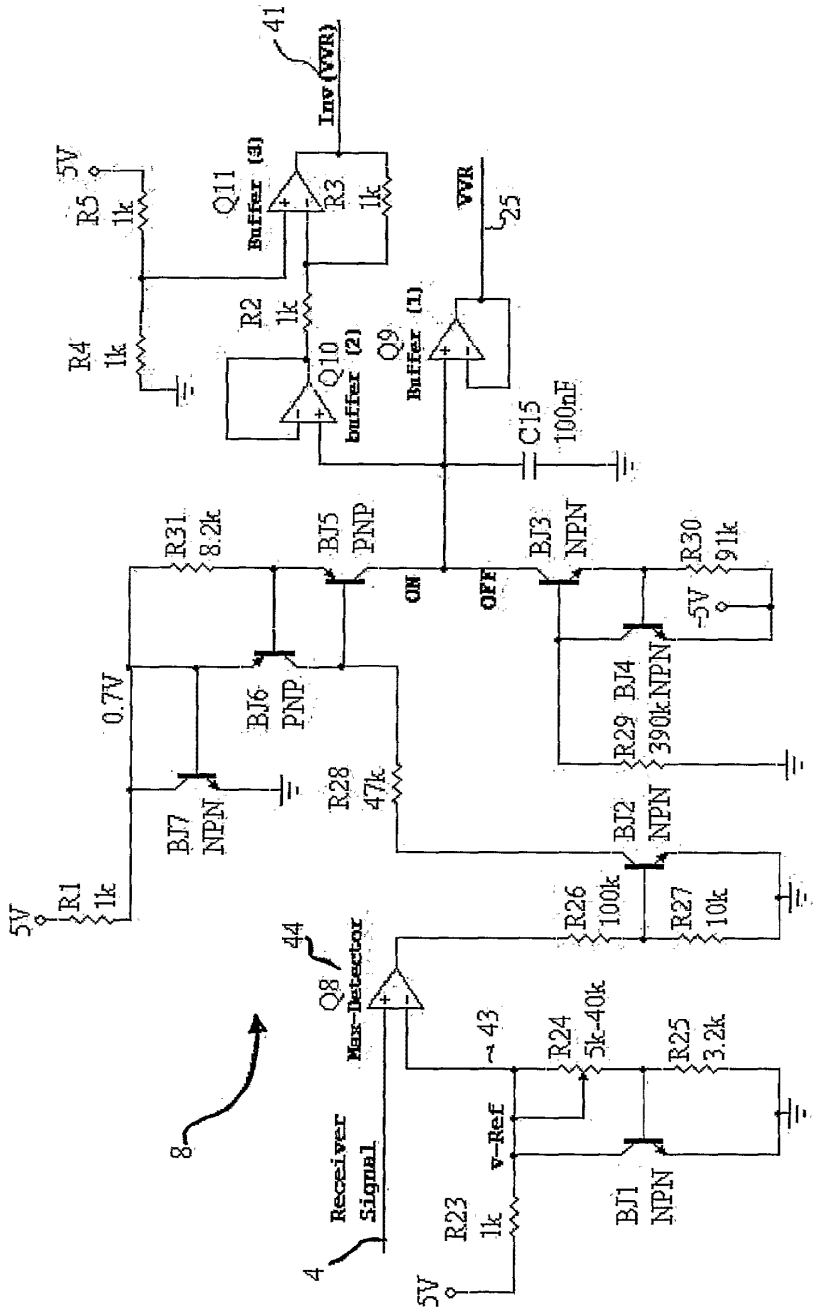

41
Inv (VVR)
5V
R5 1k
Q11 Buffer (3)
R3 1k
VVR
25
R4 1k
R2 1k
Q10 buffer (2)
Q9 Buffer (1)
C15 100nF
R31 8.2k
BJ5 PNP
BJ3 NPN
R30 91k
ON
OFF
-5V
0.7V
BJ6 PNP
BJ4 NPN
R29 390k
R28 47k
BJ7 NPN
BJ2 NPN
5V
R1 1k
R26 100k
R27 10k
44
Q8 Max-Detector
43
R24 5k-40k
R25 3.2k
8
Receiver Signal
v-Ref
BJ1 NPN
4
R23 1k
5V

Fig. 4

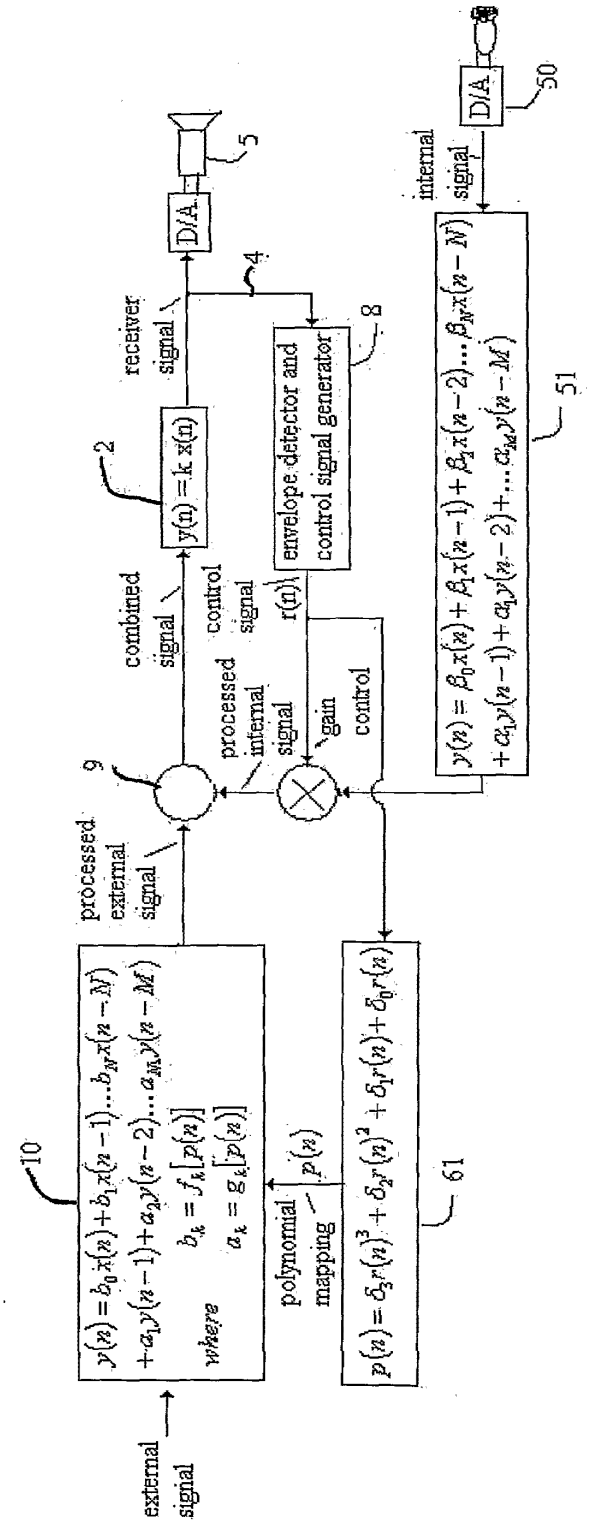

external signal
$y(n) = b_0x(n) + b_1x(n-1) \ldots b_Nx(n-N)$
$+ a_1y(n-1) + a_2y(n-2) \ldots a_My(n-M)$
where $b_k = f_k[p(n)]$ $a_k = g_k[p(n)]$
10
polynomial mapping
$p(n)$
$p(n) = \delta_3 r(n)^3 + \delta_2 r(n)^2 + \delta_1 r(n) + \delta_0 r(n)$
61
processed external signal
9
processed internal signal
gain control
combined signal
control signal
r(n)
2
$y(n) = k\ x(n)$
receiver signal
D/A
5
4
envelope detector and control signal generator
8
internal signal
D/A
50
$y(n) = \beta_0x(n) + \beta_1x(n-1) + \beta_2x(n-2) \ldots \beta_Nx(n-N)$
$+ \alpha_1y(n-1) + \alpha_2y(n-2) + \ldots \alpha_My(n-M)$
51

Fig. 5

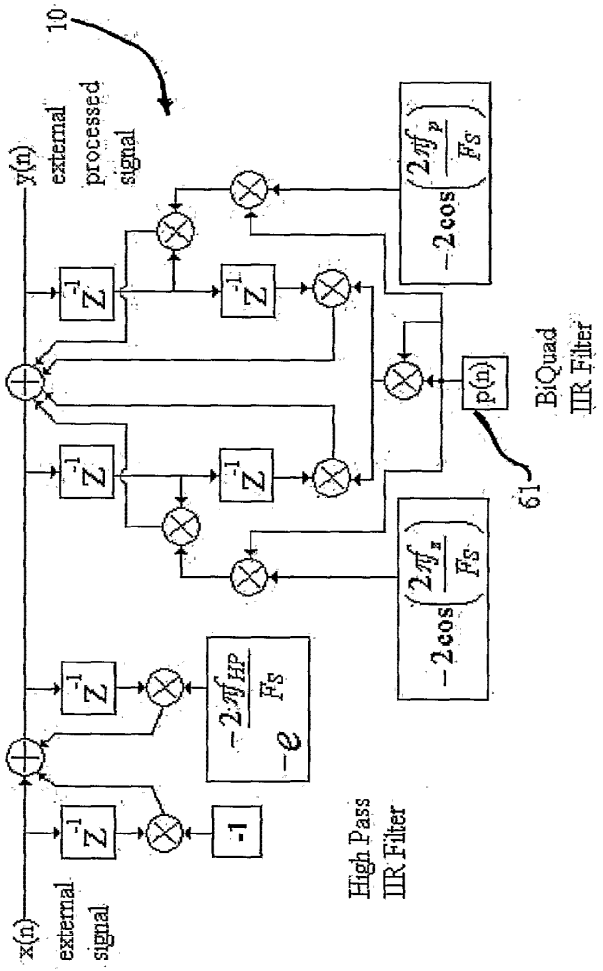

10
y(n)
external
processed
signal
x(n)
external
signal
High Pass
IIR Filter
BiQuad
IIR Filter
p(n)
61
-1
$-2\cos\left(\frac{2\pi f_z}{Fs}\right)$
$-2\cos\left(\frac{2\pi f_p}{Fs}\right)$
$-e^{-\frac{2\pi f_{HP}}{Fs}}$

Fig. 6

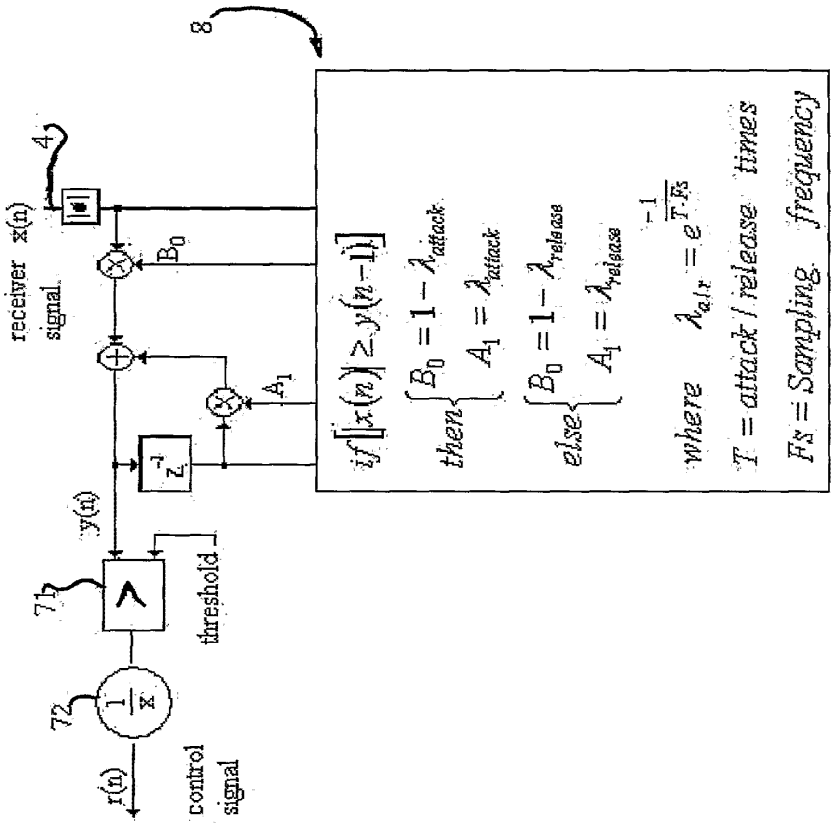

receiver signal x(n)
4
8
B0
A1
y(n)
71
threshold
72
r(n)
control signal
if [|x(n)| ≥ y(n-1)]
then { B0 = 1 - λattack ; A1 = λattack
else { B0 = 1 - λrelease ; A1 = λrelease
where λa/r = e^(-1/(T·Fs))
T = attack / release times
Fs = Sampling frequency

Fig. 7

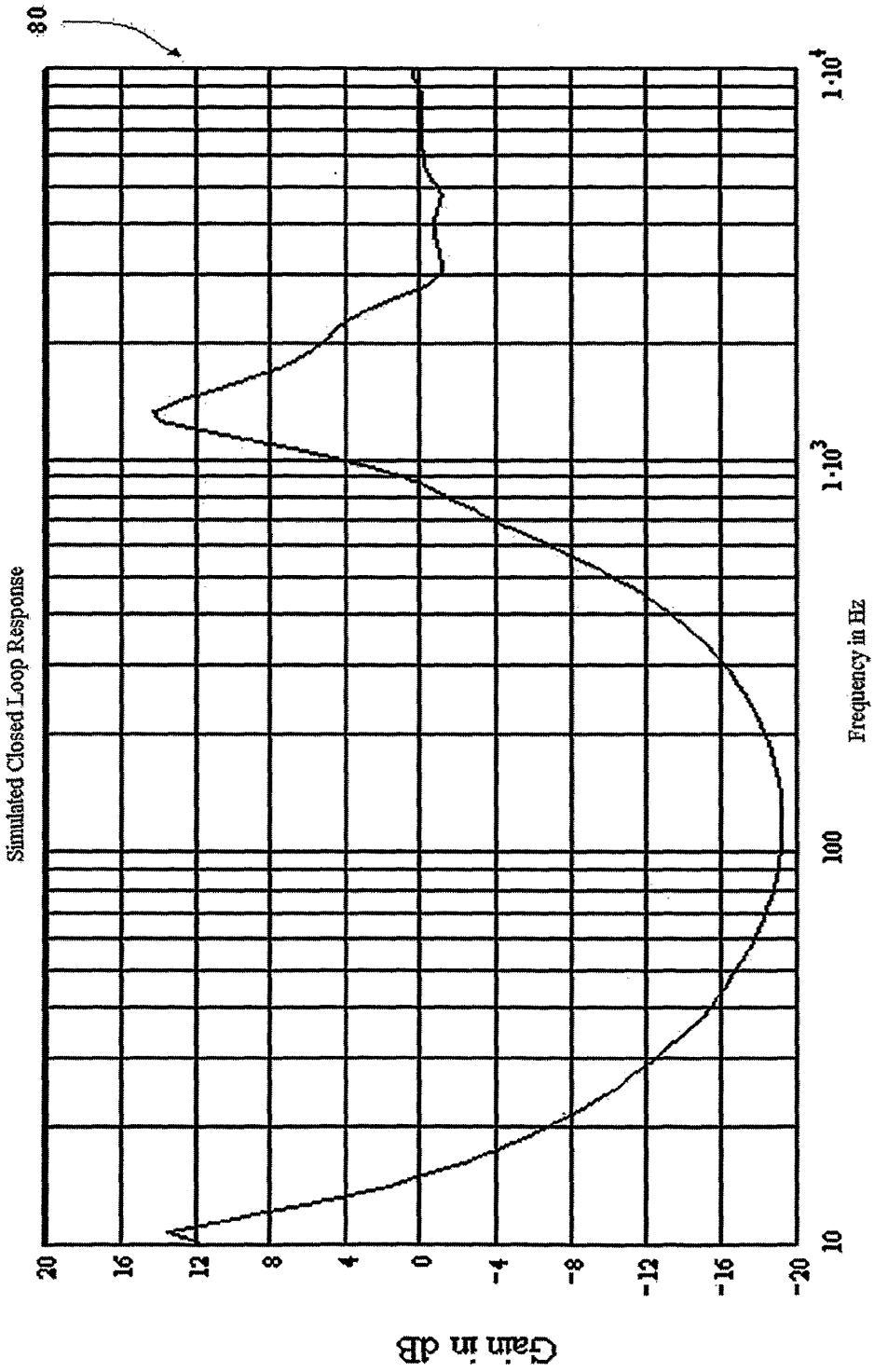

80
Simulated Closed Loop Response
Gain in dB
20
16
12
8
4
0
-4
-8
-12
-16
-20
10
100
1·10^3
1·10^4
Frequency in Hz

Fig. 8

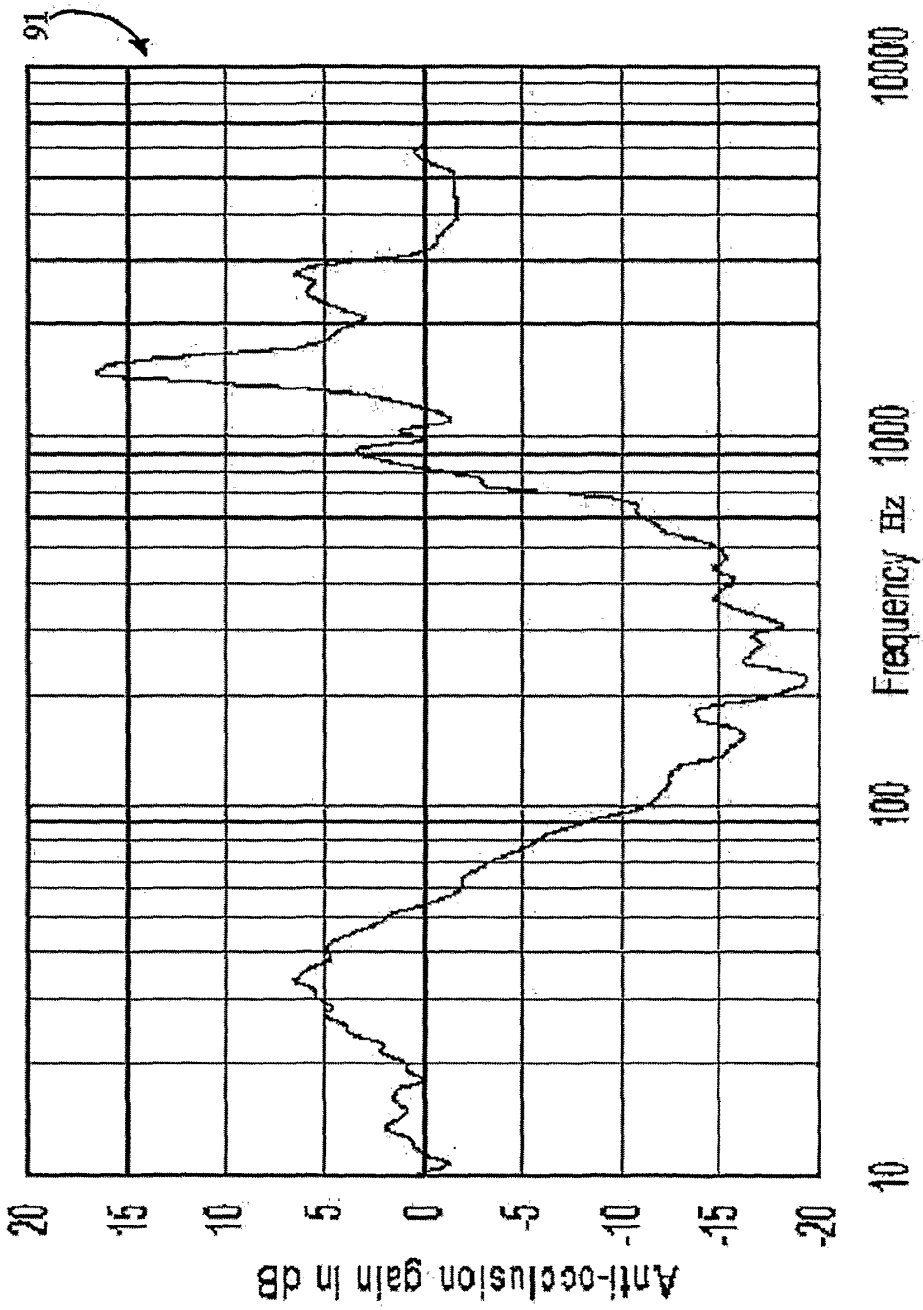

91
Anti-occlusion gain in dB
20
15
10
5
0
-5
-10
-15
-20
10
100
1000
10000
Frequency Hz

Fig. 9

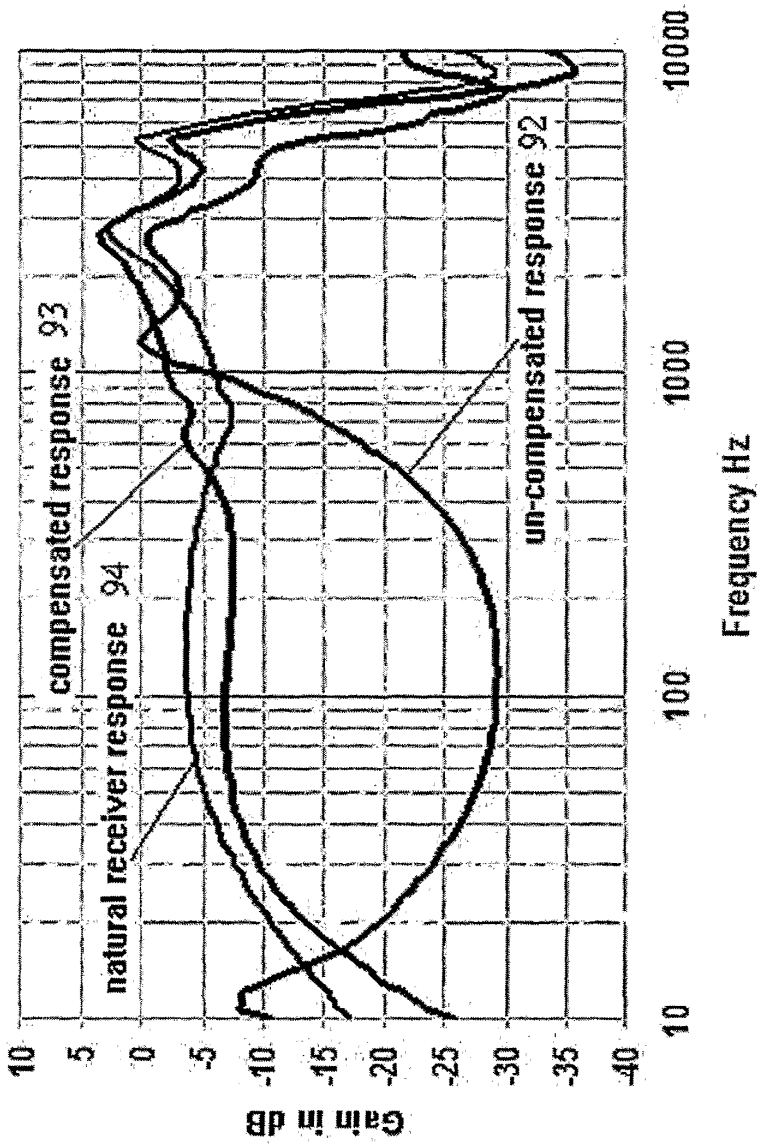

Gain in dB
10
5
0
-5
-10
-15
-20
-25
-30
-35
-40
10
100
1000
10000
Frequency Hz
compensated response 93
natural receiver response 94
un-compensated response 92

Fig. 10

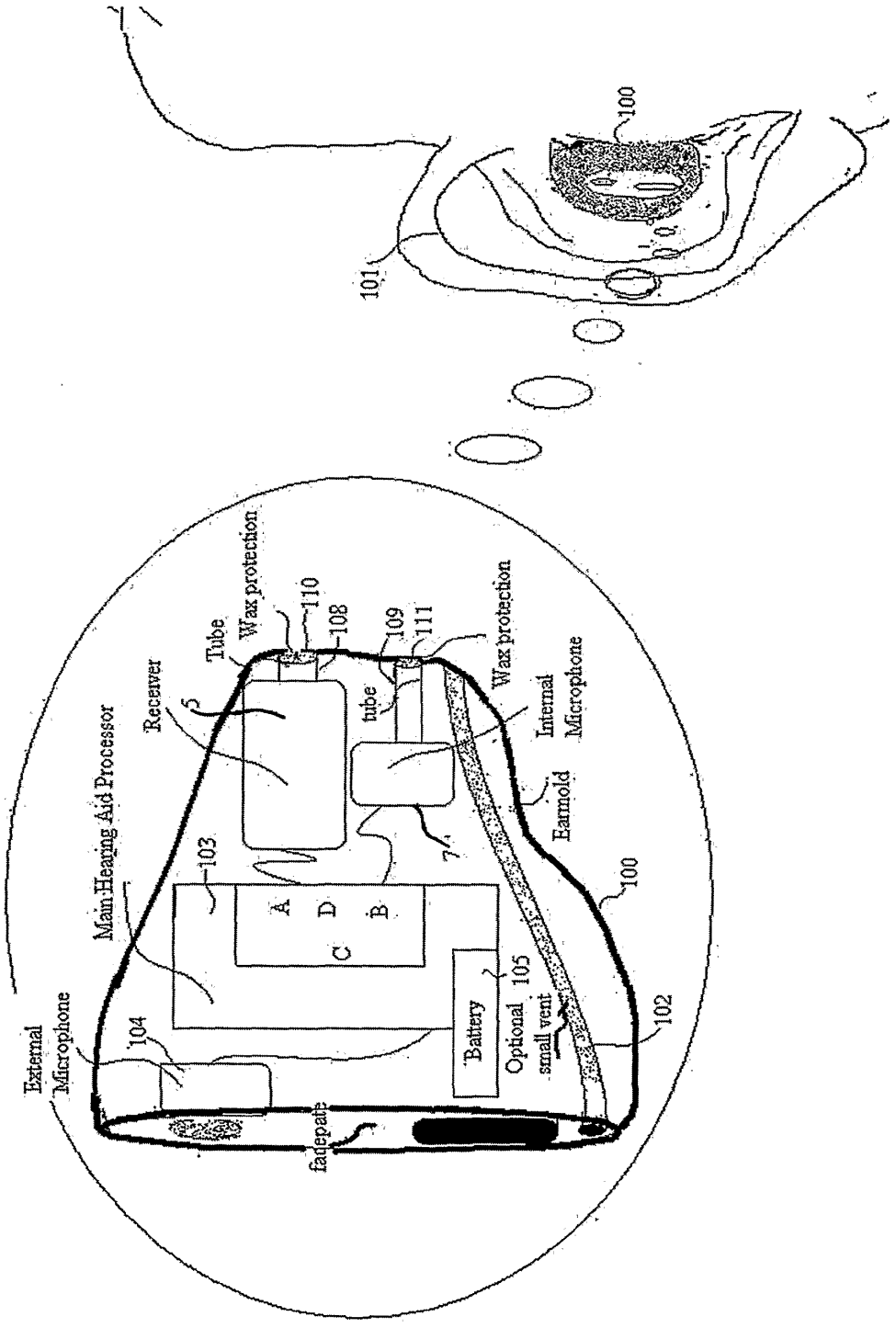

100
101
Tube
Wax protection
110
108
109
111
Wax protection
Receiver
5
tube
Internal Microphone
Earmold
Main Hearing Aid Processor
103
7
100
A
D
B
C
Battery
105
Optional small vent
102
External Microphone
104
fadepath

Fig. 11

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

A. CLASSIFICATION OF SUBJECT MATTER

Int. Cl. [7]: H04R 29/00; A61B 5/00

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)

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)
WPAT: hearing aid, occlusion, gain, frequency range, sound level, signal strength and similar terms.
USPTO: hearing aid, occlusion, reduce

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 6687377 B2 (VOIX et al.) 3 February 2004 Col. 3 line 33-Col. 6 line 22	1-24
A	WO 2002/085063 A2 (WIDEX A/S et al.) 24 October 2002 Page 4 line 14-Page 10 line 32	1-24
A	US 6766031 B1 (WESTERMANN) 20 July 2004 Col. 2 line 18-Col. 5 line 28	1-24
A	CA 2381516 A1 (GENNUM CORPORATION, CA) 12 October 2002 Page 1 line 20-Page 18 line 2	1-24

[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 application or patent 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
3 November 2005	9 NOV 2005
Name and mailing address of the ISA/AU AUSTRALIAN PATENT OFFICE PO BOX 200, WODEN ACT 2606, AUSTRALIA E-mail address: pct@ipaustralia.gov.au Facsimile No. (02) 6285 3929	Authorized officer **JUZER KHANBHAI** Telephone No : (02) 6283 2176

Form PCT/ISA/210 (second sheet) (January 2004)

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5195139 A (GAUTHIER) 16 March 1993 Col. 1 line 39-Col. 3 line 22	1-24

Form PCT/ISA/210 (continuation of second sheet) (January 2004)

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

This Annex lists the known "A" publication level patent family members relating to the patent documents cited in the above-mentioned international search report. The Australian Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

Patent Document Cited in Search Report		Patent Family Member					
US	6687377	AU	20394/02	CA	2477010	CN	1483298
		EP	1356255	US	2002076057	WO	0250499
		ZA	200305523				
WO	2002/085063	CA	2442538	EP	1250027	US	2002150269
US	6766031	AU	69189/98	CA	2286037	DK	42297
		EP	0980641	WO	9847318		
CA	2381516	N/A					
US	5195139	AU	20280/92	EP	0584259	WO	9221218

Due to data integration issues this family listing may not include 10 digit Australian applications filed since May 2001.

END OF ANNEX

UNITED STATES
PATENT AND TRADEMARK OFFICE

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Alexandria, VA 22313 - 1450
www.uspto.gov

ELECTRONIC PAYMENT RECEIPT

APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/182,570	08/24/2023 10:32:10 PM ET	11472-36-3_CONT-16838277

Title of Invention

ACOUSTIC SEALING ANALYSIS SYSTEM

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	11665493
CONFIRMATION #	5671	FILED BY	Peter Chiabotti
PATENT CENTER #	62677233	AUTHORIZED BY	-
CUSTOMER #	150139	FILING DATE	02/23/2021
CORRESPONDENCE ADDRESS	-	FIRST NAMED INVENTOR	John Usher

Payment Information

PAYMENT METHOD	PAYMENT TRANSACTION ID	PAYMENT AUTHORIZED BY
DA / 500951	E20238NM33278500	Peter Chiabotti

FEE CODE	DESCRIPTION	ITEM PRICE($)	QUANTITY	ITEM TOTAL($)
2820	REQUEST FOR CONTINUED EXAMINATION (RCE)- 2ND AND SUBSEQUENT REQUEST (SEE 37 CFR 1.114)	800.00	1	800.00
			TOTAL AMOUNT:	$800.00

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 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.

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 519 625 B1

(12) **EUROPEAN PATENT SPECIFICATION**

(45) Date of publication and mention of the grant of the patent:
12.05.2010 Bulletin 2010/19

(51) Int Cl.:
H04R 25/00 *(2006.01)*

(21) Application number: **04255520.1**

(22) Date of filing: **13.09.2004**

(54) **External ear canal voice detection**

Sprachdetektion im Gehörgang

Détection de la voix dans le conduit auditif externe

(84) Designated Contracting States:
AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IT LI LU MC NL PL PT RO SE SI SK TR

(30) Priority: **11.09.2003 US 660454**

(43) Date of publication of application:
30.03.2005 Bulletin 2005/13

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(56) References cited:
WO-A-98/45937
WO-A-03/073790
WO-A-2005/004534
WO-A-2005/125269

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).

EP 1 519 625 B1

Printed by Jouve, 75001 PARIS (FR)

Description

[0001] This document generally relates to ear-level communication systems and particularly, but not by way of limitation, to such systems communicating with remote devices via full duplex audio.

[0002] Ear-level communication systems are used for hearing impaired individuals to use a communication device such as a cellular phone or other devices transmitting audio signals. They also allow private, hand-free use of such a communication device by individuals with normal hearing.

[0003] An ear-level communication system generally includes tethered or wireless headsets with a microphone boom or a microphone within a cord of the headset to detect voice of the person wearing the headsets. An alternative way to detect voice is via bone connection. Examples include a "clam" shell device that couples to the helix and a custom carmold with a vibration detector on the outer shell. Voice is picked up by detecting bone vibration caused by, and representative of, the voice of the person wearing the device.

[0004] Bi-directional communication has been achieved by a simplex system including a device in the ear canal that both transmits sound to and picks up sound within the ear canal. The device is tethered to a push-to-talk (PIT) system such as a walkie talkie. It is not full duplex and therefore requires interaction by the user in order to switch from listen mode to talk mode.

[0005] Voice operated exchange (VOX) is used to conserve energy when voice is not being detected or transmitted. For example, a headset system includes two ear pieces - for full duplex communication with one earpiece for transmitting and the other for receiving voice - and uses ear canal voice detection to trigger a VOX circuit in a control unit wired to the headset. The system requires a headset with two headphones and wires connecting the headphone to a control unit such as a device clipped on belt. In another example, a full duplex system includes a single headphone to transmit and receiving voice, again with a VOX circuit in a control unit wired to the headphone.

[0006] Such ear-level communication systems are likely to be worn by a person for long periods of time. Users with hearing loss may use such a system on a regular basis, both as a conventional hearing aid allowing communication directly with a person and as a hearing aid allowing communicating through another device. Users with normal hearing may wear such a system to be communicatively connected to another person or facility without the need to hold a device at any time. For appearance or secrecy reasons, minimal visibility is generally desirable. Wired connection between a headset or earpiece and a controller makes the system easily visible and is cumbersome to users who need to communicate while being physically active.

[0007] Thus, there is a need for an ear-level communication system that is not easily visible. A need related to miniaturization is the longevity of the system between battery replacements or recharges.

[0008] WO 98/45937A discloses a communications apparatus comprising ear units, a control unit and a radio transceiver.

[0009] Ear level full duplex audio communication systems each include one or two ear attachments devices, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, that wirelessly communicates to a remote device. Such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. Such a system also functions as an ear piece wirelessly extended from the remote device that allows an individual to privately communicate with or through the remote device without the need of holding the device.

[0010] WO 03/073790A discloses a system adapted for use in at least one of a first ear having a first ear canal and a second ear having a second ear canal comprising: a first ear-level device configured to be one of an in-the-ear device and a behind-the-ear device, the first ear-level device including: a first microphone adapted for use in one of the first ear canal and the second ear canal, and adapted to receive an occluded sound within the one of the first ear canal and the second ear canal; and a wireless receiver, connected to a first processor, to receive a first wireless signal representing a remote sound; wherein the first processor, is connected to the first microphone, and is adapted to process the occluded sound and the remote sound; the first ear level device further including a speaker adapted for use in the one of the first ear canal and the second ear canal and adapted to transmit the processed remote sound to the one of the first ear canal and the second ear canal; a wireless transmitter, connected to the first processor; a first earmold housing at least the first microphone and the speaker, at least a portion of the first earmold configured for placement in the one of the first ear canal and the second ear canal; and a voice operated exchange (VOX) circuit connected to the first processor.

[0011] In the present invention, the wireless transmitter is adapted to receive the processed occluded sound from the first processor and transmit a second wireless signal representing the occluded sound; and the VOX circuit includes an occluded sound gating module adapted to gate the occluded sound on, and to gate the remote sound off, when the occluded sound is detected while the remote sound is not detected, and a remote sound gating module adapted to gate the remote sound on, and to gate the occluded sound off, when the incoming remote sound is detected while the occluded sound is not detected.

[0012] In a preferred embodiment, a system includes an ITE module and a BTE module attached to the ITE module. The ITE module includes a microphone for use in an ear canal. The microphone receives an occluded sound from within the ear canal. The BTE module includes a processor and a wireless transmitter. The processor processes the occluded sound. The wireless trans-

IPR2024-01015 Page 00466

mitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0013] In a preferred embodiment, a system includes a first ear-level device and a second ear-level device for use in two opposite ears. The first ear-level device includes a microphone, a first processor, and a wireless transmitter. The microphone is for use in one ear canal to receive an occluded sound from within that ear canal. The first processor converts the occluded sound to an electrical signal. The wireless transmitter receives the electrical signal and transmits an outgoing wireless signal representing the occluded sound. The second ear-level device includes a wireless receiver, a second processor, and a speaker. The wireless receiver receives an incoming wireless signal representing a remote sound. The second processor converts the incoming wireless signal to the remote sound. The speaker is for use in the opposite ear canal to transmit the remote sound to that ear canal.

[0014] The present invention also relates to a method comprising: detecting an occluded sound using a first microphone adapted for use in an ear canal; detecting an incoming radio signal representing a remote sound from a remote device; starting a voice operated exchange (VOX) housed within an ear-level communication device when at least one of the occluded sound and the incoming radio signal is detected if the VOX is not already on, the ear-level communication device being one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device; gating the occluded sound on, and gating the remote sound off, when the occluded sound is detected while the incoming radio signal is not detected; and gating the remote sound on, and gating the occluded sound off, when the incoming radio signal is detected while the occluded sound is not detected.

[0015] Preferred embodiments of the present invention will now be described by way of example only and with reference to the accompanying drawings, in which:

FIG. 1 is an illustration of an embodiment of an ear-level communication system and potions of an environment in which it is used. This ear-level communication system uses a single ear-level device.

FIG. 2 is an illustration of an embodiment of another ear-level communication system and portions of an environment in which it is used. This ear-level communication system uses a pair of ear-level devices.

FIG. 3A is an illustration of one exemplary exterior configuration of an in-the-ear (ITE) device used as the single ear-level device of FIG 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 3B is an illustration of one exemplary exterior configuration of an in-the-canal (ITC) device used as the single ear-level device in FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 3C is an illustration of one exemplary exterior configuration of a completely-in-the-canal (CIC) device used as the single-ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device of FIG. 1.

FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single car-level device of FIG. 1.

FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the pair of ear-level devices of FIG. 1B.

FIG. 6 is an illustration of one exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 7 is a block diagram showing one embodiment of the circuit of the ear-level device shown in FIG. 6 used as the single ear-level device of FIG. 1.

FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as the pair of ear-level devices of FIG. 2.

FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device.

FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication between the single ear-level device and a remote device.

[0016] The following embodiments are described in sufficient detail to enable those skilled in the art to practice the invention, and it is to be understood that the embodiments may be combined, or that other embodiments may be utilized and that structural, logical and electrical changes may be made without departing from the scope of the present invention, as defined by the appended claims.

[0017] It should be noted that references to "an", "one", or "various" embodiments in this disclosure are not necessarily to the same embodiment, and such references contemplate more than one embodiment.

[0018] This document discusses, among other things, ear-level full duplex audio communication systems for communicating with remote devices. These systems each include one or two ear-level devices each worn in or about an ear, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, wirelessly coupled to a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. When used as a hearing aid, such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. When being used as an ear piece wirelessly extended from the remote device,

IPR2024-01015 Page 00467

such system allows an individual with normal hearing to privately communicate with or through the remote device without the need of holding the device or wearing any device wired to the remote device.

[0019] In this document, "voice" includes to the sound of speech made by a person, and sound generally includes an audible signal that includes voice and other signals that can be heard and/or detected by an audio detector. Unless particularly indicated, for example, when voice is distinguished from sound by speech recognition, "voice" and "sound" are used interchangeably throughout thus document because they are not practically separated. For example, detection of a sound refers to the same action as detection of a voice because, while the purpose is to detect the voice, other sounds are picked up as well.

[0020] FIG. 1 is an illustration of an embodiment of an ear-level communication system 100 and portions of an environment in which it is used. System 100 includes an ear-level device 110 attached to an ear 101A. Bar-level device 110 communicates with a remote device 190 through a wireless telemetry link 195. The other ear 101B is free of device attachment. Ear-level device 110 is a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of ear-level device 110 is voice activated. That is, ear-level device 110 is activated only when it detects a need to transmit or receive a sound.

[0021] Ear-level device 110 is a full duplex audio device that allows two-way simultaneous conversation between ear 101A and remote device 190. In one embodiment, when being worn by a person having ear 101A, ear-level device 110 picks up sound from the ear canal of ear 101A and delivers sound to the same ear canal. It detects the occluded sound from the ear canal when the person speaks and transmits to the same ear canal a sound received from remote device 190 and/or a sound picked up from the environment surrounding the person. In one embodiment, as illustrated in FIG. 1, ear-level device 110 is configured for right ear attachment. In an alternative embodiment, ear-level device 110 is configured for left ear attachment.

[0022] FIG. 2 is an illustration of an embodiment of an ear-level communication system 200 and portions of an environment in which it is used. System 200 includes a pair of ear-level devices 210A, attached to an ear 101 A, and 210B, attached to ear 101B. Ear-level devices 210A and 210B communicate with a remote device 290 through a wireless telemetry link 295. Ear-level devices 210A and 210B are each a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of each of ear-level devices 210A and 210B is voice activated. That is, ear-level devices 210A and 210B are each activated only when a need to transmit or receive a sound is detected by the device itself.

[0023] In one embodiment, ear-level device 210A is an audio transmitter that picks up sound from the ear canal of ear 101A, and ear-level device 210B is an audio receiver that delivers sound to the same ear canal. Ear-level device 210A is activated when it detects a sound from the ear canal. Ear-level device 210B is activated when it detects a signal from remote device 290. When both are being worn by a person, ear-level devices 210A and 210B supports full duplex audio allowing two-way simultaneous conversation between the person and remote device 290. In one embodiment, as illustrated in FIG. 2, ear-level device 210A is configured for right ear attachment, and ear-level device 210B is configured for left ear attachment. In an alternative embodiment, ear-level device 210A is configured for left ear attachment, and ear-level device 210B is configured for right ear attachment.

[0024] FIGS. 3A, 3B, and 3C are illustrations of exemplary exterior configuration of ear-level device 110 or one of ear-level devices 210A and 210B. For the purpose of this document, only a device attached to a right ear is shown for each exemplary exterior configuration. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a device configured for right ear attachment or a device configured for left ear attachment. Generally, a device configured for right car attachment has an exterior configuration that is substantially symmetrical to that of a device of the same type configured for left ear attachment. In system 100, a single ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of car-level device 210A and 210B is a device configured for right ear attachment, while the other is a device configured for left ear attachment.

[0025] An occluded sound 305, an ambient sound 306, and a remote sound 307 are illustrated in each of FIGS. 3A, 3B, and 3C. Remote sound 307 is represented by a wireless radio signal transmitted to ear-level device 110. In this document, "occluded sound" includes occluded sound 305, "ambient sound" includes ambient sound 306, and "remote sound" includes remote sound 307.

[0026] FIG. 3A is an illustration of an exemplary exterior configuration of an in-the-ear (ITE) device 310A used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3A, ITE device 310A is attached to an ear 101 with a pinna 302 and an ear canal 303. ITE device 310A includes a portion inserted into ear canal 303 and another portion fit into the cavity formed by pinna 302.

[0027] FIG. 3B is an illustration of an exemplary exterior configuration of an in-the-canal (ITC) device 310B used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3B, ITC device 310B is attached to ear 101 with pinna 302 and ear canal 303. ITC device 310B is a form of ITE device that also includes a portion inserted in ear canal 303 and another portion fit into the cavity formed by pinna 302, but the latter portion is smaller than that of ITE device 310A. Its overall

IPR2024-01015 Page 00468

size is therefore generally smaller than that of ITE deviee 310A.

[0028] FIG. 3C is an illustration of an exemplary exterior configuration of a completely-in-the-canal (CIC) device 310C used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3C, CIC device 310C is attached to ear 101 with pinna 302 and ear canal 303. CIC device 310C is a form of ITC device but includes only a portion that is completed inserted into ear canal 303. Its overall size is therefore generally smaller than that of ITC device 310B.

[0029] In one embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a custom earmold. In another embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0030] The choice of a device type, such as from the types shown in FIGS. 3A, 3B, and 3C, depends on the need of individual users. For example, ITE device 310A has a relatively large size to accommodate a relatively large amplifier to serve the need of individuals suffering severe hearing loss. The relatively large size can also be utilized to house a relatively large size battery. On the other hand, CIC device 310C is almost invisible, while the room for the circuit and the battery is very limited. Generally, the selection of the device type is a compromise among considerations such as the user's degree of hearing loss (if applicable), ear canal size, desire of invisibility, desired longevity between battery replacements or recharges, and cost.

[0031] FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device 110. The circuit is housed in an earmold configured for use as one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0032] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an car canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not "isolated" when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, end wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110.

[0033] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected

IPR2024-01015 Page 00469

by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440.

[0034] For the purpose of description, each of the gating modules is "on" when sound is "gated on" and "off' when the sound is "gated off" A sound is "gated on" when it is allowed to pass and be processed, and is "gated off' when it is blanked or substantially attenuated. Blanking refers to a substantially complete blockage of a sound, or in other words, that a detected sound is practically ignored by processor 460 such that it does not cause any echo or ringing that is audible by an ear. Substantial attenuation refers to an attenuation after which the attenuated sound does not cause any echo or tinging having an intolerable intensity.

[0035] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay.

[0036] Occluded sound gating module 441 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected occluded sound with a predetermined threshold for the occluded sound. When the amplitude of the detected occluded sound exceeds the threshold for the occluded sound, and when controller 431 permits, occluded sound gating module 441 is turned on. When the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off. In one embodiment, after the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off after a predetermined delay.

[0037] In one embodiment, remote sound gating module 451 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a remote sound with a predetermined threshold for the remote sound. When the amplitude of the detected remote sound exceeds the threshold for the remote sound, and when controller 431 permits, remote sound gating module 451 is turned on. When the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off. In one embodiment, after the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off after a predetermined delay.

[0038] In another embodiment, remote sound gating module 451 includes a switch or attenuator driven by the detection of the incoming wireless signal representing the remote sound. Remote sound gating module 451 is turned on upon detection of the incoming wireless signal when controller 431 permits, and turned off when the incoming wireless signal is no longer detected. In one embodiment, remote sound gating module 451 is turned off after the incoming wireless signal is absent for a predetermined period.

[0039] Controller 431 coordinates the on/off states of all the gating modules so the sounds are processed in an orderly fashion. Even if a sound is detected, the gating module corresponding to that sound is turned on only when controller 431 permits. In one embodiment, assuming that sounds A and B are gated by gating modules A and B, respectively, if both sounds A and B are detected, whichever sound is detected earlier is gated on. That is, if sound A exceeds its threshold first, gating module A is turned on and remain on until sound A drops below threshold A. Then, gating module B is turned only if sound B exceeds its threshold after gating module A has been turn off.

[0040] In one embodiment, controller 431 coordinates the on/off states of all the gating modules according to predetermined or programmed gating coordination rules. According to one exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows only the transmission of the remote sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. According to another exemplary rule, ambient sound gating module 421 is on, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows the transmission of both the remote sound and the ambient sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. This also prevents the ambient sound from being transmitted to remote device 190. According to yet another exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is on, and remote sound gating module 451 is off. When the person wearing ear-level device 110 speaks, both external microphone 420 and ear canal microphone 440 detect the voice. The voice as detected by ear-level

IPR2024-01015 Page 00470

device 110 is first transmitted to remote device 190. Without the gating modules (or when ambient sound gating module 421 and occluded sound gating module 441 are both on), the same voice as detected by external microphone 420 is transmitted to ear canal 303, and therefore again detected by ear canal microphone 440, and again be transmitted to remote device 190. This rule allows only the transmission of the occluded sound to remote device 190, and prevents the same sound from being transmitted to remote device 190 twice. Other rules are applied as a person skilled in the art should see fit based on an understanding after reading this entire document.

[0041] FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as each of ear-level devices 210A and 210B. The circuit of each of ear-level devices 210A and 210B is housed in an earmold configured for use as of one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0042] Ear-level device 210A is an audio transmitter that includes an ear canal microphone 540, a processor 560A, a wireless transmitter 550A, an antenna 555A, a transmitter VOX circuit 530A, and a battery 580A. Ear canal microphone 540 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 210A. Processor 560A converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 290. In one embodiment, processor 560A includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560A includes a speech recognition module used to enhance the sound transmitted from ear-level device 210A. Wireless transmitter 550A and antenna 555A form a telemetry interface supporting signal transmission from ear-level device 210A to remote device 290. Wireless transmitter 550A receives the electrical signal representing the sound picked up from the ear canal from processor 560A and transmits an outgoing wireless signal representing the same sound to remote device 290. VOX circuit 530A activates a major portion of ear-level device 210A only when sound is detected by ear canal microphone 540. In one embodiment, VOX circuit 530A includes a voice-controlled switch connecting between battery 580A and the portions of ear-level device 210A that are voice activated. The switch is driven by the sound detected by ear canal microphone 540. It is turned on when sound detected by ear canal microphone 540 exceeds a predetermined threshold. In one embodiment, VOX circuit 530A activates processor 560A when the sound is detected. In one further embodiment, VOX circuit 530A also activates wireless transmitter 550A when the sound is detected. Battery 580A supplies the power needed for the operation of ear-level device 210A. In one embodiment, battery 580A is a rechargeable battery that can be recharged without being taken out from ear-level device 210A.

[0043] Ear-level device 210B is an audio receiver that includes an external microphone 520, an ear canal speaker 570, a processor 560B, a wireless receiver 550B, an antenna 555B, a receiver VOX circuit 530B, and a battery 580B. External microphone 520 picks up ambient sound such that the person wearing ear-level devices 210A and 210B is not "isolated" when communicating to remote device 290. It allows that person to talk to a nearby person directly, without the need to take off ear-level devices 210A and/or 210B. Bar canal speaker 570 transmits sound received from remote device 290 and/or external microphone 520 to the same ear canal. Processor 560B converts the signals received from remote device 290 and/or external microphone 520 to a sound audible to the person wearing ear-level device 210B. In one embodiment, processor 560B includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560B includes a speech recognition module used to enhance the audio signals received by ear-level device 210B. Wireless receiver 550B and antenna 555B form a telemetry interface supporting signal transmission from remote device 290 to ear-level device 210B. Wireless receiver 550B receives an incoming wireless signal representing a remote sound from remote device 290 and transmits an electrical signal representing the remote sound to processor 560B. VOX circuit 530B activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 520 and wireless receiver 550B. In one embodiment, VOX circuit 530B includes a voice-controlled switch connecting between battery 580B and the portions of ear-level device 210B that are voice activated. The switch is driven by the sound detected by at least one of external microphone 520 and wireless receiver 550B. It is turned on when sound detected by external microphone 520 exceeds a predetermined threshold and/or when wireless transceiver 550B receives the incoming wireless signal representing the remote sound. In one embodiment, VOX circuit 530B activates processor 560B when the sound is detected. In one further embodiment, VOX circuit 530B also activates ear-level speaker 570 when the sound is detected. Battery 580B supplies the power needed for the operation of ear-level device 210B. In one embodiment, battery 580B is a rechargeable battery that can be recharged without being taken out from ear-level device 210B.

[0044] System 200 accommodates a larger overall circuit size without necessarily increasing the size of each ITE device. While two separate ear-level devices are required, because system 200 includes separate audio transmitter (ear-level device 210A) and receiver (ear-level device 210B) disposed in the ear canals of opposite ears, ear canal speaker 570 and ear canal microphone 540 do not form a loop causing echo and ringing.

IPR2024-01015 Page 00471

[0045] FIG. 6 is an illustration of an exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device 610 used as ear-level device 110 or one of ear-level devices 210A and 210B. BTE device 610 is attached to ear 101 with pinna 302 and ear canal 303. In one embodiment, BTE device 610 is used when none of ITE device 310A, PTC device 310B, and CIC device 310C provides sufficient space accommodating the required circuit and battery sizes. For the purpose of this document, only a device attached to a right ear is shown for each exterior structure. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a BTE device configured for right ear attachment or a BTE device configured for left ear attachment. Generally, a BTE device configured for right ear attachment has a substantially symmetrical appearance to a device of the same type configured for left ear attachment In system 100, ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of ear-level device 210A and 210B is a device structured for right ear attachment, while the other is a device structured for left ear attachment.

[0046] BTE device 610 includes a behind-the-earmodule (BTE module) 611 and an in-the-ear module (ITE module) 612. ITE module 612 includes a custom or modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0047] FIG. 7 is a block diagram showing one embodiment of the circuit of ear-level device shown in FIG. 6 used as ear-level device 110. The exterior configuration of ear-level device 110 is of the type illustrated as BTE device 610.

[0048] Ear-level device 110 configured as BTE device 610 differs from ear-level device 110 configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 7 illustrates one embodiment in which BTE module 611 includes external microphone 420, processor 460, wireless transceiver 450, antenna 455, VOX circuit 430, and battery 480, and ITE module 612 includes ear canal microphone 440 and ear canal speaker 470. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, behind-ear portion 611 includes only battery 480 to maximize the duration of use of ear-level device 110 between battery replacements or recharges.

[0049] FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as each of ear-level devices 210A and 210B. The exterior configuration of each of ear-level devices 210A and 210B is of the type illustrated as BTE device 610.

[0050] Ear-level devices 210A/210B configured as BTE device 610 differ from ear-level devices 210A/210B configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 8 illustrates one embodiment of ear-level device 210A including a BTE module 611A and an ITE module 612A, and ear-level device 210B including a BTE module 611B and an ITE module 612B. BTE module 611A includes processor 560A, wireless transmitter 550A, antenna 555A, transmitter VOX circuit 530A, and battery 580A. ITE module 612A includes ear canal microphone 540. Behind-the-ear portion 611B includes external microphone 520, processor 560B, wireless receiver 550B, antenna 555B, receiver VOX circuit 530B, and battery 580B. ITE module 612B includes ear canal speaker 570. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, BTE modules 611A and 611B include only batteries 580A and 580B, respectively, to maximize the duration of use of ear-level devices 210A and 210B, respectively, between battery replacements or recharges.

[0051] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication using system 100 or system 200. The method provides for private, simultaneous two-way communication between a person and a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. In this embodiment. VOX is used for power management. A VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 or system 200 that are voice activated. The VOX is "on" when the voice-activated portions of the circuit are activated, i.e., when the power switch remains on. It is "started" by turning on the power switch, and "stopped" by turning off the power switch.

[0052] Steps 910-918 illustrate the process of detecting a sound from an ear canal and transmitting it to a remote device. A microphone disposed in an ear canal detects an occluded sound within the ear canal at 910. If the occluded sound is detected at 911, the VOX of the system is started at 912 if it is not already on. The occluded sound is enhanced at 914 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice of a speech, and the noise is any sound other than the voice of the speech. The occluded sound is then converted to an outgoing radio signal at 916. The outgoing radio signal representing the occluded sound is transmitted to the remote device at 918.

[0053] Steps 920-928 illustrate the process of detecting a sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects an incoming radio signal representing a remote sound from the remote device at 920. If the incoming radio signal is detected at 921, the VOX of the system is started at 922 if it is not already on. The incoming radio signal is converted to a first incoming audio signal (the remote sound) at 924. The first incoming audio signal is enhanced at 926 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice from the remote device, and the noise is any sound other than the voice.

IPR2024-01015 Page 00472

A speaker disposed in the ear canal transmits the first incoming audio signal representing the sound from the remote device to the ear canal at 928.

[0054] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928.

[0055] If no signal (sound or radio signal) is detected at 911, 921, and 931, the VOX is stopped at 940. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times. In one embodiment, the VOX is stopped after a predetermined period of time to avoid frequent activation/deactivation during a conversation.

[0056] Systems 100 and 200 are each a full duplex system capable of performing steps 910-918, 920-928, and 930-938 are performed in parallel and simultaneously. In an embodiment in which system 100 performs the method, ear-level device 110 performs steps 910-918, 924-928, and 930-938. If a signal is detected at 911, 921, or 931, VOX 430 activates at least portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively. If no signal is detected at 911, 921, or 931, VOX 430 deactivates at least the portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively.

[0057] In an embodiment in which system 200 performs the method, ear-level device 210A performs steps 910-918, and ear-level device 210B performs steps 920-928 and 930-930. If a signal is detected at 911, transmitter VOX 530A activates at least portions of device 210A performing steps 914-918. If a signal is detected at 921 or 931, receiver VOX 530B activates at least portions of device 210B performing steps 924-928 or 934-938, respectively. If no signal is detected at 911, transmitter VOX 530A deactivates at least the portions of device 210A performing steps 914-918. If no signal is detected at 921 or 931, receiver VOX 530B deactivates at least the portions of device 210B performing steps 924-928 or 934-938, respectively.

[0058] FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication using system 100. In this embodiment, in addition to power management, the VOX is also used for gating, i.e., blanking or attenuating, sounds. The VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 that are voice activated and gating modules capable of blanking or attenuating the occluded sound, the remote sound, and the ambient sound. Examples of such gating modules include occluded sound gating module 441, remote sound gating module 451, and ambient sound gating module 421, designated for the occluded sound, the remote sound, and the ambient sound, respectively.

[0059] Steps 1010-1019 illustrate the process of detecting the occluded sound from the ear canal and transmitting it to the remote device. The microphone disposed in the ear canal detects the occluded sound within the ear canal at 1010. If the occluded sound is detected at 1011, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1012 if they are not already powered. The VOX determines whether to start gating the occluded sound on at 1013. In one embodiment, the occluded sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. Examples of gating coordination rules are described above with respect to FIG. 4B. If the occluded sound is gated on at 1013, it is processed at 1014 by following steps that are identical or similar to steps 914-916-918. The occluded sound is gated off at 1019 if it has been gated on but is no longer detected. In one embodiment, the occluded sound is gated off after it is not detected for a predetermined period.

[0060] Steps 1020-1028 illustrate the process of detecting the remote sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects the incoming radio signal representing the remote sound from the remote device at 1020. If the remote sound is detected at 1021, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1022 if they are not already powered. The VOX determines whether to start gating the remote sound on at 1023. In one embodiment, the remote sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. In another embodiment, the remote sound is gated on when the incoming radio signal is detected and when the effective gating coordination rules permits. If the remote sound is gated on at 1023, it is processed at 1024 by following steps that are identical or similar to steps 924-926-928. The remote sound is gated off at 1029 if it has been gated on but is no longer detected. In one embodiment, the remote sound is gated off after it is not detected for a predetermined period.

[0061] Steps 1030-1038 illustrate the process of detecting the ambient sound and transmit it to the ear canal. A microphone outside the ear canal detects the ambient sound at 1030. If the ambient sound is detected at 1031, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1032 if they are not already powered. The VOX determines whether to start gating the ambient sound on at 1033. In one embodiment, the ambient sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. If the

IPR2024-01015 Page 00473

ambient sound is gated on at 1033, it is processed at 1034 by following steps that are identical or similar to steps 934-936-938. The ambient sound is gated off at 1039 if it has been gated on but is no longer detected. In one embodiment, the ambient sound is gated off after it is not detected for a predetermined period.

[0062] If no signal (sound or radio signal) is detected at 1011,1021, and 1031, the VOX is stopped at 1040. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times, In one embodiment, the VOX is stopped after a predetermined period of time to avoid Sequent activation/deactivation during a conversation.

[0063] It is to be understood that the above detailed description is intended to be illustrative, and not restrictive. Other embodiments will be apparent to those of skill in the art upon reading and understanding the above description. For example, two ear-level devices 110, or a pair of devices each being similar to ear-level device 110, can be used to deliver a stereo sound. The scope of the invention should, therefore, be determined with reference to the appended claims.

Claims

1. A system adapted for use in at least one of a first ear (101A) having a first ear canal (303) and a second ear (101B) having a second ear canal (303) comprising:

   a first ear-level device (110, 210A) configured to be one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device, the first ear-level device including:

   a first microphone (440) adapted for use in one of the first ear canal and the second ear canal, and adapted to receive an occluded sound (305) within the one of the first ear canal (303) and the second ear canal (303); and
   a wireless receiver, connected to a first processor (460), to receive a first wireless signal representing a remote sound;

   wherein the first processor (460) is connected to the first microphone (440) and is adapted to process the occluded sound (305) and the remote sound; the first ear level device further including:

   a speaker adapted for use in the one of the first ear canal (303) and the second ear canal (303) and adapted to transmit the processed remote sound to the one of the first ear canal (303) and the second ear canal (303);
   a wireless transmitter, connected to the first processor (460) and adapted to receive the processed occluded sound from the first processor (460) and transmit a second wireless signal representing the occluded sound;
   a first earmold housing at least the first microphone (440) and the speaker (470), at least a portion of the first earmold configured for placement in the one of the first ear canal (303) and the second ear canal (303); and
   a first voice operated exchange (VOX) circuit (430) connected to the first processor (460);
   the first VOX circuit (430) including:

   an occluded sound gating module (441) adapted to gate the occluded sound on, and to gate the remote sound off, when the occluded sound is detected while the remote sound is not detected; and
   a remote sound gating module (451) adapted to gate the remote sound on, and to gate the occluded sound off, when the incoming remote sound is detected while the occluded sound is not detected.

2. A system according to claim 1, wherein the first earmold houses the first microphone (440), the first processor (460) and the wireless transmitter, and wherein the first earmold is configured for use as the ITE device (310A).

3. A system according to claim 1, wherein the first ear level device (110,210A) comprises an ITE module including the first microphone (440) and a BTE module attached to the ITE module, the BTE module including the first processor and the wireless transmitter.

4. A system according to claim 1, wherein the first microphone (440) and the speaker (470) are implemented as one common device receiving the occluded sound and transmitting the processed remote sound.

5. A system according to claim 4, further comprising a second microphone (420) connected to the first processor (460), the second microphone (420) adapted to receive an ambient sound from outside the one of the first ear canal (303) and the second ear canal (303), and wherein:

   the first processor (460) further processes the ambient sound; and
   the speaker (470) is configured to transmit the processed remote and ambient sounds to the one of the first ear canal (303) and the second ear canal (303).

6. A system according to claim 5, wherein the first VOX

IPR2024-01015 Page 00474

circuit (430) comprises an ambient sound gating module (421), coupled to the first microphone (420), to blank or substantially attenuate the ambient sound.

7. A system according to any of the preceding claims, wherein the first processor (460) comprises a speech recognition module.

8. A system according to any of claims 4 - 7, wherein the first earmold houses the first microphone (440), the first processor (460), the wireless transmitter, the wireless receiver, and the speaker (470), and wherein the first earmold is configured for use as the ITE device.

9. A system according to claim 8, wherein the first earmold is configured for use as an in-the-canal (ITC) device (310B).

10. A system according to claim 9, wherein the first earmold is configured for use as a completely-in-the-canal (CIC) device (310C).

11. A system according to any of claims 4 - 7, wherein the first ear level device (110,210A) comprises an ITE module including the first microphone (440) and the speaker (470) and a BTE module attached to the ITE module and including the second microphone (420), the first processor (460), the wireless transmitter and the wireless receiver.

12. A system according to claim 11, wherein the BTE module comprises the first voice operated exchange (VOX) circuit (430).

13. A system according to claim 11, wherein the BTE module comprises a rechargeable battery (480).

14. A system according to any of claims 7 - 13, further comprising a remote device (190, 290) communicatively coupled to the wireless transmitter and the wireless receiver via a telemetry link (195, 295) providing for simultaneous bidirectional communication between the first ear-level device (110, 210A) and the remote device (190, 290).

15. A system according to claim 14, wherein the remote device (190, 290) comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

16. A system according to claim 1, wherein the first microphone (440) is adapted for use in the first ear canal (303) and receives the occluded sound within the first ear canal (303), wherein the at least the portion of the first earmold is configured for placement in the first ear canal (303), and further comprising a second ear-level device (210B) including:

a wireless receiver to receive a first wireless signal representing a remote sound;
a second processor (260), connected to the wireless receiver, to convert the first wireless signal to the remote sound;
a speaker (470) connected to the second processor, the speaker adapted for use in the second ear canal (303) to transmit the remote sound to the second ear canal (303); and
a second earmold housing at least the speaker, at least a portion of the second earmold configured for placement in the second ear canal (303).

17. A system according to claim 16, wherein the second ear-level device (210B) further comprises a second microphone (420) connected to the second processor (460), the second microphone (420) adapted to receive an ambient sound from outside the second ear canal (303), and wherein:

the second processor (460) further processes the ambient sound; and
the speaker (470) is configured to transmit the remote sound and the processed ambient sound to the second ear canal (303).

18. A system according to any of claims 16 and 17, wherein the first ear-level device (110,210A) comprises the first voice operated exchange (VOX) circuit (430) connected to the first processor (460), and the second ear-level device (210B) further comprises a second VOX circuit (430) connected to the second processor.

19. A system according to any of claims 16 - 18, wherein the first processor and the second processor each comprise a speech recognition module.

20. A system according to any of claims 16 - 19, wherein the first and second ear-level devices are each the BTE device.

21. A system according to any of claims 16 - 19, wherein the first and second ear-level devices are each the ITE device.

22. A system according to claim 21, wherein the first and second ear-level devices are each an in-the-canal (ITC) device (310B).

23. A system according to claim 22, wherein the first and second ear-level devices are each a completely-in-the-canal (CIC) device (310C).

24. A system according to any of claims 16 - 23, further

IPR2024-01015 Page 00475

comprising a remote device (190,290) communicatively coupled to the first and second ear-level devices.

25. A system according to claim 24, wherein the remote device (190, 290) comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

26. A method, comprising:

detecting an occluded sound using a first microphone adapted for use in an ear canal (303);
detecting an incoming radio signal representing a remote sound from a remote device (190, 290);
starting a voice operated exchange (VOX) housed within an ear-level communication device when at least one of the occluded sound and the incoming radio signal is detected if the VOX is not already on, the ear level communication device being one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device;
gating the occluded sound on, and gating the remote sound off, when the occluded sound is detected while the incoming radio signal is not detected; and
gating the remote sound on, and gating the occluded sound off, when the incoming radio signal is detected while the occluded sound is not detected.

27. A method according to claim 26, wherein detecting the occluded sound comprises comparing an amplitude of the occluded sound to an occluded sound threshold, wherein the occluded sound is detected when the amplitude of the occluded sound exceeds the occluded sound threshold.

28. A method according to any of claims 26 and 27, wherein detecting the incoming radio signal comprises comparing an amplitude of the remote sound to a remote sound threshold, wherein the incoming radio signal is detected when the amplitude of the remote sound exceeds the remote sound threshold.

29. A method according to any of claims 26 - 28, further comprising detecting an ambient sound using a second microphone, and wherein starting the VOX comprises starting the VOX when at least one of the occluded sound, the incoming radio signal, and the ambient sound is detected if the VOX is not already on.

30. A method according to claim 29, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound off, when the incoming radio signal is detected while the occluded sound is not detected.

31. A method according to claim 29, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound on, when the incoming radio signal is detected while the occluded sound is not detected.

32. A method according to claim 29, further comprising gating the occluded sound on, gating the remote sound off, and gating the ambient sound off, when the occluded sound is detected while the incoming radio signal is not detected.

Patentansprüche

1. System, das zur Verwendung in einem ersten Ohr (101A) mit einem ersten Gehörgang (303) und/oder einem zweiten Ohr (101B) mit einem zweiten Gehörgang (303) eingerichtet ist, mit:

einer ersten ohrebenen- bzw. ohrseitigen Vorrichtung (110, 210A), die dafür eingerichtet ist, eine In-Ohr-Vorrichtung (ITE) oder eine Hinter-Ohr-Vorrichtung (BTE) zu sein, wobei die erste ohrseitige Vorrichtung beinhaltet:

ein erstes Mikrofon (440), das zur Verwendung in dem ersten Gehörgang oder dem zweiten Gehörgang eingerichtet ist, und das dafür eingerichtet ist, einen verdeckten Ton (305) in jeweils dem ersten Gehörgang (303) oder dem zweiten Gehörgang (303) zu empfangen; und
einen Drahtlosempfänger, der mit einem ersten Prozessor (460) verbunden ist, um ein erstes Drahtlossignal zu empfangen, welches einen Fernton darstellt;

wobei der erste Prozessor (460) mit dem ersten Mikrofon (440) verbunden und dafür eingerichtet ist, den verdeckten Ton (305) und den Fernton zu verarbeiten;
wobei die erste ohrseitige Vorrichtung ferner beinhaltet:

einen Lautsprecher, der zur Verwendung in dem jeweiligen ersten Gehörgang (303) oder dem zweiten Gehörgang (303) eingerichtet ist, und der dafür eingerichtet ist, den verarbeiteten Fernton an den jeweiligen ersten Gehörgang (303) oder den zweiten Gehörgang (303) zu übertragen;
einen Drahtlossender, der mit dem ersten Prozessor (460) verbunden und dafür eingerichtet ist, den verarbeiteten verdeckten Ton von dem ersten Prozessor (460) zu empfangen und ein zweites Drahtlossignal, das den verdeckten Ton darstellt, zu senden;

IPR2024-01015 Page 00476

ein erstes Ohrformgehäuse wenigstens bei dem ersten Mikrofon (440) und dem Lautsprecher (470), wobei wenigstens ein Teil der ersten Ohrform dafür eingerichtet ist, in dem ersten Gehörgang (303) oder dem zweiten Gehörgang (303) platziert zu werden; und
eine erste stimmbetriebene Austauschschaltung (VOX) (430), die mit dem ersten Prozessor (460) verbunden ist;

wobei die erste VOX-Schaltung (430) beinhaltet:

ein Ausblendmodul (441) für einen verdeckten Ton, das dafür eingerichtet ist, den verdeckten Ton einzublenden und den Fernton auszublenden, wenn der verdeckte Ton bestimmt wird, während der Fernton nicht bestimmt wird; und
ein Ausblendmodul (441) für einen Fernton, das dafür eingerichtet ist, den Fernton einzublenden und den verdeckten Ton auszublenden, wenn der eingehende Fernton bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

2. System nach Anspruch 1, bei dem eine erste Ohrform das erste Mikrofon (440), den ersten Prozessor (460) und den Drahtlossender einschließt, und bei dem die erste Ohrform zur Verwendung als ITE-Vorrichtung (310) eingerichtet ist.

3. System nach Anspruch 1, bei dem die erste ohrseitige Vorrichtung (110, 210A) ein ITE-Modul, das das erste Mikrofon (440) beinhaltet, und ein zweites BTE-Modul, das an das ITE-Modul angebracht ist, beinhaltet, wobei das ITE-Modul den ersten Prozessor und den Drahtlossender beinhaltet.

4. System nach Anspruch 1, bei dem das erste Mikrofon (440) und der Lautsprecher (470) als eine gemeinsame Vorrichtung implementiert sind, die den verdeckten Ton empfängt und den verarbeiteten Fernton sendet.

5. System nach Anspruch 4, ferner mit einem zweiten Mikrofon (420), das mit dem ersten Prozessor (460) verbunden ist, wobei das zweite Mikrofon (420) dafür eingerichtet ist, einen ümgebungston von außerhalb das jeweiligen ersten Gehörgangs (303) oder des zweiten Gehörgangs (303) zu empfangen, und wobei:

der erste Prozessor (460) den Umgebungston ferner bzw. weiterverarbeitet; und
der Lautsprecher (470) dafür eingerichtet ist, den verarbeiteten Fern- und Umgebungston an den jeweiligen ersten Gehörgang (303) und zweiten Gehörgang (303) zu senden.

6. System nach Anspruch 5, bei dem die erste VOX-Schaltung (430) ein Umgebungstonausblendmodul (421) aufweist, das mit dem ersten Mikrofon (420) verbunden ist, um den Umgebungston auf Null zu setzen oder ihn wesentlich zu dämpfen.

7. System nach einem der vorstehenden Ansprüche, bei dem der erste Prozessor (460) ein Spracherkennungsmodul aufweist.

8. System nach einem der Ansprüche 4 bis 7, bei dem die erste Ohrform, das erste Mikrofon (440), den ersten Prozessor (460), den Drahtlossender, den Drahtlosempfänger und den Lautsprecher (470) einschließt, und bei dem die erste Ohrform dafür eingerichtet ist, als die ITE-Vorrichtung verwendet zu werden.

9. System nach Anspruch 8, bei dem die erste Ohrform zur Verwendung als In-Kanal-Vorrichtung (ITC) (310B) verwendet zu werden.

10. System nach Anspruch 9, bei dem die erste Ohrform zur Verwendung als vollständig-In-Kanal-Vorrichtung (CIC) (310C) verwendet zu werden.

11. System nach einem der Ansprüche 4 bis 7, bei dem die erste ohrseitige Vorrichtung (110, 210A) ein ITE-Modul aufweist, das das erste Mikrofon (440) und den Lautsprecher (470) und ein BTE-Modul, das an das ITE-Modul angebracht ist, und das das zweite Mikrofon (420), den ersten Prozessor (460), den drahtlos Sender und den drahtlos Empfänger beinhaltet, beinhaltet.

12. System nach Anspruch 11, bei dem das BTE-Modul die erste sprachbetriebene Austauschschaltung (VOX) (340) aufweist.

13. System nach Anspruch 11, bei dem das BTE-Modul eine widerauflädbare Batterie (480) aufweist.

14. System nach einem der Ansprüche 7 bis 13, das ferner eine entfernte Vorrichtung (190, 290) aufweist, die mit dem Drahtlossender und dem Drahtlosempfänger über eine Telemetrieverbindung (195, 295) in Kommunikationsverbindung steht, um eine gleichzeitige bi-direktionale Kommunikation zwischen der ersten ohrseitigen Vorrichtung (110, 210A) und der entfernten Vorrichtung (190, 290) bereitzustellen.

15. System nach Anspruch 14, bei dem die entfernte Vorrichtung (190, 290) wenigstens einen Computer, einen Personal Digital Assistant (PDA), ein Mobiltelefon, ein Walkie Talkie und/oder einen Sprachübersetzer aufweist.

16. System nach Anspruch 1, bei dem das erste Mikro-

IPR2024-01015 Page 00477

fon (440) dafür eingerichtet ist, in dem ersten Gehörgang (303) verwendet zu werden, und dieses den verdeckten Ton in dem ersten Gehörgang (303) empfängt, wobei der wenigstens eine Abschnitt der ersten Ohrform zur Anordnung in dem ersten Gehörgang (303) eingerichtet ist, und ferner mit einer zweiten ohrseitigen Vorrichtung (210B), die beinhaltet:

einen Drahtlosempfänger, um ein erstes Drahtlossignal zu empfangen, das einen Fernton darstellt;
einen zweiten Prozessor (260), der mit dem Drahtlosempfänger verbunden ist, um das erste Drahtlossignal in den Fernton umzuwandeln;
einen Lautsprecher (470), der mit dem zweiten Prozessor verbunden ist, wobei der Lautsprecher zur Verwendung in dem zweiten Gehörgang (303) eingerichtet ist, um den Fernton an den zweiten Gehörgang (303) zu übertragen; und
einer zweiten Ohrform, die wenigstens den Lautsprecher einschließt, wobei wenigstens ein Teil der zweiten Ohrform zur Anordnung in dem zweiten Gehörgang (303) eingerichtet ist.

17. System nach Anspruch 16, bei dem die zweite ohrseitige Vorrichtung (210B) ferner ein zweites Mikrofon (420) aufweist, das mit dem zweiten Prozessor (460) verbunden ist, wobei das zweite Mikrofon (420) dafür eingerichtet ist, einen Umgebungston von außerhalb des zweiten Gehörgangs (303) zu empfangen, und bei dem:

der zweite Prozessor (460) den Umgebungston weiterverarbeitet; und
der Lautsprecher (470) dafür eingerichtet ist, den Fernton und den verarbeiteten Umgebungston an den zweiten Gehörgang (303) zu senden.

18. System nach einem der Ansprüche 16 oder 17, bei dem die erste ohrseitige Vorrichtung (110, 210A) die erste sprachbetriebene Austauschschaltung (VOX) (430) mit dem ersten Prozessor (460) verbunden aufweist, und wobei die zweite ohrseitige Vorrichtung (210B) ferner eine zweite VOX-Schaltung (430) aufweist, die mit dem zweiten Prozessor verbunden ist.

19. System nach einem der Ansprüche 16 bis 18, bei dem der erste Prozessor und der zweite Prozessor jeweils ein Spracherkennungsmodul aufweisen.

20. System nach einem der Anspruch 16 bis 19, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils die BTE-vorrichtungen darstellen.

21. System nach einem der Ansprüche 16 bis 19, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils die ITE-Vorrichtungen darstellen.

22. System nach Anspruch 21, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils eine In-Gehörgangs-Vorrichtung (ITC) (310B) darstellen.

23. System nach Anspruch 22, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils Vollständig-In-Gehörgangs-Vorrichtungen (CIC) (310C) darstellen.

24. System nach einem der Ansprüche 16 bis 23, das ferner eine entfernte Vorrichtung (190, 290) aufweist, die kommunikativ mit den ersten und zweiten ohrseitigen Vorrichtungen verbunden ist.

25. System nach Anspruch 24, bei dem die entfernte Vorrichtung (190, 290) wenigstens einen Computer, einen Personal Digital Assistant (PDA), ein Mobiltelefon, ein Walkie Talkie und/oder einen Sprachübersetzer aufweist.

26. Verfahren, das beinhaltet:

Bestimmen eines verdeckten Tons unter Verwendung eines ersten Mikrofons, das zur Verwendung in einem Gehörgang (303) eingerichtet ist;
Bestimmen eines eingehenden Funksignals, das einen Fernton von einer entfernten Vorrichtung (190, 290) darstellt;
Starten einer sprachbetriebenen Austauscheinrichtung (VOX), die in einer ohrseitigen Kommunikationsvorrichtung eingeschlossen ist, wenn der verdeckte Ton und/oder das eingehende Funksignal bestimmt wird, wenn die VOX nicht bereits an ist, wobei die ohrseitige Kommunikationsvorrichtung eine In-Ohr-Vorrichtung (ITE) oder eine Hinter-Ohr-Vorrichtung (BTE) darstellt;
Einblenden des verdeckten Tons und Ausblenden des Ferntons, wenn der verdeckte Ton bestimmt wird, während das eingehende Funksignal nicht bestimmt wird; und
Einblenden des Ferntons und Ausblenden des verdeckten Tons, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

27. Verfahren nach Anspruch 26, bei dem das Bestimmen des verdeckten Tons, das Vergleichen einer Amplitude des verdeckten Tons mit einem Schwellwert für einen verdeckten Ton beinhaltet, wobei der verdeckte Ton bestimmt wird, wenn die Amplitude des verdeckten Tones den Schwellwert für den verdeckten Ton überschreitet.

IPR2024-01015 Page 00478

28. Verfahren nach einem der Ansprüche 26 oder 27, bei dem das Bestimmen des eingehenden Funksignals das Vergleichen einer Amplitude des Ferntons mit einem Ferntonschwellwert beinhaltet, wobei das eingehende Funksignal bestimmt wird, wenn die Amplitude des Ferntons den Ferntonschwellwert überschreitet.

29. Verfahren nach einem der Ansprüche 26 bis 28, das ferner das Bestimmen eines Umgebungstons unter Verwendung eines zweiten Mikrofons beinhaltet, und wobei das Starten der VOX das Starten der VOX beinhaltet, wenn wenigstens der verdeckte Ton, das eingehende Funksignal und/oder der Umgebungston bestimmt wird, während die VOX noch nicht bereits an ist.

30. Verfahren nach Anspruch 29, das ferner das Ausblenden des verdeckten Tons, das Einblenden des Ferntons und das Ausblenden des Umgebungstons beinhaltet, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

31. Verfahren nach Anspruch 29, das ferner das Ausblenden des verdeckten Tons, das Einblenden des Ferntons und das Einblenden des Umgebungstons beinhaltet, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

32. Verfahren nach Anspruch 29, das ferner das Einblenden des verdeckten Tones, das Ausblenden des Ferntons und das Ausblenden des Umgebungstons beinhaltet, wenn der verdeckte Ton bestimmt wird, während das eingehende Funksignal nicht bestimmt wird.

Revendications

1. Un système adapté en vue d'une utilisation dans au moins une parmi une première oreille (101A) ayant un premier conduit auditif (303) et une seconde oreille (101B) ayant un second conduit auditif (303), comprenant:

   un premier dispositif d'oreille (110, 210A) configuré pour être un parmi un dispositif placé à l'intérieur de l'oreille (ITE) et un dispositif contour d'oreille (BTE), le premier dispositif d'oreille comprenant :

   un premier microphone (440) adapté en vue d'une utilisation dans un parmi le premier conduit auditif et le second conduit auditif, et adapté pour recevoir un son occlus (305) à l'intérieur de celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; et
   un récepteur sans fil, connecté à un premier processeur (460), pour recevoir un premier signal sans fil représentant un son distant ;

   dans lequel le premier processeur (460) est connecté au premier microphone (440) et est adapté pour traiter le son occlus (305) et le son distant; le premier dispositif d'oreille comprenant en outre :

   un haut-parleur adapté en vue d'une utilisation dans celui parmi le premier conduit auditif (303) et le second conduit auditif (303) et adapté pour transmettre le son distant traité à celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; un émetteur sans fil, connecté au premier processeur (460) et adapté pour recevoir le son occlus traité du premier processeur (460) et émettre un second signal sans fil représentant le son occlus ;
   un premier embout auriculaire logeant au moins le premier microphone (440) et le haut-parleur (470), au moins une portion du premier embout auriculaire étant configurée en vue d'être placée dans celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; et
   un premier circuit d'échange à commande vocale (VOX) (430) connecté au premier processeur (460) ;
   le premier circuit VOX (430) comprenant :

   un module de contrôle de son occlus (441) adapté pour laisser passer le son occlus et pour bloquer le son distant lorsque le son occlus est détecté alors que le son distant n'est pas détecté ; et
   un module de contrôle de son distant (451) adapté pour laisser passer le son distant et pour bloquer le son occlus lorsque le son distant entrant est détecté alors que le son occlus n'est pas détecté.

2. Système selon la revendication 1, dans lequel le premier embout auriculaire loge le premier microphone (440), le premier processeur et l'émetteur sans fil, et dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif ITE (310 A).

3. Système selon la revendications 1, dans lequel le premier dispositif d'oreille (110, 210A) comprend un module ITE comprenant le premier microphone (440) et un module BTE fixé au module ITE, le module BTE comprenant le premier processeur et l'émetteur sans fil.

4. Système selon la revendication 1, dans lequel le premier microphone (440) et le haut-parleur (470) sont

IPR2024-01015 Page 00479

mis en oeuvre sous la forme d'un dispositif commun recevant le son occlus et émettant le son distant traité.

5. Système selon la revendication 4, comprenant en outre un second microphone (420) connecté au premier processeur (460), le second microphone (420) étant adapté pour recevoir un son ambiant de l'extérieur de celui parmi le premier conduit auditif (303) et le second conduit auditif (303), et dans lequel :

le premier processeur (460) traite en outre le son ambiant ; et
le haut-parleur (470) est configuré pour transmettre les sons distant et ambiant traités à celui parmi le premier conduit auditif (303) et le second conduit auditif (303).

6. Système selon la revendication 5, dans lequel le premier circuit VOX (430) comprend un module de contrôle de son ambiant (421), couplé au premier microphone (420), pour occulter ou atténuer sensiblement le son ambiant.

7. Système selon l'une quelconque des revendications précédentes, dans lequel le premier processeur (460) comprend un module de reconnaissance vocale.

8. Système selon l'une quelconque des revendications 4 à 7, dans lequel le premier embout auriculaire loge le premier microphone (440), le premier processeur (460), l'émetteur sans fil, le récepteur sans fil, et le haut-parleur (470), et dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif ITE.

9. Système selon la revendication 8, dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif intra-conduit (ITC) (310B).

10. Système selon la revendication 9, dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif complètement intra-conduit (CIC) (310C).

11. Système selon l'une quelconque des revendications 4 à 7, dans lequel le premier dispositif d'oreille (110, 210A) comprend un module ITE comprenant le premier microphone (440) et le haut-parleur (470) et un module BTE fixé au module ITE et comprenant le second microphone (420), le premier processeur (460), l'émetteur sans fil et le récepteur sans fil.

12. Système selon la revendication 11, dans lequel le module BTE comprend le premier circuit d'échange à commande vocale (VOX) (430).

13. Système selon la revendication 11, dans lequel le module BTE comprend une batterie rechargeable (480).

14. Système selon l'une quelconque des revendications 7 à 13, comprenant en outre un dispositif distant (190, 290) couplé en communication à l'émetteur sans fil et au récepteur sans fil via une liaison de télémesure (195, 295) assurant une communication bidirectionnelle simultanée entre le premier dispositif d'oreille (110, 210A) et le dispositif distant (190, 290).

15. Système selon la revendication 14, dans lequel le dispositif distant (190, 290) comprend au moins un parmi un ordinateur, un assistant numérique personnel (PDA), un téléphone cellulaire, un talkie-walkie, ou un traducteur de langue.

16. Système selon la revendication 1, dans lequel le premier microphone (440) est adapté en vue d'une utilisation dans le premier conduit auditif (303) et reçoit le son occlus à l'intérieur du premier conduit auditif (303), ans lequel au moins la portion du premier embout auriculaire est configurée en vue d'être placée dans le premier conduit auditif (303), et comprenant en outre un second dispositif d'oreille (210B) comprenant :

un récepteur sans fil pour recevoir un premier signal sans fil représentant un son distant ;
un second processeur (260), connecté au récepteur sans fil, pour convertir le premier signal sans fil dans le son distant ;
un haut-parleur (470) connecté au second processeur, le haut-parleur étant adapté en vue d'une utilisation dans le second conduit auditif (303) pour transmettre le son distant au second conduit auditif (303) ; et
un second embout auriculaire logeant au moins le haut-parleur, au moins une portion du second embout auriculaire étant configurée en vue d'être placée dans le second conduit auditif (303).

17. Système selon la revendication 16, dans lequel le second dispositif d'oreille (210B) comprend en outre un second microphone (420) connecté au second processeur (460), le second microphone (420) étant adapté pour recevoir un son ambiant de l'extérieur du second conduit auditif (303), et dans lequel :

le second processeur (460) traite en outre le son ambiant ; et le haut-parleur (470) est configuré pour transmettre le son distant et le son ambiant traité au second conduit auditif (303).

18. Système selon l'une quelconque des revendications 16 et 17, dans lequel le premier dispositif d'oreille

IPR2024-01015 Page 00480

(110, 210) comprend le premier circuit d'échange à commande vocale (VOX) (430) connecté au premier processeur (460), et le second dispositif d'oreille (210B) comprend en outre un second circuit VOX (430) connecté au second processeur.

19. Système selon l'une quelconque des revendications 16 à 18, dans lequel le premier processeur et le second processeur comprennent chacun un module de reconnaissance vocale.

20. Système selon l'une quelconque des revendications 16 à 19, dans lequel les premier et second dispositifs d'oreille sont respectivement le dispositif BTE.

21. Système selon l'une quelconque des revendications 16 à 19, dans lequel les premier et second dispositifs d'oreille sont respectivement le dispositif ITE.

22. Système selon la revendication dans lequel les premier et second dispositifs d'oreille sont respectivement un dispositif intra-conduit (ITC) (310B).

23. Système selon la revendication 22, dans lequel les premier et second dispositifs d'oreille sont respectivement un dispositif complètement intra-conduit (CIC) (310C).

24. Système selon l'une quelconque des revendications 16 à 23, comprenant en outre un dispositif distant (190, 290) couplé en communication aux premier et second dispositifs d'oreille.

25. Système selon la revendication 24, dans lequel le dispositif distant (190, 290) comprend au moins un parmi un ordinateur, un assistant numérique personnel (PDA), un téléphone cellulaire, un talkie walkie, ou un traducteur de langue.

26. Un procédé comprenant les étapes consistant à :

détecter un son occlus au moyen d'un premier microphone adapté en vue d'une utilisation dans un conduit auditif (303) ;
détecter un signal radio entrant représentant un son distant provenant d'un dispositif distant (190, 290) ;
lancer un échange à commande vocale (VOX) logé à l'intérieur d'un dispositif de communication d'oreille lorsqu'au moins un parmi le son occlus et le signal radio entrant est détecté si le VOX n'est pas déjà en marche, le dispositif de communication d'oreille étant un parmi un dispositif placé à l'intérieur de l'oreille (ITE) et un dispositif contour d'oreille (BTE) ;
laisser passer le son occlus, et bloquer le son distant lorsque le son distant occlus est détecté alors que le signal radio entrant n'est pas détecté ; et
laisser passer le son distant, et bloquer le son occlus lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

27. Procédé selon la revendication 26, dans lequel la détection du son occlus comprend la comparaison d'une amplitude du son occlus avec un seuil de son occlus, dans lequel le son occlus est détecté lorsque l'amplitude du son occlus dépasse le seuil du son occlus.

28. Procédé selon l'une quelconque des revendications 26 et 27, dans lequel la détection du signal radio entrant comprend la comparaison d'une amplitude du son distant avec un seuil de son distant, dans lequel le signal radio entrant est détecté lorsque l'amplitude du son distant dépasse le seuil du son distant.

29. Procédé selon l'une quelconque des revendications 26 à 28, comprenant en outre la détection d'un son ambiant au moyen d'un second microphone, et dans lequel le lancement du VOX comprend le lancement du VOX lorsqu'au moins un parmi le son occlus, le signal radio entrant et le son ambiant est détecté si le VOX n'est pas déjà en marche.

30. Procédé selon la revendications 29, comprenant en outre l'étape consistant à bloquer le son occlus, laisser passer le son distant et bloquer le son ambiant lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

31. Procédé selon la revendication 29, comprenant en outre l'étape consistant à bloquer le son occlus, laisser passer le son distant et laisser passer le son ambiant lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

32. Procédé selon la revendication 29, comprenant en outre l'étape consistant à laisser passer le son occlus, bloquer le son distant, et bloquer le son ambiant lorsque le son occlus est détecté alors que le signal radio entrant n'est pas détecté.

IPR2024-01015 Page 00481

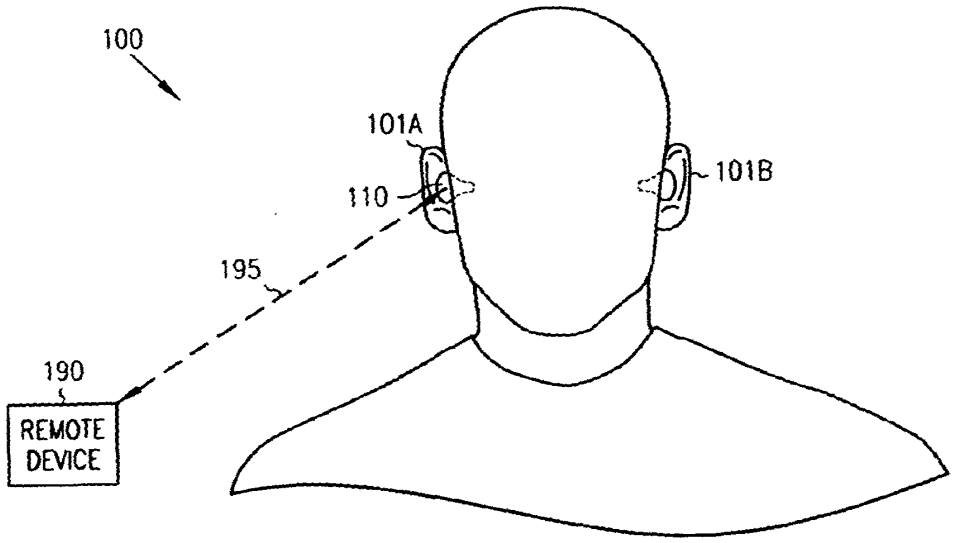

100
101A
101B
110
195
190
REMOTE DEVICE

FIG. 1

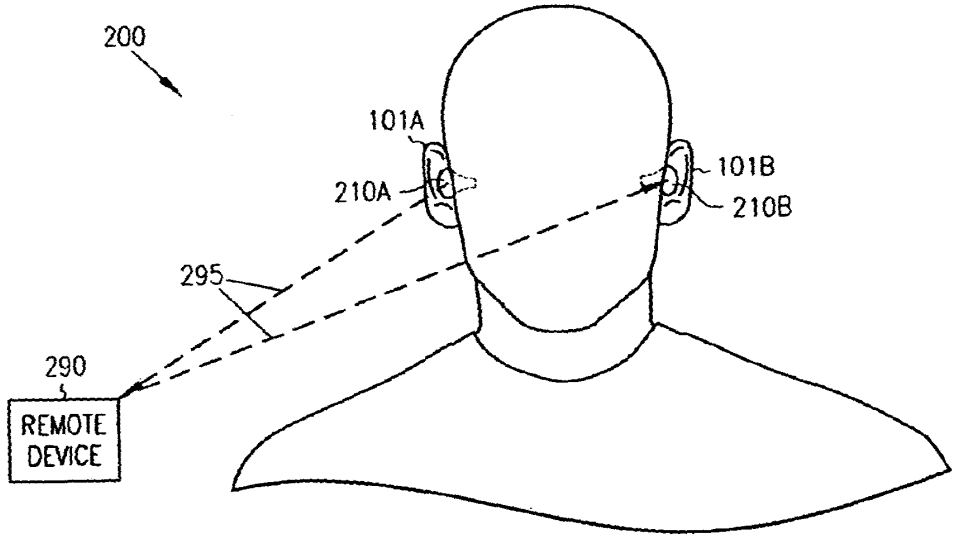

200
101A
101B
210A
210B
295
290
REMOTE DEVICE

FIG. 2

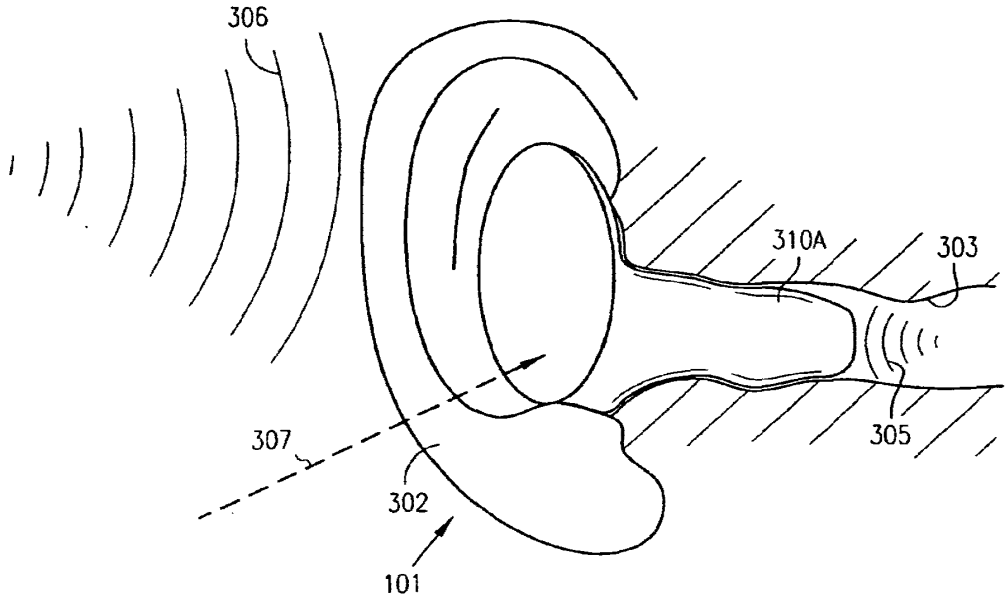


FIG. 3A

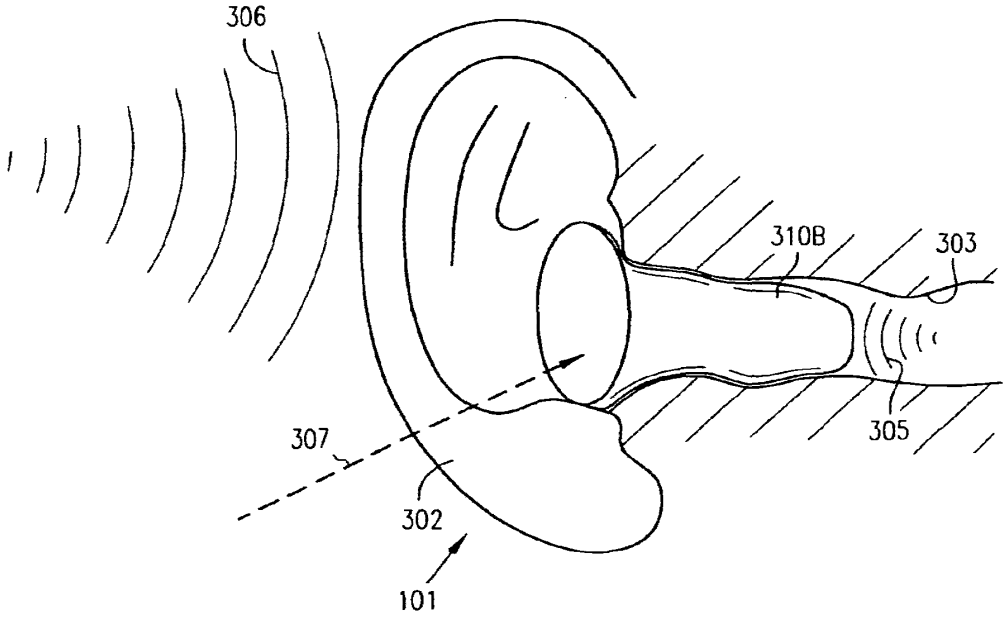


FIG. 3B

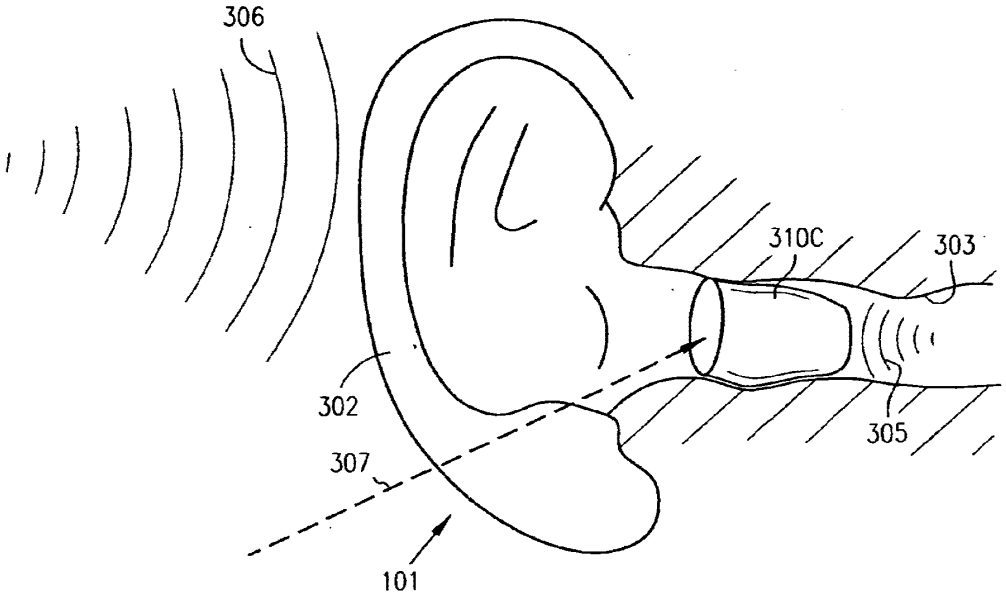


FIG. 3C

IPR2024-01015 Page 00484

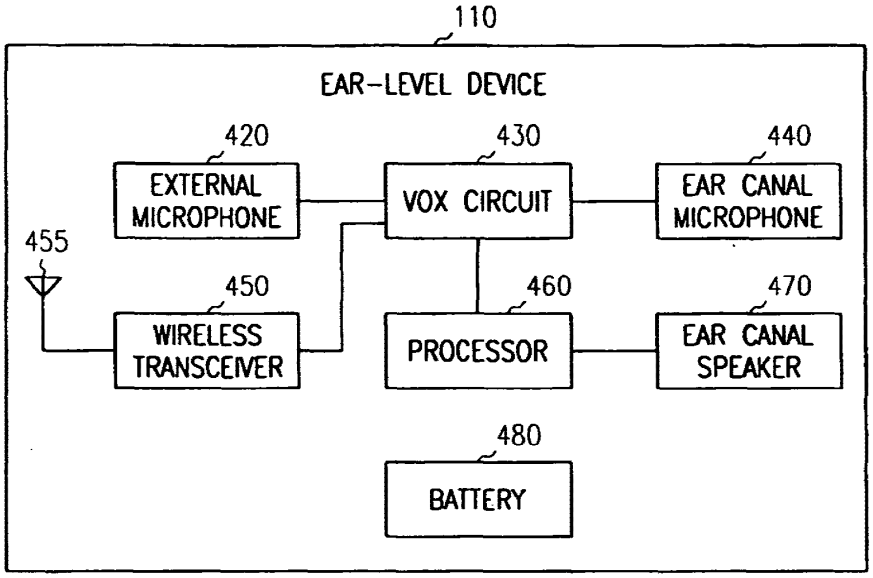

110
EAR-LEVEL DEVICE
420
EXTERNAL MICROPHONE
430
VOX CIRCUIT
440
EAR CANAL MICROPHONE
455
450
WIRELESS TRANSCEIVER
460
PROCESSOR
470
EAR CANAL SPEAKER
480
BATTERY

FIG. 4A

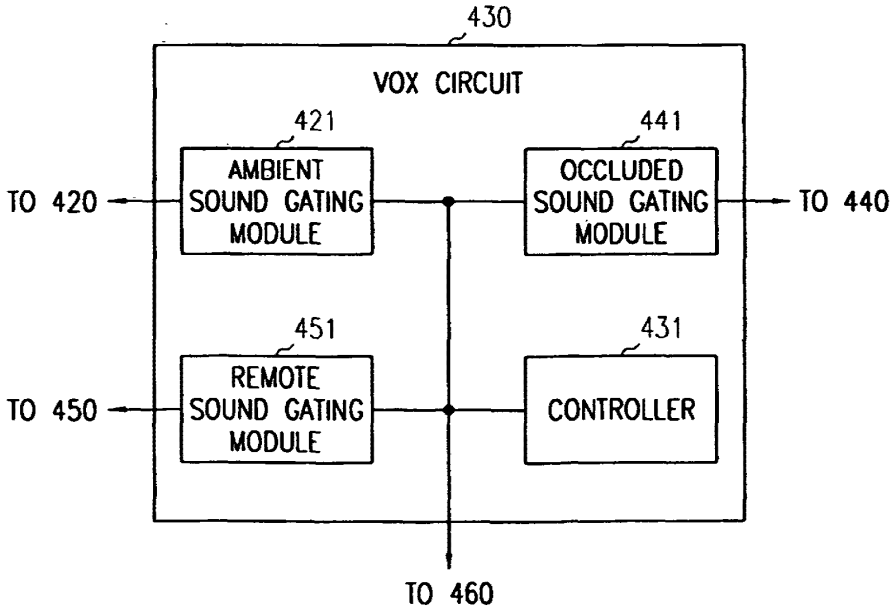

430
VOX CIRCUIT
421
AMBIENT SOUND GATING MODULE
441
OCCLUDED SOUND GATING MODULE
TO 420
TO 440
451
REMOTE SOUND GATING MODULE
431
CONTROLLER
TO 450
TO 460

FIG. 4B

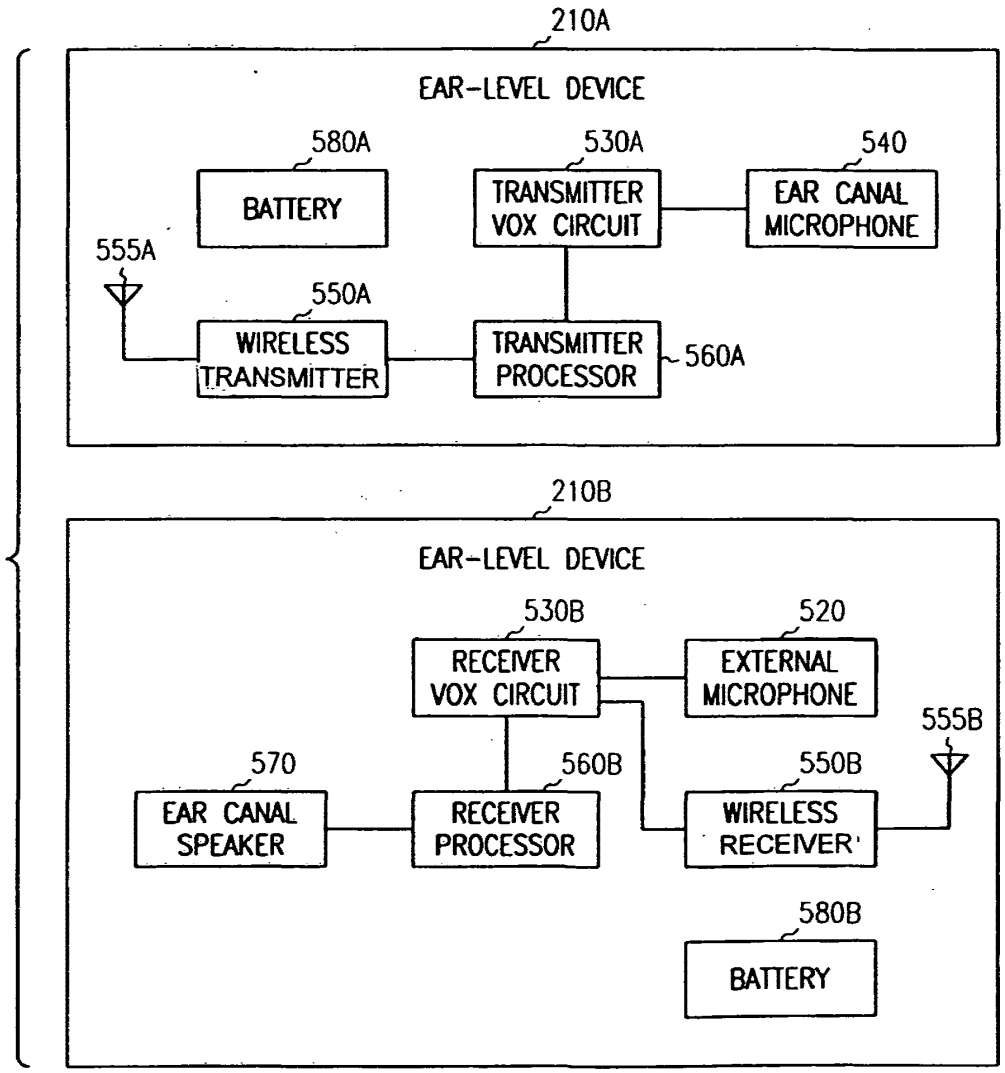


FIG. 5

IPR2024-01015 Page 00486

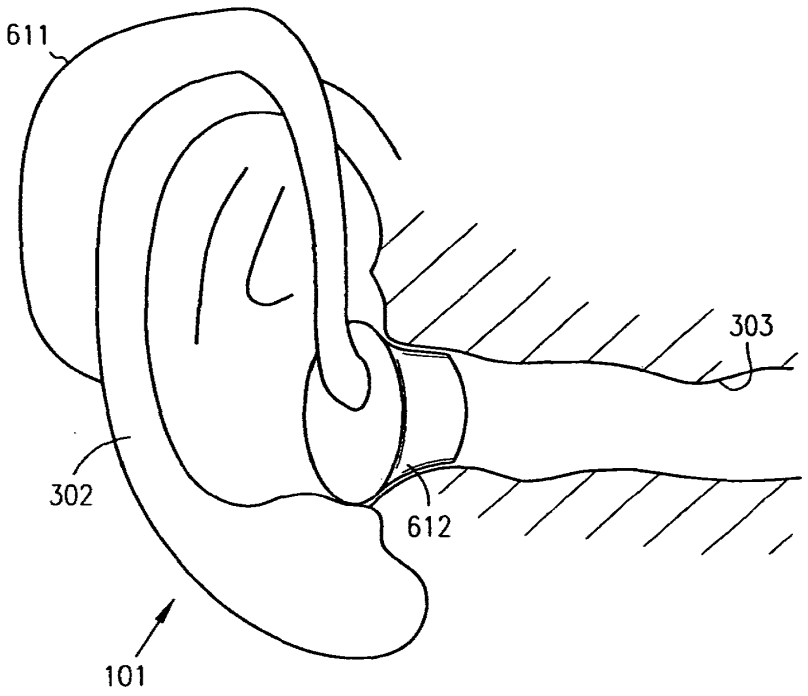

611
303
302
612
101

FIG. 6

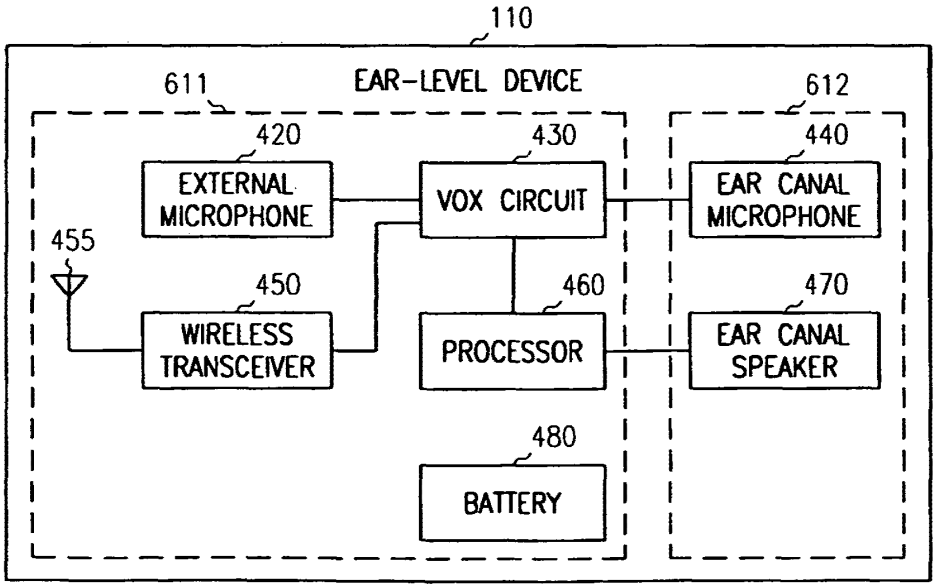

110
611
EAR-LEVEL DEVICE
612
420
430
440
EXTERNAL MICROPHONE
VOX CIRCUIT
EAR CANAL MICROPHONE
455
450
460
470
WIRELESS TRANSCEIVER
PROCESSOR
EAR CANAL SPEAKER
480
BATTERY

FIG. 7

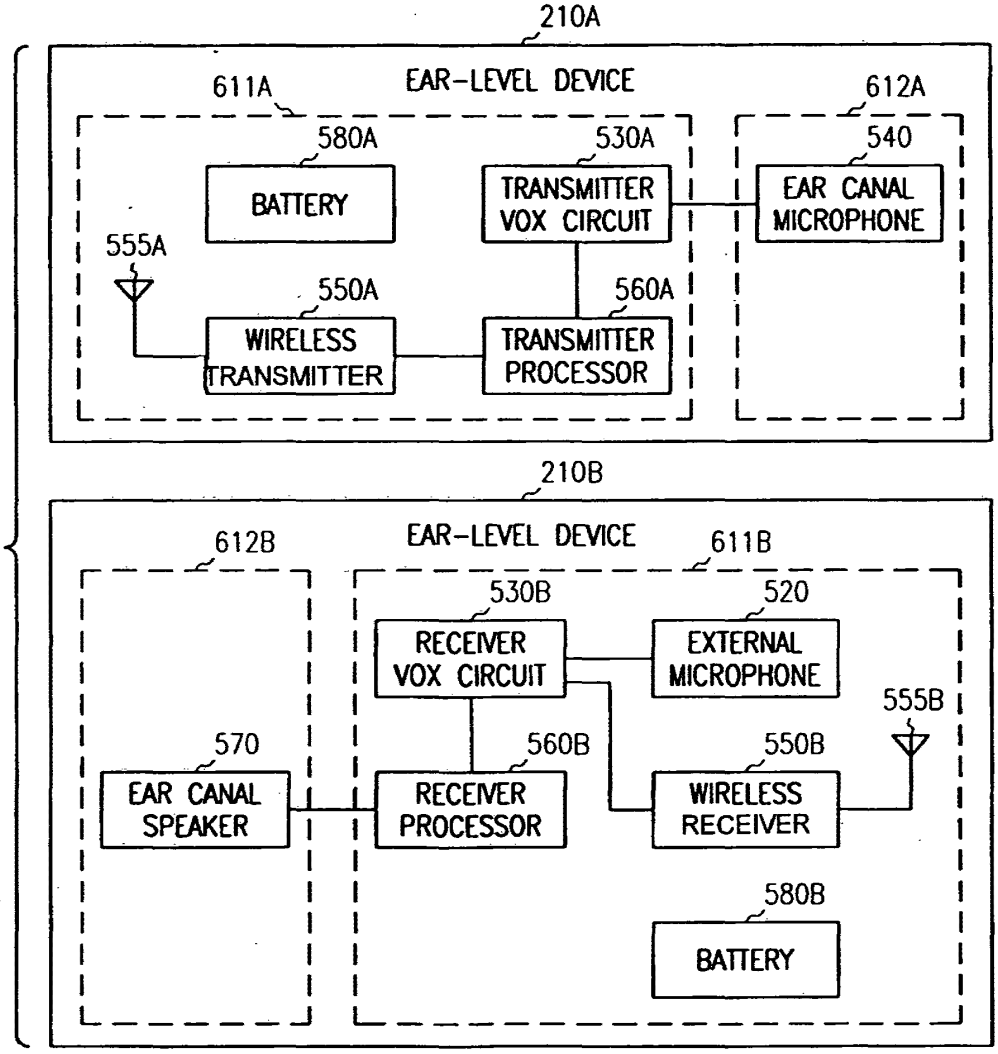

210A
611A
EAR-LEVEL DEVICE
612A
580A
530A
540
BATTERY
TRANSMITTER VOX CIRCUIT
EAR CANAL MICROPHONE
555A
550A
560A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
210B
612B
EAR-LEVEL DEVICE
611B
530B
520
RECEIVER VOX CIRCUIT
EXTERNAL MICROPHONE
555B
570
560B
550B
EAR CANAL SPEAKER
RECEIVER PROCESSOR
WIRELESS RECEIVER
580B
BATTERY

FIG. 8

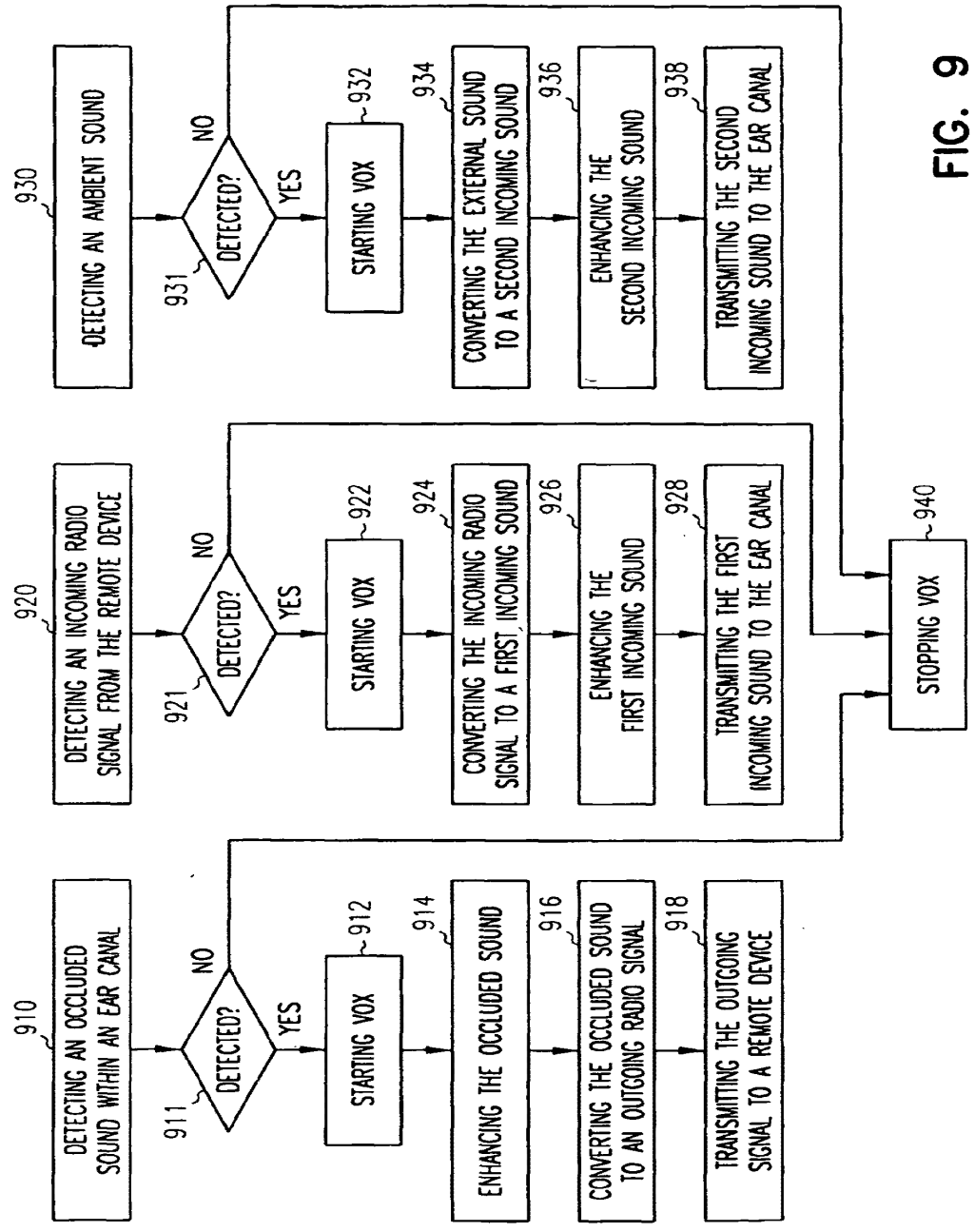


FIG. 9

IPR2024-01015 Page 00489

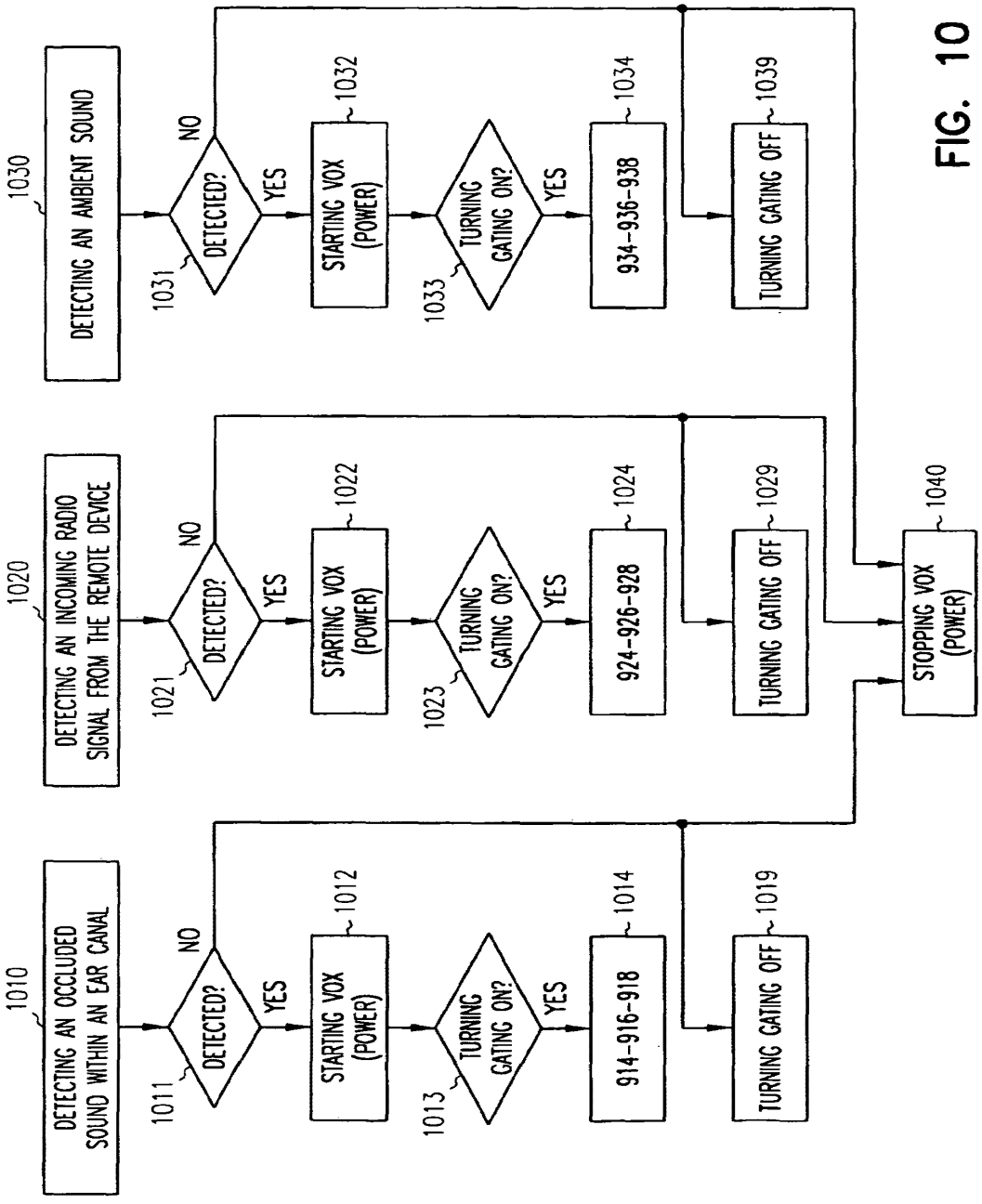


FIG. 10

IPR2024-01015 Page 00490

REFERENCES CITED IN THE DESCRIPTION

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- WO 03073790 A **[0010]**

IPR2024-01015 Page 00491

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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 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.

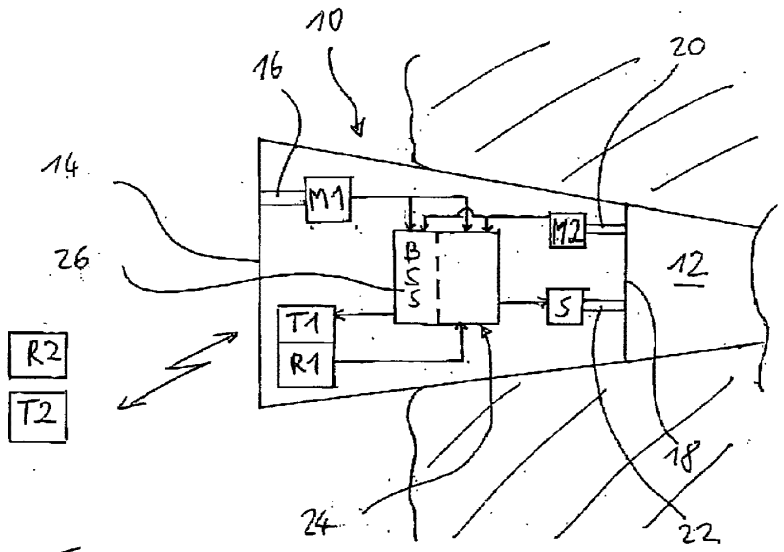


Fig. 1

EP 1 640 972 A1

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

IPR2024-01015 Page 00493

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

IPR2024-01015 Page 00494

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

IPR2024-01015 Page 00495

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

IPR2024-01015 Page 00496

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

IPR2024-01015 Page 00497

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

IPR2024-01015 Page 00498

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.

IPR2024-01015 Page 00499

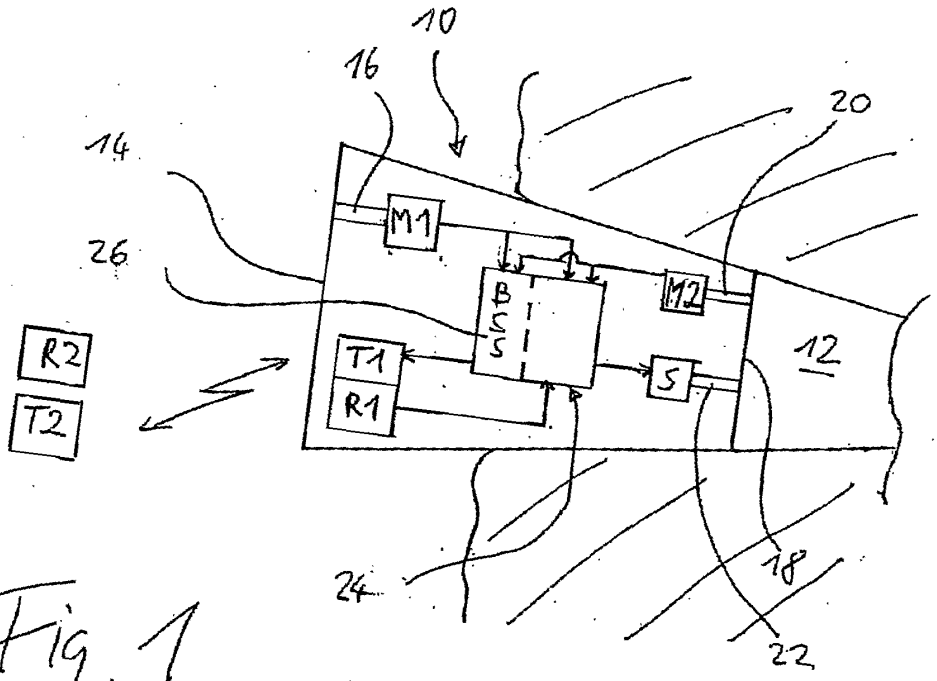


Fig. 1

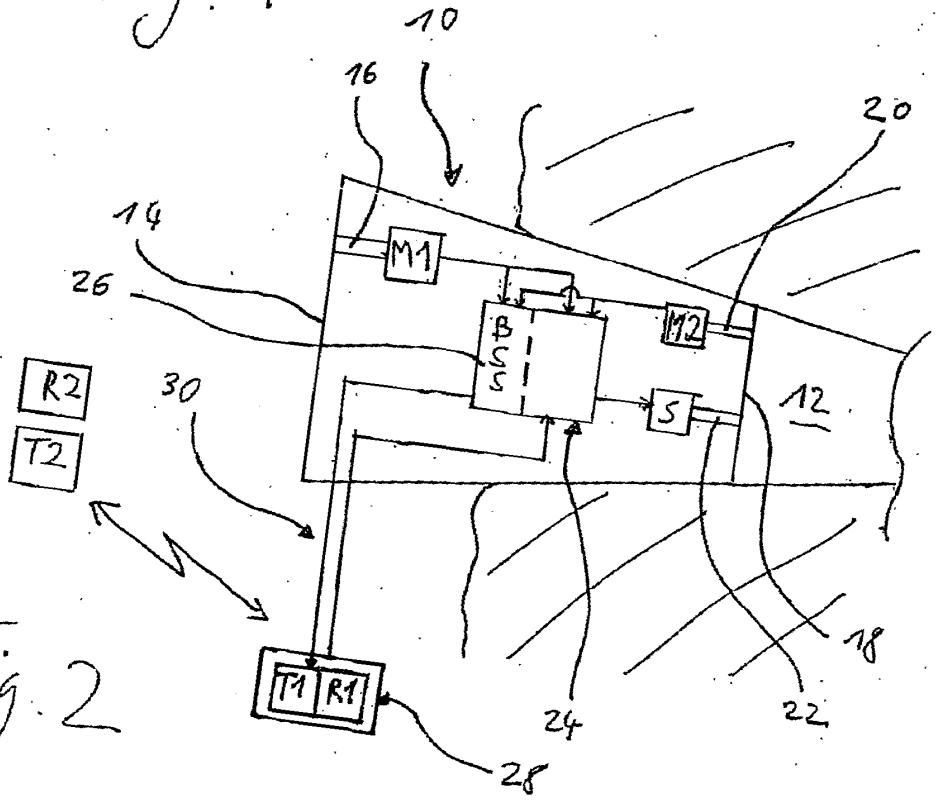


Fig. 2

IPR2024-01015 Page 00500

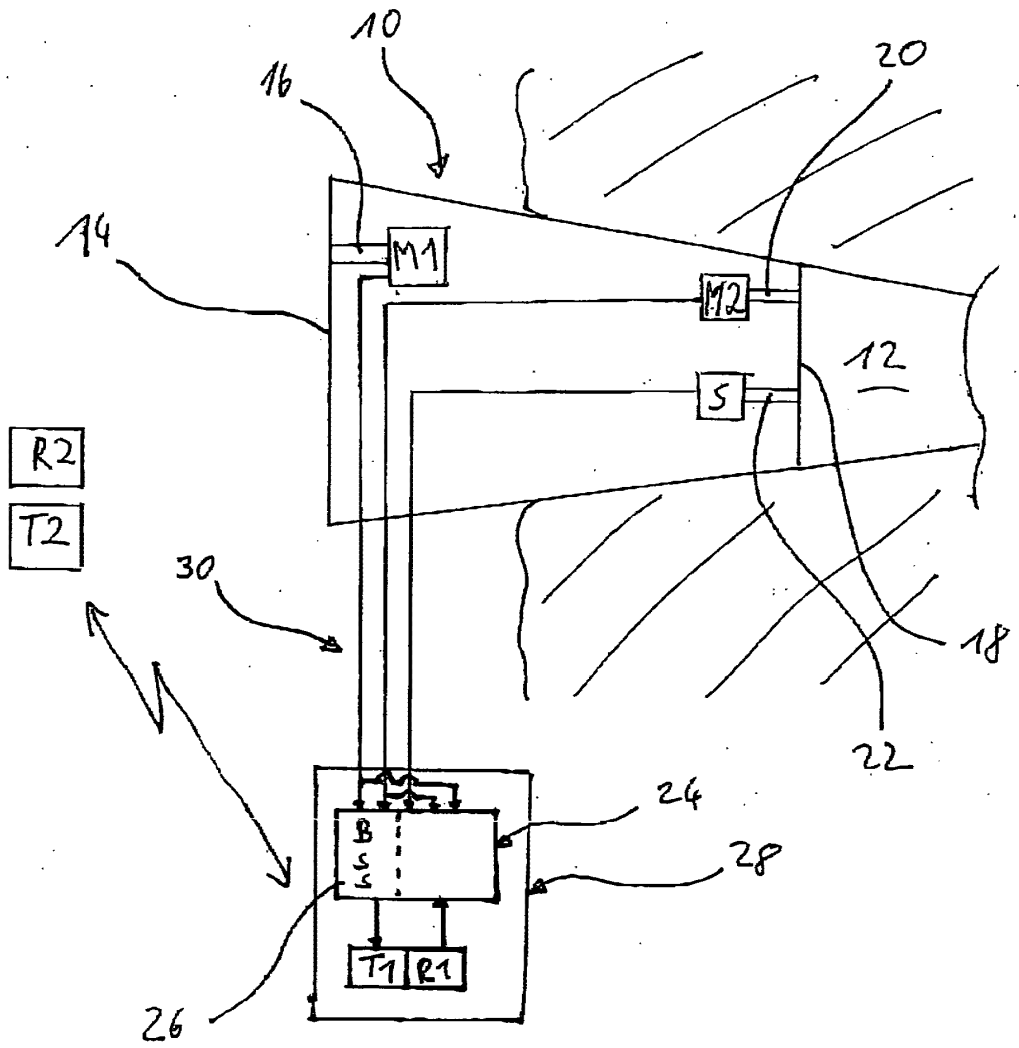


Fig. 3

IPR2024-01015 Page 00501

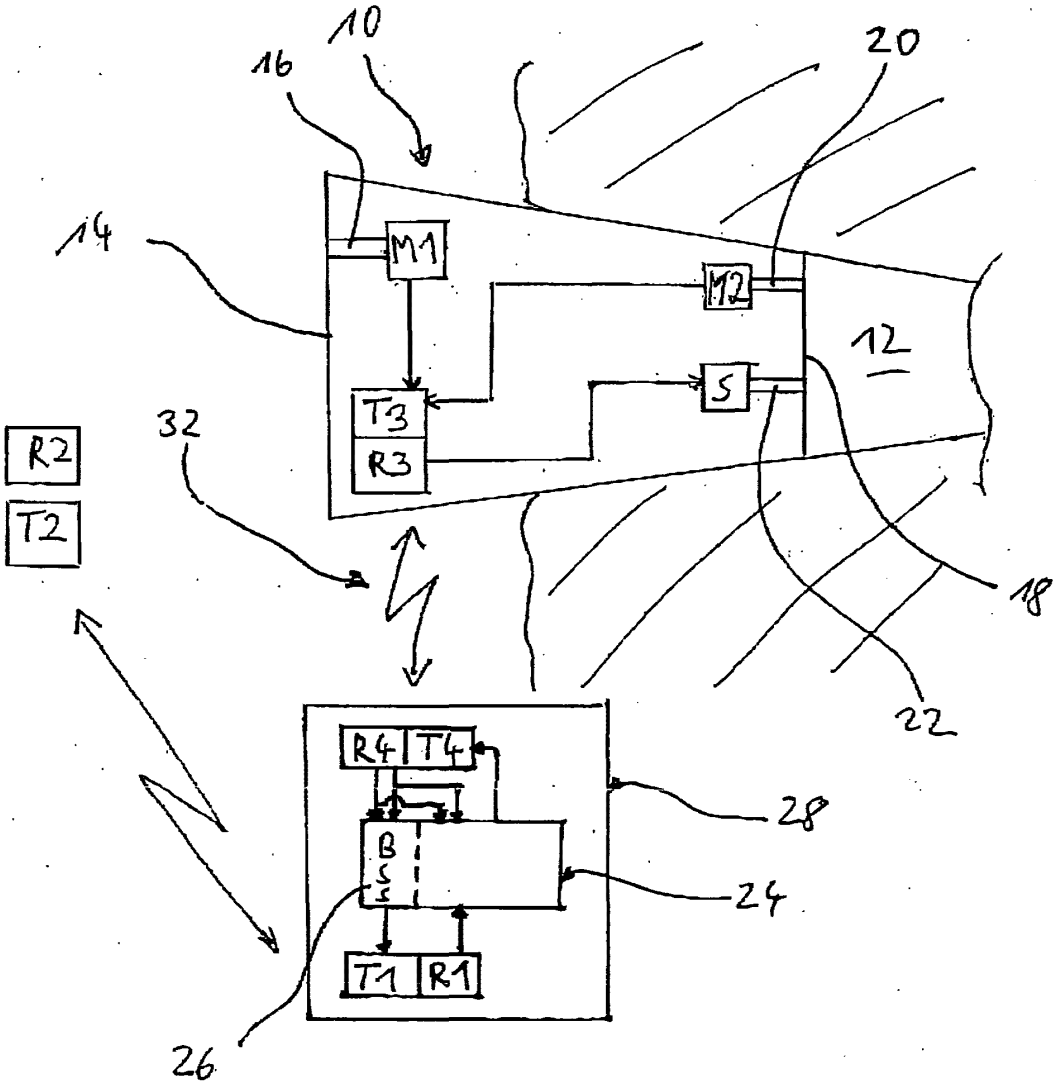


Fig. 4

IPR2024-01015 Page 00502

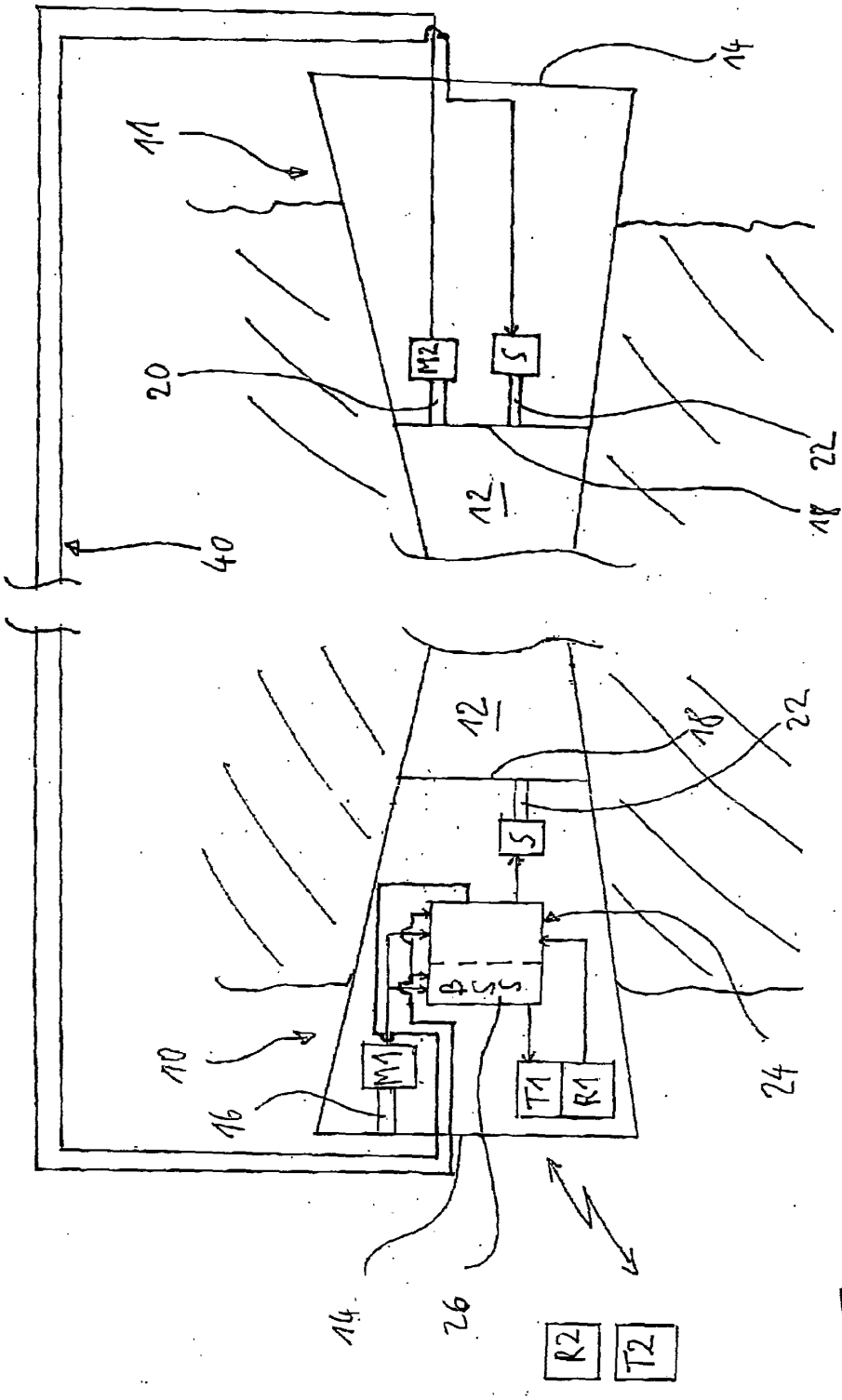


Fig. 5

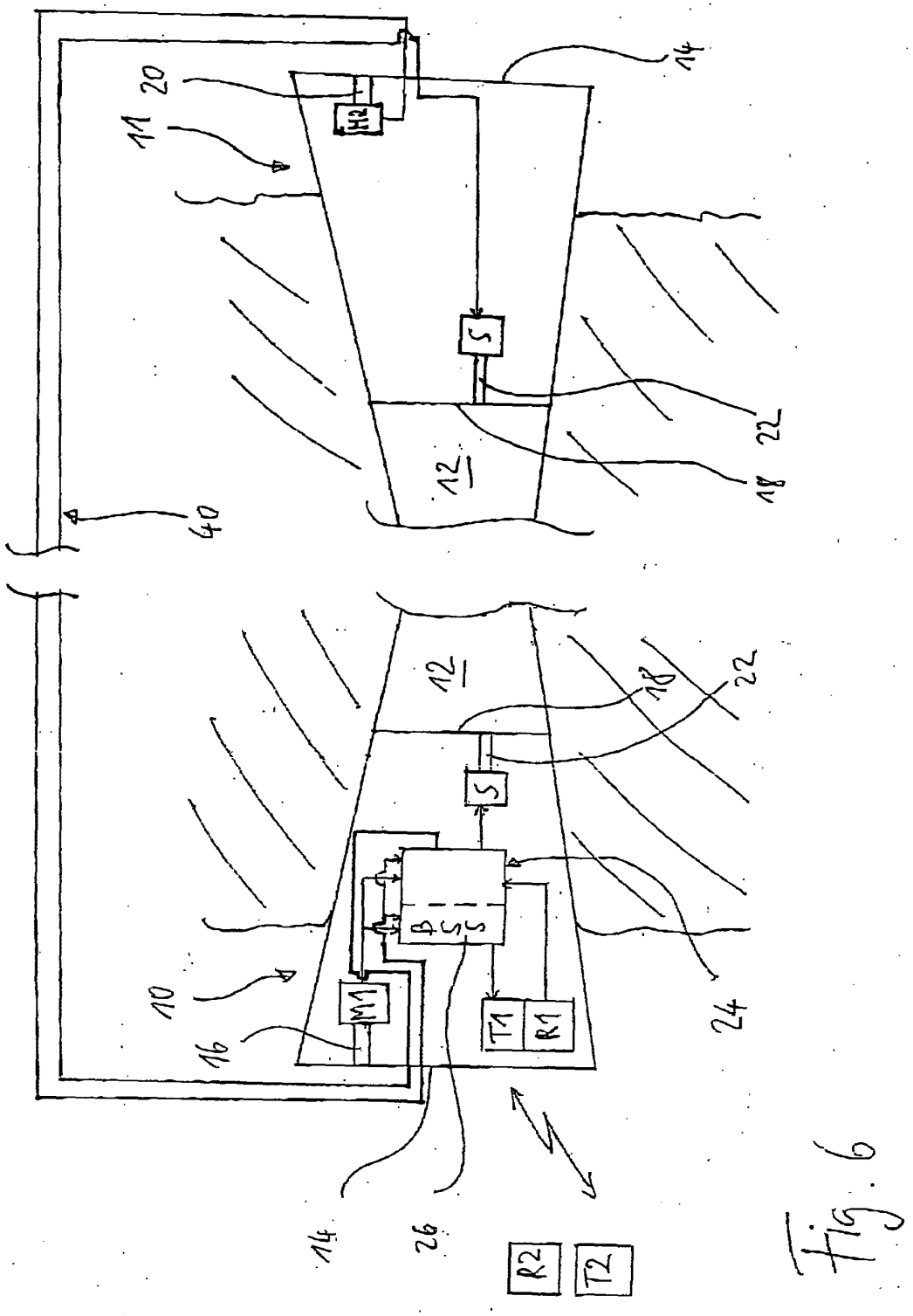


Fig. 6

IPR2024-01015 Page 00504

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 05 02 8366

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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
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G10L
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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

X : particularly relevant if taken alone
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IPR2024-01015 Page 00505

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

IPR2024-01015 Page 00506

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ELECTRONIC ACKNOWLEDGEMENT RECEIPT

APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/182,570	**08/24/2023 10:32:10 PM ET**	**11472-36-3_CONT-16838277**

Title of Invention

ACOUSTIC SEALING ANALYSIS SYSTEM

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	11665493
CONFIRMATION #	5671	FILED BY	Peter Chiabotti
PATENT CENTER #	62677233	FILING DATE	02/23/2021
CUSTOMER #	150139	FIRST NAMED INVENTOR	John Usher
CORRESPONDENCE ADDRESS	-	AUTHORIZED BY	-

Documents

TOTAL DOCUMENTS: 12

DOCUMENT	PAGES	DESCRIPTION	SIZE (KB)
11472-36-4 CONT-17182570 RCE.pdf	3	Request for Continued Examination (RCE)	776 KB
11472-36-4 CONT-17182570 Supplemental IDS.DOCX.pdf	2	Information Disclosure Statement (IDS) Form (SB08)	29 KB
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EP1401240A1-PDF.PDF	14	Foreign Reference	263 KB
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WO2004114722.PDF	19	Foreign Reference	669 KB

EP1519625B1-PDF.PDF	27	Foreign Reference	1036 KB
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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

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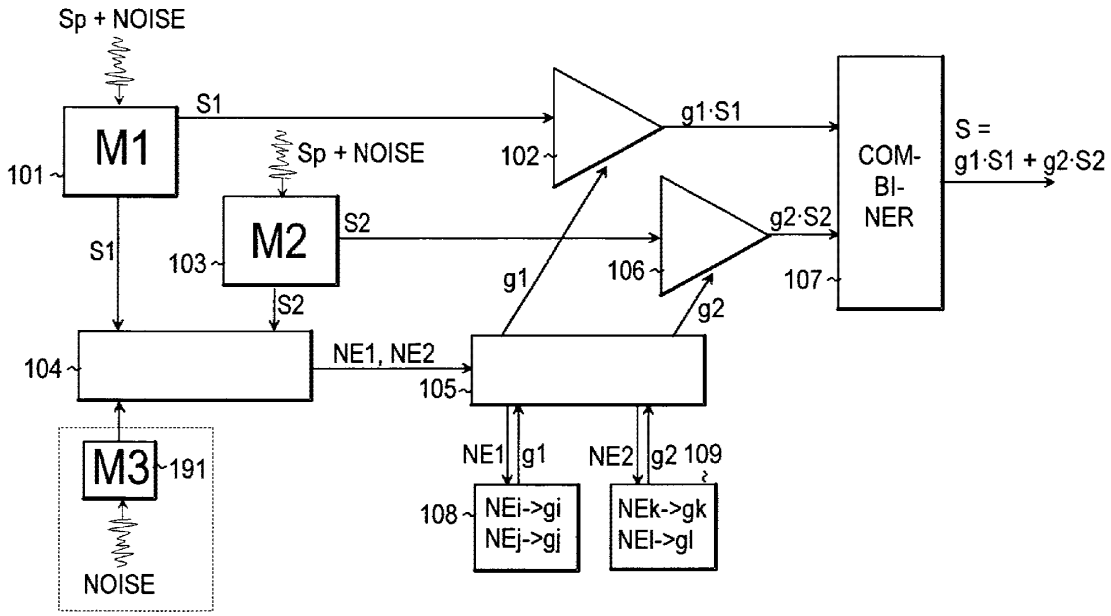
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(54) **A system and method for reducing the effect of background noise**

(57) Acoustical signals are collected with two microphones having different sensitivity to background noise. The level of background noise is estimated, and depending of the level of background noise, i) the amplification gain of the signals is modified before the signals are combined to a combined signal, or ii) the way of combining the signals is changed, to produce a signal that has less background noise than one of the microphones.

FIG 1



EP 1 385 324 A1

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Description

Field of the invention

[0001] The invention relates generally to enhancing speech quality in a telecommunications system, and more specifically to reducing the effect of background noise in the communication between users.

Background of the invention

[0002] Due to the recent explosive increase in the mobile device penetration, it is no more surprising to see people going on the sidewalk or pedestrian area with a mobile phone on their ear. The mobile phone is considered as a practical tool and it is supposed to be carried with to almost anywhere. People are supplied with means for being within reach in principle independently of time and location. As a consequence it may sometimes happen that the surroundings are not optimal for convenient telephoning. In fact, the sight of people trying to make a phone call in noisy places, such as restaurants or street corners, is getting more common all the time.

[0003] Technically, to provide the user with a microphone system which operates in a difficult acoustic environment has not been a difficult task in many years. For example, U.S. patent 5,327,506 discloses a speaker/microphone system that should be applicable in background noise levels up to far beyond over 100 decibels. Such a system is suitable for persons in high ambient noise environments, such as for helicopter pilots. The idea in 506 is that the outgoing voice sounds are taken from the air within the external auditory canal of a person while substantially eliminating audio vibration pickup of sound transmitted by bone conduction.

[0004] Another approach for high background noise microphone arrangements is found in U.S. patent 5,659,620. Here the microphone holder is designed for arrangement in its entirety in the outer ear outside the auditory meatus, and the microphone unit being mounted in vibration transmitting tissue contact in a part of the edge of the holder facing the bottom of tragus. As one of the benefits, all surround tones are not cut off thus increasing the comfort of wearing such a device. When people talk, vibrations in the throat and vocal cords are generated. The vibrations propagate through the bony and liquid-filled tissue, and the idea in 620 is that the jawbone which passes near the front part of the outer ear produces a strong transmission and transfers considerably more of the persons speech than if the signal were collected in the throat.

[0005] The idea of getting almost a background free aural image from inside the ear has also been adapted within the area of mobile communications, as illustrated in the international patent application WO 02/07477. In order to reduce and isolate bone conduction of sound, a sound isolation cup serves as a jacket that surrounds a miniature transducer inside the housing of a microphone/loudspeaker element that is placed in the ear.

[0006] However, under low background noise circumstances, the solutions above suffer from a poorer acoustical quality than an ordinary microphone system. In the 620 approach discussed above, the low spectrum dominates the transmitted sound spectrum because of the acoustical properties of the tissue. This means that in order to keep the sound quality acceptable, or at least to have a proper balance in the sound, the low frequencies have to be filtered. In the other two approaches discussed above, because the acoustic conduct from the speech producing organs to the ear is far from optimal, the sound quality gets badly distorted. Hence, prior art solutions can only be considered as enabling solutions, and in order to keep in pace with the challenging market, the wishes of the quality-conscious consumers of a mobile device with a good sound quality and a possibility to use the device also under high background noise conditions need to be fulfilled. The prior art solutions do not serve this purpose.

[0007] It is an object of the invention to bring about a system and method which provides the users of mobile devices with the best possible audio quality under high background noise circumstances and otherwise reduces the effect of the background noise.

Summary of the invention

[0008] This and other objectives of the invention are accomplished in accordance with the principles of the present invention by providing a system in which the processing of collected audio information is changed according to the estimated level of background noise thus reducing its overall effect.

[0009] More specifically, audio information is collected with two transducers having different sensitivity to the background noise. The background noise is estimated and i) amplification gain of the signals is modified before the signals are combined to a combined signal, or ii) the way of combining the signals is changed, which can be done either digitally or analogously.

[0010] According to the first embodiment of the invention, gain of at least one of the amplifiers adapted to amplify the detected signal is adjusted based on the noise estimate. The signal is then amplified using the adjusted gain thus producing at least one amplified signal. After this step, a combined signal is constructed from one or more of the amplified signals, and in the combined signal, the proportion of background noise is less than in one of the detected signals.

[0011] According to one aspect of the present invention, the system has also storage means for storing at least one noise estimate, and, correspondingly, also comparison means to compare said stored noise estimate with a predefined threshold value. After the comparison, a differential noise estimate is passed to gain control means.

IPR2024-01015 Page 00512

[0012] According to another aspect of the present invention, the system comprises storage means for storing at least two gain control parameters and comparison means for comparing at least two stored gain control parameters. The result of the comparison may be a differential gain parameter to be passed to the amplification means.

[0013] According to another aspect of the present invention, the gain control is adjusted to change the gain between the first and second microphones so that the gain of the first microphone is increased whereas the gain of the second microphone is decreased if the noise estimate indicates that the background noise level is above a first threshold value. Further, the gain control can be adjusted to change the gain of the second microphone substantially close to zero value if the background noise estimate shows that the background noise level is above a second threshold value. Respectively, they can be adapted to adjust the relative gain of the first microphone substantially close to one if the background noise estimate shows that the background noise level is above a pre-set threshold value.

[0014] In one preferred embodiment of the invention the first microphone is to be placed in the auditory canal of a user of the mobile handset; also a jawbone microphone can be used. The second microphone may be a microphone mounted in the mobile handset or in a vehicle, or it may be fixed in a hands-free set of a mobile handset. Examples of such mobile handsets where the invention may be used are mobile phones, PDA devices, laptop computers, dictation machines and so forth.

Brief Description of the Drawings

[0015] In the following, the invention and its preferred embodiments are described more closely referring to the examples shown in FIG. **1-7** in the appended drawings, wherein:

FIG. **1** illustrates one system for reducing the effect of background noise,

FIG. **2** illustrates another system for reducing the effect of background noise,

FIG. **3** shows one microphone set-up according to a preferred embodiment of the invention,

FIG. **4A** shows another microphone set-up according to another preferred embodiment of the invention,

FIG. **4B** illustrates a variation of the set-up shown in FIG. **4A**,

FIG. **5** shows one audio data collection set-up including two microphones,

FIG. **6** illustrates the relationship between noise level estimate and gains with which the signals detected by the microphones are amplified, and

FIG. **7** illustrates how a switch can be included into the system.

Detailed description of the invention

[0016] FIG. **1** illustrates a first system for reducing the effect of background noise, for example, in a microphone. Mechanical vibrations or pressure variations, preferably originating from speech $\mathbf{S_p}$ distorted by noise, is collected with a first transducer serving as a microphone **101**, producing a signal $\mathbf{S_1}$ which is then amplified with a gain $\mathbf{g_1}$ using an amplifier **102** to produce an amplified signal $\mathbf{g_1S_1}$. The system also includes another microphone **103** and a corresponding amplifier **106**, the microphone **103** similarly being exposed to speech $\mathbf{S_p}$ and background noise and producing first a signal $\mathbf{S_2}$ which is after amplification denoted by $\mathbf{g_2S_2}$, where $\mathbf{g_2}$ corresponds to the gain of the amplifier **106**. Both microphones **101** and **103** are transducers that convert acoustical or mechanical vibrations to an electric signal, i.e. normally to a small current, and they exhibit different sensitivity to background noise. The reason for the different noise sensitivity may be that the first microphone **101** is a bone conductor microphone or an earplug type microphone, whereas the second microphone **103** is a normal type transducer commonly used as a microphone in mobile terminals, and therefore more sensitive to noise than the aforementioned types of microphones.

[0017] Preferably, the microphones **101** and **103** are assembled to receive human speech $\mathbf{S_p}$, and in particular such a speech that is going to be further processed in a mobile terminal to be sent to a remote party of the discussion. It is by no means necessary that the speech $\mathbf{S_p}$ to be collected by the microphone (**101** or **103**) is originating from a single person, but may be originating from a conference call etc.

[0018] The output signals $\mathbf{S_1}$ and $\mathbf{S_2}$ of the microphones **101** and **103** are transmitted to a noise estimator **104** which includes means adapted to estimate background noise level thus producing noise estimates $\mathbf{NE_1}$ and $\mathbf{NE_2}$. The noise level can be estimated for both microphones **101** and **103** separately, the output signals **S1** and **S2** of both microphones **101** and **103** may be used to obtain a combined signal from which the noise is estimated, or only the output of the microphone **103** more sensitive to background noise may be used for noise estimation. In the latter two cases only one noise estimate is produced.

[0019] For example, the noise estimation can be done by averaging short periods of the signal **S1** or **S2**. During a silent period, i.e. when the user does not speak, the measured signal level is a good approximation of the

IPR2024-01015 Page 00513

background noise. This kind of approach is commonly used in the Discontinuous Transmission DTX widely used in the Global System for Mobile communications GSM, when instead of the coded silence only so called DTX parameters describing the nature of the background noise are sent, thus reducing obsolete usage of the air interface. The DTX parameters are received in the remote end or in the network between persons talking to each other and some noise similar to background is generated, calling this generated noise Comfort Noise. The idea of Comfort Noise comes from the observation that if during a telephone conversation the loudspeaker gets silent while one person is talking, he or she does not feel comfortable. Modern GSM mobile phones, for example, thus already have means for estimating and describing background noise. Similar means can be applied in the noise estimator 104 for serving the implementation of the invention.

[0020] The noise estimates $\mathbf{NE_1}$ and $\mathbf{NE_2}$ generated are passed to a gain control means **105**. The gain control means **105** adjust the gain $\mathbf{g_1}$, $\mathbf{g_2}$ of the both amplifiers **102** and **106**. In the adjustment, a noise estimate $\mathbf{NE_1}$ or $\mathbf{NE_2}$ is used to determine the desired amplification gain. For this purpose the system may include storage means for noise-amplification relationships, including means **108** and **109**, e.g. tables or functions which respond with a desired gain level $\mathbf{g_x}$ to a query including a corresponding noise level $\mathbf{NE_x}$. For example, means 108 return a gain value $\mathbf{g_1}$ in response to noise estimate $\mathbf{NE_1}$, and means **109** return a gain value $\mathbf{g_2}$ in response to noise estimate $\mathbf{NE_2}$.

[0021] In other words, means **108** and **109** tell the gain control means **105** what is the optimal relation between a noise estimate NE_x and an amplifier gain g_x. Means **108** and **109** are preferably microphone specific, i.e. there are two tables in the system. The gain control means **105** get the noise estimate NE_x, retrieve the gain values g_x corresponding the estimate of present noise NE_x and determine how the gain $\mathbf{g_1}$ and $\mathbf{g_2}$ of the amplifiers **102** and **106** is to be adjusted. In principle, the gain values $\mathbf{g_1}$ and $\mathbf{g_2}$ get values between 0 and 1 so that $g_1 + g_2 = 1$.

[0022] After passing the gain adjustment information $\mathbf{g_1}$ and $\mathbf{g_2}$ to the amplifiers **102** and **106**, the amplifiers amplify the signals S_1 and S_2 using the amplifier gain set based on the gain adjustment information. After amplifying, a combined signal **S** is constructed in the combining means **107**, the construction being made using the amplified signals $\mathbf{g_1S_1}$ and $\mathbf{g_2S_2}$. This may include changing the delay and phase of at least one of the signals g_xS_x so that the distortion resulting from the amplification and the signal path is minimised, and finally summing up the signals g_xS_x so that the combined signal **S** represents a linear combination of the signals $\mathbf{S_1}$ and $\mathbf{S_2}$ detected by the microphones **101** and **103**. This constructed signal S is then ready for being processed further, including all of the numerous steps necessary for transmitting the audio information over the air interface. The combined signal **S** has a smaller proportion of background noise than the output signal $\mathbf{S_2}$ of microphone **103** collected. In this way the effect of background noise can be reduced.

[0023] Preferably, the tables **108** and **109** are generated so that corresponding to a change ΔNE $(=NE_{t=t2}-NE_{t=t1})$ in the noise level NE_x there is a corresponding change Δg $(=g_{t=t2}-g_{t=t1})$ in the gain value g_x. As characteristic to one aspect of the present invention, the gain adjustments of the amplifiers **102**, **106** are opposed to each other; i.e. when the background noise is increasing, the gain $\mathbf{g_1}$ of the signal detected by the first (low-noise) microphone **101** is increased whereas the gain $\mathbf{g_2}$ of the signal detected by the second (more sensitive to background noise) microphone **103** is decreased. It is favourable to normalise the gains so that the relative gain for the both microphones equals to 1.

[0024] In other words, the means **108** and **109** are modified to correspond to the different noise behaviour of the two transducers so that the linear combination of the signals $\mathbf{S = g_1S_1 + g_2S_2}$ detected by the two transducers serving as microphones **101** and **103** can be subjectively (by listening) considered optimal. This may include different threshold values: The first threshold value corresponds to the value of at least one of the noise estimates, at which value the quality of signal $\mathbf{S_2}$ of the second microphone **103** that is more sensitive to background noise starts to deteriorate; this is the point where the relative gain of the first microphone **101** is increased from substantially zero gain value to a small value. The second threshold value corresponds to the value of at least one of the noise estimates NE_x at which the quality of the detected signal of the second microphone **103** is so much worse than the quality of the signal $\mathbf{S_1}$ detected by the first microphone **101** that the relative gain $\mathbf{g_2}$ of the second microphone **103** is set to substantially zero gain.

[0025] As an example, if the first microphone detects a background noise level of 2,0 mV (corresponding to a noise 70 dB) and the second microphone detects a background noise level of 5,0 mV (corresponding to a noise of 72 dB), the average noise level estimated in the noise estimator may be between 70 and 72 dB. The relative gains for the microphones read from the table are then 0,4 and 0,6, respectively. This is not straightforward, because the sensitivity of the microphone plays a significant role; the signal levels may need to be adjusted to correspond some reference level. The relative gains are then to be understood against this reference level only.

[0026] When the detected background noise jumps to 2,2 mV (71 dB) and 7,0 mV (76 dB), the average noise level may be estimated to be between 71 and 76 dB. The relative gains for the microphones read from the table are then 0,95 and 0,05, respectively. The second microphone is more sensitive to the background noise, i. e. the background noise generates a bigger impulse in the microphone. Secondly, the second microphone is

IPR2024-01015 Page 00514

not yet shut-off completely, because the noise level is not above a defined shut-off threshold value for background noise.

[0027] The transition at different moments of time between subsequent values of the relative signal $\mathbf{S_1}$ and $\mathbf{S_2}$ proportions in the combined signal **S** defined by a ratio $\mathbf{g_1/g_2}$ (when $g2 \neq 0$) can thus be made smooth or at least to correspond small steps in order to avoid abrupt transitions that might be considered annoying by the listener. This corresponds to successive changes of $\mathbf{g_1}$ and $\mathbf{g_2}$. Nevertheless, it is to be noted that this is not necessary, but the transition may resemble a step function, i.e. the gain $\mathbf{g_1}$ for the signal detected by the first microphone **101** is zero below a threshold value and one otherwise, and the gain $\mathbf{g_2}$ for the signal detected by the second microphone is then reversed, i.e. it obtains values $1\text{-}\mathbf{g_1}$.

[0028] FIG. **6** shows an example of the relation between estimated noise level $\mathbf{NE_2}$ estimated from the signal $\mathbf{S_2}$, and the amplifier gain g_1. In this case, $g_2=1\text{-}\mathbf{g_1}$. At times preceding **TA**, $\mathbf{NE_2}$ is smaller than the first threshold value. Correspondingly, $\mathbf{g_1}=0$ and $g_2=1$. At **TA**, the value of $\mathbf{NE_2}$ is increasing, so $\mathbf{g_1}$ is stepwise increased. At **TB**, $\mathbf{NE_2}$ has reached the second threshold value, and $\mathbf{g_1}$ is set to one. At **TC**, $\mathbf{NE_2}$ is again smaller than the second threshold value, so the value of **g1** must be decreased. At **TD**, $\mathbf{NE_2}$ falls below the first threshold value and **g1** is set to zero value. At the time interval $\mathbf{P_1}$ the change in $\mathbf{NE_2}$ is less than the threshold $\Delta\mathbf{NE}$ set for the change of **NE**, so during this interval $\mathbf{g_1}$ is not changed. Immediately after $\mathbf{P_1}$, when the change in $\mathbf{NE_2}$ is larger than the threshold value, the $\mathbf{g_1}$ is changed by $\Delta\mathbf{g}$. During the interval $\mathbf{P_2}$, $\mathbf{g_1}$ is not changed because $\mathbf{NE_2}$ shows sharp peaks; this may present some distortion so the changes are neglected.

[0029] The second predefined threshold value can be chosen to be a fixed value as well. The value may be experimentally determined to correspond to the level of background noise in which the input signal of the second microphone **103** has been deteriorated in such a bad way that in order to get a better signal quality for the signal **S,** only the input signal $\mathbf{g_1S_1}$ is used.

[0030] FIG 7 shows how a second system according to the invention, the system having only a stepwise transition for the value of $\mathbf{g_1}$. Speech **Sp** and noise are detected by microphones **101** and **103** producing signals **S1** and **S2**. Background noise is collected by a third microphone **191** directly connected to noise estimator **104**. The noise estimator estimates the level **NE** of background noise and passes it to gain control means **705**. The gain control means **705** are adapted to compare the noise level estimate **NE** with a predefined threshold value. If the value of **NE** is less than the threshold value, **g1** is set to zero, otherwise **g1** gets the value of 1. The value of **g1** indicates the position of a switch **SW1** controlled by the gain control means **705**, the switch **SW1** being so arranged that the terminals **TE1** and **TE2** are connected when **g1=1**; **TE1** and **TE3** are connected in all other cases. The terminal **TE1** leads to amplification means **702** which produce an amplified signal **AS**. Because only one signal is fed (S1, if TE1 and TE2 are connected) to the amplifier, no literal combining of the signals (S1, S2) need to be performed.

[0031] The switch turns the relative gain $\mathbf{g_2}$ of the second microphone to zero and the relative gain of the first microphone $\mathbf{g_1}$ to one. In this case the function of the switch **SW1** and amplification means **702** may be understood as linear amplification with amplification factor 1.0 when the signal received by a microphone is amplified and zero otherwise.

[0032] Further, in order to avoid annoying abrupt transitions between the first and second microphones, or to avoid fluctuations in the microphone gains, the system like presented in FIG. **1** may comprise also storing means adapted for storing noise parameters in order to form a noise history. The noise parameters can be calculated for one microphone, both microphones separately or they can be obtained by combining the values computed from the inputs of the both microphones. One possibility is to use the second microphone input as the preferable source for noise estimation, as the first microphone is less sensitive to background noise.

[0033] If history data is collected, the system may further be adapted to have a hysteresis effect, i.e. if the noise level is having fluctuations back and forth, the system can adapt the threshold value to correspond a larger shift in the background noise. In order to have a hysteresis effect, there are storage means and comparison means; the hysteresis information may be obtained from a stored table, the table being read in a way taking into account the noise level history.

[0034] Similarly, the history data may be used for making prognoses or extrapolations for the noise level, and in this way the system may, based on the noise level measured at least in two earlier moments of time, predict a noise level in the future thus making the possibly larger transitions in gains over a longer period of time.

[0035] A third microphone **191** is optionally included in the system. Its main purpose is to detect background noise. It may be located in the vicinity of the noise estimation means **104** so that the transmitting of the signal to the noise estimation means **104** from the microphone **191** is straightforward. In this case the signals $\mathbf{S_1}$ and/or $\mathbf{S_2}$ need not to be passed from the other two microphones **101** and **103** to the noise estimator **104**. This makes the system simpler, as the signal from microphones **101** and **103** need only travel the path to amplifiers **102** and **106**. For example, in a wireless hands-free set the microphone **101** is located in one ear of the user, whereas the second microphone **103** is fixed in the car of the user. The amplifiers are located close to the microphones, especially because the signals are to be transmitted over the air interface to the combiner means **107**. This means that the noise estimator **104** and gain control means **105** need to instruct the amplifiers **102** and **106** separately; the noise may be measured by the

IPR2024-01015 Page 00515

microphone **191**, and the gain parameters $\mathbf{g_1}$ and $\mathbf{g_2}$ be broadcast to the amplifiers **102** and **106**.

[0036] FIG. **2** shows a third system according to the present invention. The transducers **101** and **103** are exposed to speech signal $\mathbf{S_p}$ distorted by background noise. They detect acoustical vibrations, i.e. mechanical vibrations or pressure changes. Amplifiers **202** and **206** amplify the signals $\mathbf{S_1}$ and $\mathbf{S_2}$ with constant gains thus producing amplified signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$, respectively. An Analog to Digital Converter ADC **207** converts the signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$ pre-amplified by amplifiers **102** and **106** into a digital form. In addition to performing this step, the **ADC** may further include combining means to form a combined signal S from at least one of the amplified signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$ in such a manner that the relative ratio of the signals may be altered. The combining may be performed digitally, thus possibly including multiplication and summing operations, or it may be performed for the analog signals before the conversion.

[0037] The noise level is detected in noise estimator **204**, including tables **208** and **209** showing noise responses of the microphones **101** and **103** in a manner similar to the tables discussed above. The noise estimates NE_x are then used to deduce the target gain factors g_x. The target gain factors $\mathbf{g_1}$ and $\mathbf{g_2}$ which correspond to combining parameters are lead to the combining means of the ADC **207**, where the ADC either, based on the combining parameters g_1 and g_2, performs the conversion and only after that digitally multiplies the converted signal by the gain factor, or, uses the target gain factor to change the relative strength of the signals before the conversion.

[0038] In a fourth system according to the present invention, the system may have storage means for storing at least one noise estimate $\mathbf{NE_1}$ or $\mathbf{NE_2}$ in the noise estimator **104**, whereas the system otherwise corresponds to the system already discussed with reference to FIG. **1**. This means that at least one noise estimate $NE_t=_{tpast}$ value may be stored, and, correspondingly, at a later point in time $t=t_1$, at least one other noise level estimate $NE_{t=t1}$ may be stored as well. The stored noise estimates may be compared using comparison means located preferably in the noise estimator **104**. The comparison may involve computing a noise level difference $\Delta NE = NE_{t=t1}-NE_{t=tpast}$ describing the changing of the background noise level. This difference may be transmitted then to gain control means **105** which change gains according to a function $\Delta g=f(\Delta NE)$. It is also possible that the difference is not passed to gain control means **105** if it does not exceed a preset threshold value, i.e. if $\Delta NE < \Delta NE_{threshold}$; only changes in the gain when the change in the background noise level is larger than the threshold value are passed to the amplifier **102** or **106**. In this case the older value measured at t_{past} may remain stored in the register and newer noise estimates $NE_{t=t1}$, calculated at times $t_1' > t_1$ are overwritten on the value $NE_{t=t1}$.

[0039] A fifth system in accordance with the present invention, the system comprises storage means such as a register in the gain control means **204** in addition to other components as discussed with reference to FIG. **2**. The storage meansmay be used for storing at least two gain control parameters. Further, the system may also include comparison means for comparing at least two stored gain control parameters. Like discussed above, the comparison of the parameters may include performing some mathematical operations, like calculating the difference between a newer and an older gain parameter value, and, in the case that a pre-set threshold value for the difference is exceeded, which can be checked again by comparison, the system may calculate a new gain to be passed to the ADC **207**. The new gain may also be selected from a plurality of different gain set-ups, if the system is to be kept simple and fast.

[0040] FIG. **3** shows a microphone set-up according to one preferred embodiment of the present invention. The mobile phone in the figure has two microphones **M1** and **M2**. The second microphone **M2** is an ordinary microphone denoted by asterisks **30** describing holes in the cover of the bottom half of the phone. The first microphone **M1** is located close to the loudspeaker which is denoted by asterisks, whereas the microphone **M1** is denoted by arc **31**. Because the loudspeaker is - usually - placed close to the ear, using a jawbone contact element as the microphone **M1**, the vibrations in the jawbone of the user can be detected thus producing a signal S_1. The sensitivity of **M1** to disturbances from the loudspeaker is reduced because **M1** detects mechanical vibrations from the jawbone of the user.

[0041] FIG. **4A** further illustrates a microphone set-up according to one other preferred embodiment of the present invention. The first microphone **M1** is located in an earplug-like element which can measure the air vibrations in the ear canal (i.e. auditory canal) of the user of the mobile device. Depending of the actual realisation of the invention, this earplug microphone can be similar to those disclosed in the referenced publications. Some of the prior art earplug microphones have a wireless transmitter, such as a BLUETOOTH transmitter.

[0042] A solution according to FIG. **4A** is especially suitable for different hands-free applications. For example, many countries in Europe have recently adopted restrictions for usage of mobile phones while driving a vehicle. In order to comply with the legislation, a driving mobile user willing to receive a phone call need either to stop or to have a hands-free equipment installed in his or her car.

[0043] Some hands-free equipment sets include a speaker, a microphone and then some wiring systems, possibly with an extra antenna. The speaker is typically mounted somewhere in the cabin, whereas the microphone is located close to the head of the driver. As known, even though some of the different speech codecs widely used in the mobile devices enhance the speech quality in noisy environments, such as in a moving car, the background noise may still be annoying and,

IPR2024-01015 Page 00516

depending of the car, even make the discussion impossible, depending on the circumstances such as other traffic, speed of the vehicle, road conditions etc. This drawback can be eliminated if the user is supplied with an earplug-like device in addition to the normal hands-free set installed in the car. As an alternative, a jawbone microphone can be used as well, while there are some of them available on the market already. Preferably the first microphone (either earplug or jawbone) is equipped with a wireless transmitter. Then either the mobile handset or the hands-free set includes a receiver to receive the transmitted audio information, such as BLUETOOTH low-power RF data.

[0044] The transmitter needs not to be wireless but may be a conventional earplug microphone as well like illustrated in FIG. **4B**, where there is a wire including two lines, one for the signal and one for the ground level. In the other end the wires may be connected to the front panel of the car, the panel supplying a plug-in for the microphone data and further connected to the hands-free set, or directly to the mobile handset.

[0045] FIG. 5 illustrates another option for implementing the invention. Here the first and second microphones are included in a headset kit of a mobile phone. Typically, the headset kit includes a plug **501** which is connected to a mobile phone or a car front panel. The handset provides hands-free operation and may also include an answer/end button located in the second microphone **M2** or somewhere else in the cable. The earplug including a loudspeaker may also include an earplug microphone **M1**, provided that the acoustical separation of the loudspeaker and microphone can be guaranteed, or a jawbone microphone, where the acoustical separation should be a minor problem.

[0046] For a person skilled in the art it is also clear that no implicit tables like those described with reference to FIG. **1** and **2** are necessary. Depending on the number of steps for different amplifications needed, the system can be coded directly if the number is low, otherwise also a numerical model which interpolates some experimentally tested values may also be used.

[0047] Although the invention was described above with reference to the examples shown in the appended drawings, it is obvious that the invention is not limited to these, but it may be modified by those skilled in the art without departing from the scope and the spirit of the invention.

Claims

1. A system for reducing the effect of background noise especially in a mobile handset, the system comprising

   - a first transducer (101) for detecting an acoustical signal producing a first signal detected (S1),
   - a second transducer (103) for detecting an acoustical signal producing a second signal detected (S2), the first and second transducers having different sensitivities to background noise,
   - amplifier means (102, 106) for amplifying the first and/or second signal (S1, S2) detected thus producing at least one amplified signal (g1S1, g2S2),
   - noise estimator means (104) adapted to estimate a background noise level present at least in one of the detected signals (S1, S2) or in a signal measured by a third transducer (191) thus producing at least one noise estimate (NE1, NE2),
   - gain control means (105), responsive to said noise estimator means (104, 204), adapted to adjust gain (g1, g2) of the amplification means (102, 106, 207), the adjustment being made depending on at least one noise estimate (NE1, NE2), and
   - combining means (107), adapted to form a combined signal (S) from at least one amplified signal (g1S1, g2S2) thus reducing the effect of background noise.

2. A system according to claim 1, **wherein** the noise estimator means are connected to the first and/or second transducer (101, 102) for receiving noise level information.

3. A system according to claim 1 or 2, the system further **comprising**

   - storage means for storing at least one noise estimate, and
   - comparison means for comparing the stored noise estimate with a predefined threshold value, the comparison producing a differential noise estimate to be passed to gain control means (105).

4. A system according to claim 1 or 2, the system further **comprising**

   - storage means for storing at least two gain control parameters, and
   - comparison means for comparing the two gain control parameters, the comparison producing a differential gain parameter to be passed to the amplification means.

5. A system according to claim 1 or 2, **wherein** the gain control means (105) are adapted to adjust gain (g1, g2) between the first and second amplifiers (102, 106) so that the gain (g1) of the first amplifier (102) is increased whereas the gain (g2) of the second amplifier (106) is decreased if the noise esti-

IPR2024-01015 Page 00517

mate (NE1, NE2) indicates that the background noise level is above a first threshold value.

6. A system according to claim 1, 2 or 5, **wherein** the gain control means (105, 207) are adapted to adjust gain (g2) of the second amplifier substantially close to zero value if the background noise estimate (NE) shows that the background noise level is above a second threshold value.

7. A system according to claim 1, 2, 5 or 6, **wherein** the gain control means (105, 207) are adapted to adjust the relative gain of the first transducer substantially close to one if the background noise estimate (NE) shows that the background noise level is above a pre-set threshold value.

8. A system according to claim 1, **wherein** the construction is performed by adding the amplified signals (g1S1, g2S2).

9. A system according to any one of the preceding claims, **wherein** the first transducer (101) is a microphone to be located in the auditory canal of a user of the mobile handset.

10. A system according to any one of the preceding claims, **wherein** the first transducer (101) is a jawbone microphone.

11. A system according to claim 9 or 10, **wherein** the second transducer (103) is a microphone mounted in the mobile handset.

12. A system according to claim 9 or 10, **wherein** the second transducer (103) is a microphone fixed in a hands-free set of a mobile handset.

13. A system according to claim 1 or 2, **wherein** said gain control means (105, 207) are located in the mobile handset or in the hands-free set of a mobile handset.

14. A method for reducing the effect of background noise especially in a mobile handset, the method **comprising** the steps of

- collecting a first acoustical signal (S1),
- collecting a second acoustical signal (S2), whereby in the collected signals (S1, S2) a different level of background noise is present,
- estimating background noise level from at least one of the signals (S1, S2) collected,
- in response to said estimating step, adjusting gain (g1, g2) of at least one amplifier (102, 106) adapted to amplify signals collected (S1, S2),
- in response to said adjustment step, amplifying said collected first and/or second signal (S1, S2) to produce at least one amplified signal (g1S1, G2S2), and
- in response to said amplifying step, constructing a combined signal (S) from at least one amplified signal (g1S1, g2S2) thus reducing the effect of background noise in the combined signal.

15. A method according to claim 14, further **comprising** the steps of

- producing and storing at least one noise estimate (NE1, NE2) in the background noise estimation step, and
- comparing said stored noise estimate (NE1, NE2) with a predefined threshold value, the comparison result being used in the gain adjustment step.

16. A method according to claim 14, further **comprising** the steps of

- in the gain adjusting step, storing at least two gain control parameters, and
- comparing at least two stored gain control parameters in order to decide whether the gain adjusting step is to be taken, the gain adjusting step being performed only if the comparison result shows that a gain control parameter being larger than a predefined threshold value.

17. A method according to claim 14, **wherein** the gain adjustment step is adapted to adjust amplification balance between the signals detected by first and second transducers (101, 103) so that the gain (g1) of the amplifier (101) amplifying the first signal (S1) collected is increased whereas the gain (g2) of the amplifier (106) amplifying the second signal (S2) is decreased if in the noise estimation step it is detected that the background noise level is higher than a first threshold value.

18. A method according to claim 14, **wherein** the gain adjustment step is adapted to adjust gain (g2) of the amplifier amplifying the second signal (S2) substantially close to zero value if the background noise estimate (NE) shows that the background noise level is above a second threshold value.

19. A method according to claim 14, **wherein** the construction step includes adding the amplified signals (g1S1, g2S2).

20. A method according to any one of claims 14-19, **wherein** the first signal is collected using a microphone to be located in the auditory canal of a user of the mobile handset.

IPR2024-01015 Page 00518

21. A method according to any one of claims 14-19, **wherein** the first signal is collected using a jawbone microphone.

22. A system for reducing the effect of background noise especially in a mobile handset, the system **comprising**

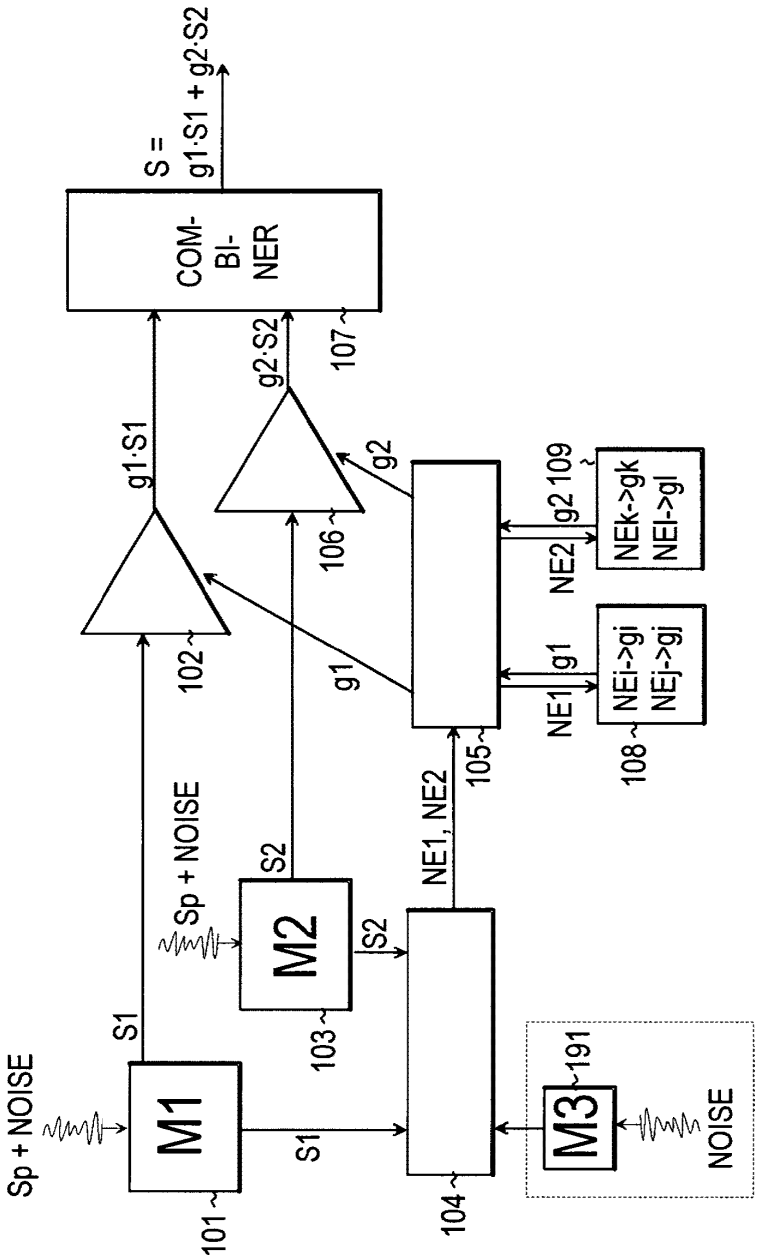
- a first transducer (101) for detecting an acoustical signal producing a first signal detected (S1),
- a second transducer (103) for detecting an acoustical signal producing a second signal detected (S2), the first and second transducer (101, 103) having different sensitivity to background noise,
- amplification means (202, 206) for amplifying the first and/or second signal (S1, S2) detected thus producing at least one amplified signal (AS1, AS2),
- combining means (207) adapted to form a combined signal (S) from at least one of the amplified signals (AS1, AS2) using at least one combining parameter (g1, g2) thus reducing the effect of background noise,
- noise estimator means (204) adapted to estimate background noise level present at least in one of the detected signals (S1, S2) or in a signal measured by a third transducer (191) thus producing at least one noise estimate (NE1, NE2),
- combining parameter generation means (208, 209) adapted to generate at least one combining parameter (g1, g2) in response to at least one noise estimate (NE1, NE2), and
- means (204) for passing the generated combining parameter (g1, g2) to the combining means (207).

23. A method for reducing the effect of background noise especially in a mobile handset, the method **comprising** the steps of

- collecting a first acoustical signal (S1),
- collecting a second acoustical signal (S2), whereby in the collected signals a different level of background noise is present,
- estimating background noise level (NE1, NE2) from at least one of the signals collected or from a third acoustical signal,
- in response to the background noise estimating step, using at least one background noise level parameter (NE1, NE2) for selecting at least one combining parameter (g1, g2), and
- in response to said selecting step, using said summing parameter for combining the first and acoustical signals (S1, S2) to form a combined signal (S) which has less background noise than the second signal collected.

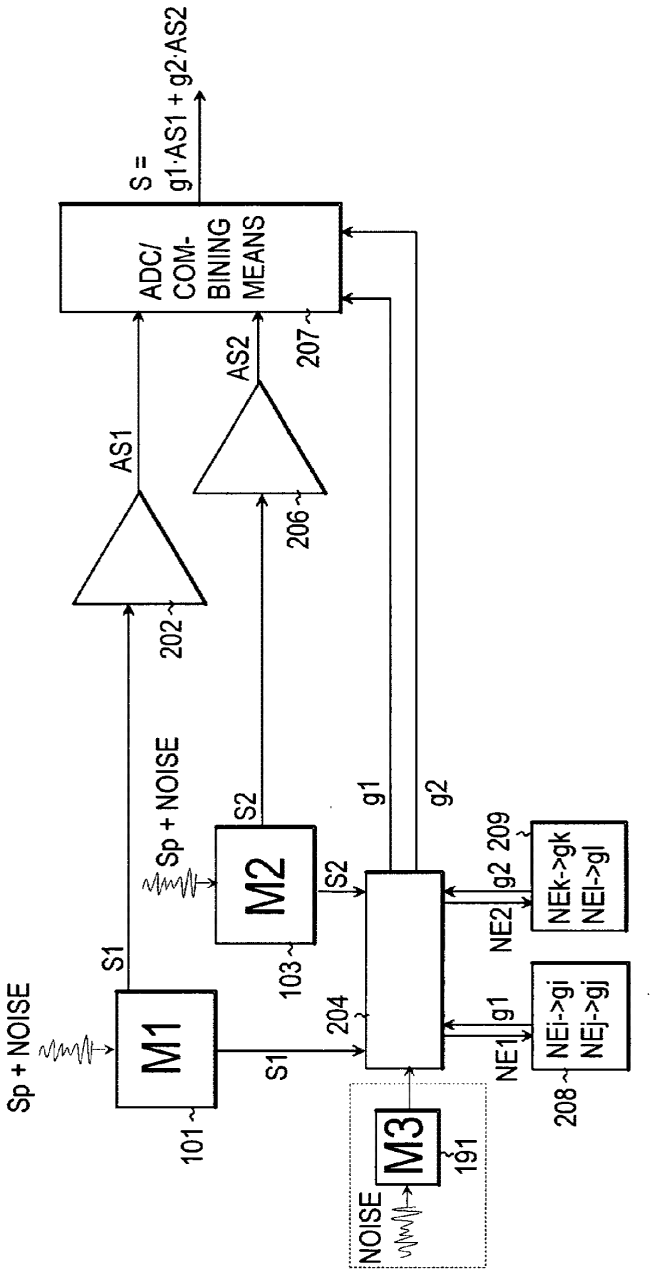
IPR2024-01015 Page 00519

FIG 1



IPR2024-01015 Page 00520

FIG 2



IPR2024-01015 Page 00521

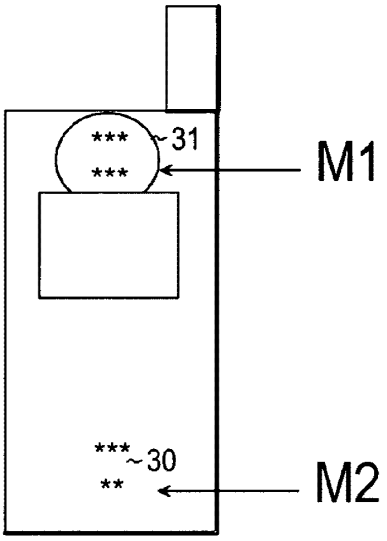


FIG 3
(LEFT)

FIG. 4A

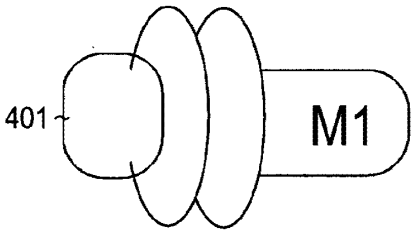
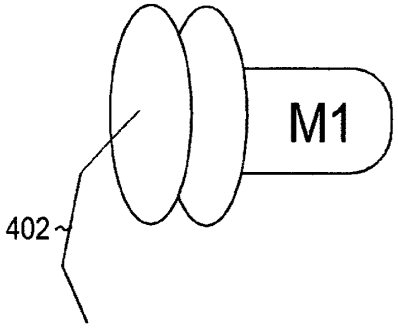


FIG 4B



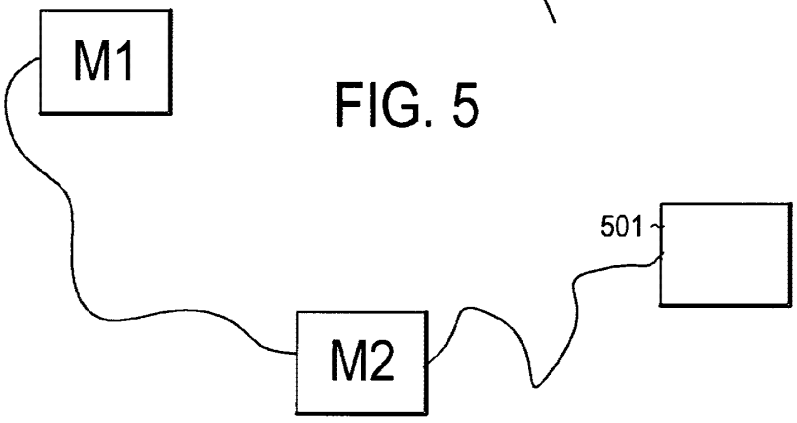


FIG. 5

IPR2024-01015 Page 00522

FIG 6

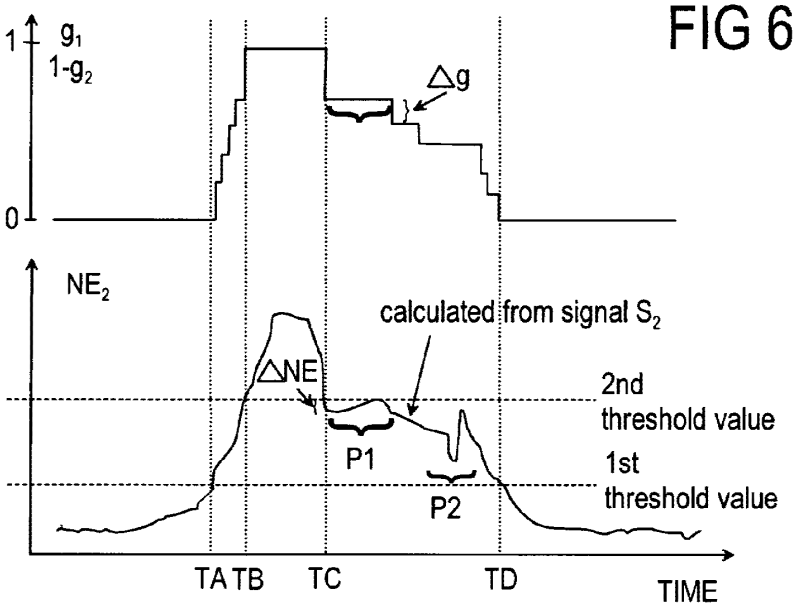
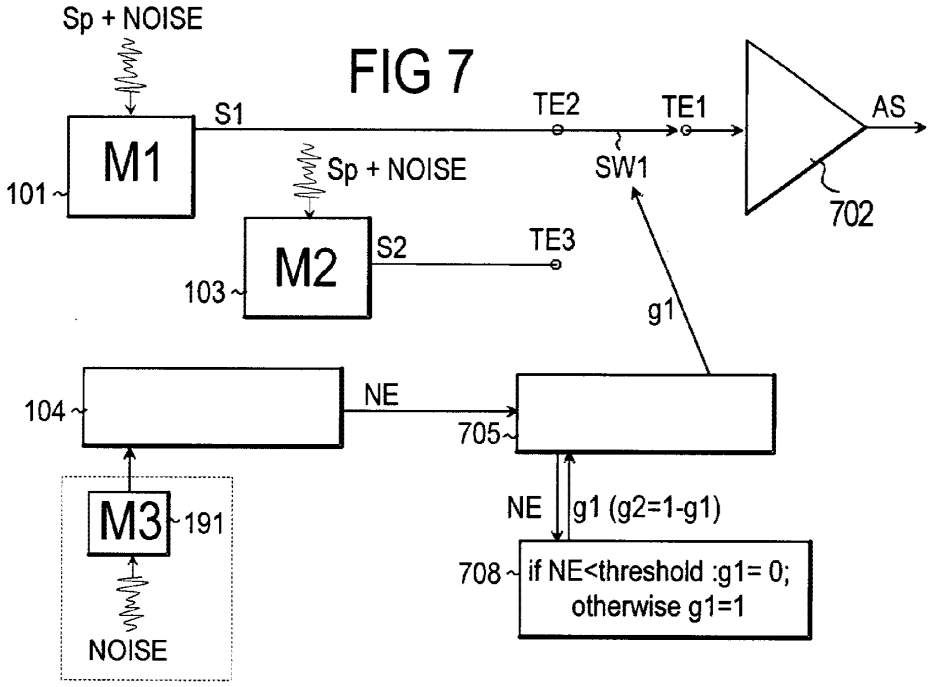


FIG 7



IPR2024-01015 Page 00523

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 01 6435

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
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TECHNICAL FIELDS SEARCHED (Int.Cl.7)

H04M

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	2 December 2003	Montalbano, F

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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IPR2024-01015 Page 00524

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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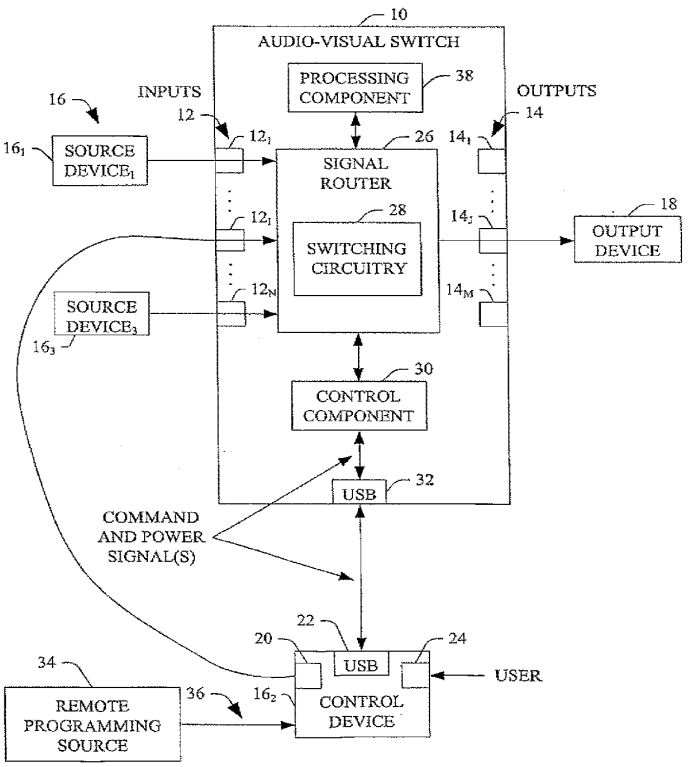
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(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LV, LY, MA, MD, 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.

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[Continued on next page]

(54) Title: USB-ENABLED AUDIO-VIDEO SWITCH



(57) Abstract: A telemedicine system (8) includes an audio-video switch (10) that receives at least two different signals and conveys at least one of the at least two different signals to an output device (18) that presents it to a user. A control device (162) monitors a position of the audio-video switch (10) and automatically controls the position to select which of the at least two different signals is passed through the audio-video switch (10) to the output device (18) in response to a signal from a remote source (34). Signals received from the remote source (34) include medically related audio video message signals, which are input to the audio-video switch (10), and a control signal that causes the audio-video switch (10) to select between passing the medically related audio-video message signals and other received signals to the output device (18).

WO 2007/092660 A1

FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, 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).

Declaration under Rule 4.17:

— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))*

Published:

— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

USB-ENABLED AUDIO-VIDEO SWITCH

DESCRIPTION

The following relates to electrical signal switching devices. It finds particular application to audio- video signal switching devices. More particularly, it is directed towards a controllable audio-video switch preferably via a Universal Serial Bus (USB) connection. However, other types of connections are also contemplated.

Conventional telemedicine includes the use of modern telecommunication and information technologies for the provision of clinical care to individuals located at a distance (e.g., at home, a nursing home, a retirement community, etc.) and to the transmission of information (e.g., messages, instructions, reminders, etc.) to provide that care. An exemplary telemedicine interaction may involve a store-and-forward form of telemedicine, a live form of telemedicine, and/or other forms of telemedicine. Store-and-forward telemedicine involves acquiring programming, data, images, video and/or audio and transmitting this information to an indiviudaul (e.g., a patient or medical specialist); it does not require the presence of the communicating parties at the same time and the bandwidth of the connection need not be high since latency is not an issue. Live telemedicine can be through television (e.g., via cable, satellite, etc.), computer (e.g., via dial-up, DSL (Digital Subscriber Line), cable modem, satellite, ISDN (Integrated Services Digital Network)), telephone, mobile telephone, etc. The live form of telemedicine typically requires the presence of the communicating parties at the same time and a high-bandwidth, low-latency connection for audio and/or video signals as well as tactile support.

One such telemedicine system is the Motiva™ telemedicine system. This system is primarily used by the elderly (however, it is not limited thereto), many of whom have limited ability in the field of interconnecting audio and/or video devices. In addition, physicians and/or other health care providers cannot be certain, with conventional telemedicine systems, whether audio and/or video information is reaching patients in a timely manner. In view of the above, there is an unresolved need for improved techniques for the delivery audio and/or video information within a telemedicine system.

In one aspect, a telemedicine system is illustrated. The telemedicine system includes an audio-video switch that receives at least two different signals and conveys at least one of the at least two different signals to an output device that presents it to a user.

A control device monitors a position of the audio-video switch automatically controls the position to select which of the at least two different signals is passed through the audio-video switch to the output device in response to a signal from a remote source. The signals received from the remote source include medically related audio video message signals, which are input to the audio-video switch, and a control signal which causes the audio-video switch to select between passing the medically related audio-video message signals and other received signals to the output device.

One advantage includes automatic switching of an audio-video switch to display messages to a user.

Another advantage resides in external control of the audio-video switch.

Another advantage resides in a simplified interconnection of the audio-video switch with power, display, and other electronic devices.

Still further advantages will become apparent to those of ordinary skill in the art upon reading and understanding the detailed description of the preferred embodiments.

The drawings are only for purposes of illustrating embodiments and are not to be construed as limiting the claims.

FIGURE 1 illustrates an exemplary audio-video switch.

FIGURE 2 illustrates an embodiment in which the AV switch includes input and output mechanisms that facilitate controlling the signal(s) that is passed to an output device.

FIGURE 3 illustrates an embodiment in which video and/or audio signals received by two or more inputs are provided to the output device.

FIGURE 1 illustrates a system 8 having an exemplary audio-video (AV) switch 10. The AV switch 10 includes N inputs 12_1, ..., 12_I, ..., 12_N (collectively referred to herein as inputs 12) and M outputs 14_1, ..., 14_J, ..., 14_M (collectively referred to herein as outputs 14), wherein I, N, J, and M are integers. Each of the inputs 12 is associated with one of composite video, s-video, RGB, component video or the like and includes corresponding connectors (e.g., BNC, RCA, s-video, wireless, etc.), which provide interfaces to one or

more of suitable source devices 16. Each of the outputs 14 is also associated with one of composite video, s-video, RGB, component video or the like and includes corresponding connectors (e.g., BNC, RCA, s-video, wireless, etc.), which provide interfaces for a suitable output device 18.

Three source devices 16_1, 16_2, and 16_3 are illustrated; however, in other instances, more (e.g., up to N) or less source devices 16 may be connected to the AV switch 10. Examples of suitable source devices 16 include, but are not limited to, a Motiva™ or other telemedicine receiver, a cable television receiver, a satellite receiver, a DVD player, a VCR player, a CD player, a computer, a digital camera, a digital video recorder, a control device, and the like. The output device 18 is connected to the AV switch 10 via the output 14_J. However, in other instances the output device 18 can be connected to one of the other outputs 14 of the AV switch 10. The particular output 14 used depends on the type of connection (e.g., composite video, s-video, RGB, component video, etc.). Suitable output devices 18 include, but are not limited to, a television, a computer, a cell phone, a personal data assistant (PDA), a stereo system, a radio, and the like.

In this example, the source device 16_2 is a control device such as a telemedicine receiver, a computer, a cable box, etc. The control device 16_2 receives video and/or audio signals (e.g., messages, programs, warnings, instructions, movies, etc.) from a remote source via cable, satellite, phone line, RF signals or the like and with a transmitter 20 transmits the signals to the input 12_1 of the AV switch 10. Control and/or power signals are also to the AV switch 10, for example, through a communications port such as a USB port 22 or the like, as described in detail below. The video and/or audio signals may be routed through the AV switch 10 and presented to the user via the output device 18. In some instances, only video is provided (e.g., textual messages). In other instances, only audio is provided (e.g., vocal recordings). In still other instances, a combination of video and audio (audio/video) are provided (e.g., a video recording with supporting sound). Typical audio and/or video input might include an instructional or motivational video for dealing with a medical condition, a reminder to take medications, a request to monitor physiological conditions, etc.

The control signal may include a command that controls which input signal(s) received at the inputs 12 passes through the AV switch 10 to the output device 18. The power signal powers the AV switch 10. A user can communicate with the control device

16_2 through a communication interface 24, which can include a TV remote, an alpha-numeric panel, a touch-screen display, a mouse, a roller ball, a digital pen, a microphone, various sensors (e.g., RF, IR, optical, etc.), and/or other input devices. Optionally, user communications signals can be received by the AV switch 10 and communicated to the control device, e.g., via the USB port. The other source devices 16_1 and 16_3 provide video and/or audio signals to the AV switch 10 via the inputs 12_1 and 12_N, respectively.

The signals received by the AV switch 10 (from the control device 16_2, the source device 16_1, the source device 16_N, etc.) are conveyed to a signal router 26 through various mediums (e.g., wire, conductive traces, wirelessly, optically, etc.). The signal router 26 includes video and audio switching circuitry 28 that selectively passes one or more of the received video and/or audio signals to the output device 18 based on a control command from a user or the control device 16_2.

A control component 30 controls the signal router 26. In one instance, the control device 16_2 sends control commands to the AV switch 10 that control the control component 30. As depicted, the control device 16_2 can communicate (uni or bi-directionally) with the control component 30 through a Universal Serial Bus (USB) port. For example, the USB port 22 of the control device 16_2 can be connected to a USB port 32 of the AV switch 10 through suitable media (e.g., a USB cable, wirelessly, etc.). In other instances, other ports such as serial, parallel, infrared, optical, radio frequency, etc. are additionally or alternatively used for (uni or bi-directional) communication. Through the port 22, the control device 16_2 can monitor the position of the AV switch 10, take over control of the AV switch 10, and automatically control it to ensure video and/or audio from the control device 16_2 is provided to the output device 18 for presentation to the user.

Alternately, the AV switch 10 can be controlled to provide programming from a different input. For example, received programming or other audio/video signals can be stored in a (local or remote) recorder source device and played back when the output device 18 is turned on, at a selected time, on-demand, or the like. Recording and later playback facilitates converting low bandwidth signals from the remote source 34 into high quality displays or other outputs and/or reducing signal size (e.g., via compression) prior to storing and/or playing a recorded audio-video signal. In addition, the control device 16_2 can also use the USB port 20 (and/or various other ports) to read information (e.g., a unique ID, a serial number, a software/firmware version, error logs, etc.) from the AV

switch, write information (e.g., software/firmware updates, applications, etc.) to the AV switch 10, and execute routines (e.g., diagnostics, etc.) within the AV switch 10.

The control commands sent by the control device 16_2 may originate with a health care provider/facility or other remote programming source 34 transmitting information to the control device 16_2 (e.g., via cable, satellite, a LAN line 36, etc.). In one instance, the information transmitted to the control device 16_2 includes the control command. For example, the control device 16_2 may receive a control command that invokes the signal router 26 to pass the data (audio and/or video information) received at the input connected to the control device 16_2 through the AV switch 10 to the output device 18 through the output 14_J. In another example, the control device 16_2 may receive a control command that invokes the signal router 26 to pass audio and/or video (e.g., an instructional video) received from a different one of the source devices 12 (e.g., a community health channel provided through a cable box) to the output device 18.

In another instance, the information transmitted to the control device 16_2 by the source 34 or other source includes one or more messages. Examples of suitable messages include, but are not limited to, at least one of instructional programming relating to a disease or physiological condition, reminders to take medicines, reminders of medical appointments, requests to monitor physiological conditions, motivational programming, requests to call a doctor's office, instructions associated with a scheduled medical procedure, test results, and real-time communications from medical personnel. Reception of such message may invoke the control device 16_2 to send the control command (over the USB connection) to the AV switch 10 to control the signal router 26 accordingly. In addition, the received message(s) is sent to the AV switch by the control device 16_2 and routed through the signal router 26 to the output device 18.

The control command may also originate from the user. For example, the user may manually enter a command at the control device 16_2 through an interface 24. In another example, the user may use a remote control to send control commands to the control device 16_2 via the interface 24. This enables the user to select among source devices, play or replay the information from the remote source, and the like.

In one instance, a security mechanism is provided in which a party (e.g., the user, a health care provider, a third party such as family, etc.) attempting to communicate with (e.g., control, convey audio-video to, convey messages to, etc.) the control device 16_2

and/or the AV switch 10 is validated prior to allowing such communication. For instance, the control device 16_2 and/or the AV switch 10 may be programmed to only accept signals from a predetermined set of authorized health care facilities and/or individuals and/or other entities. A particular party desiring to communicate with (e.g., control, etc.) the control device 16_2 and/or the AV switch 10 may first have to provide identification indicia, an authorization code, a password, and the like that is validated by the control device 16_2 and/or the AV switch 10 before any such communication is allowed. Such validation could include comparing the received identification indicia, authorization code, password, etc. with stored authorized identification indicia, authorization code, password, etc.

The party may attempt to communicate with the control device 16_2 and/or the AV switch 10 through local controls (e.g., knobs, buttons, touch screen, etc. residing on the control device 16_2 and/or the AV switch 10), a remote control (e.g., a universal wireless remote programmed to communicate with the control device 16_2 and/or the AV switch 10), voice activated controls (e.g., executing voice recognition software in connection with a microphone), the Internet, the source 34, etc. In one instance, communication from an unauthorized party is ignored. In this instance, information about the unauthorized party (e.g., obtained from the attempted communication) may be stored and/or sent to security and/or administrative authority for analysis. The predetermined set of authorized entities and information used for validation may be updated by an authorized entity such as the user adding a party (e.g., via a control panel, a wireless remote, voice controls, etc.), an authorized health care facility/provider writing to or overwriting (e.g., via downloading) previously stored authorization information via the source 34, etc.

Each received message may be associated with tag that indicates a priority of the corresponding message. In this instance, the control device 16_2 can ascertain the relative importance of the message via the tag to determine whether the message should be presented to the user via the output device 18 or a notification (e.g.,. via the output device 18 or other visual and/or audio components) that a message has arrived, or whether the message should be stored for later retrieval without currently interrupting the user. Messages provided by health care providers/facilities can be associated with a relatively high priority in order to allow the control device 16_2 to take control over the AV switch 10, overriding the user's source selection, and set the switch to display incoming messages from the control device 16_2 to the user at the output device 18.

As briefly stated above, one or more audio and/or video signals received by the inputs 12 can be provided to the output device 18. For instance, the control component 30 can control the signal router 26 such that an audio and/or video signal conveyed from only one of the inputs 12 is passed through the AV switch 10 to the display device 18. In this instance, the user may initially select the audio and/or video signal that passes to the output device 18. However, when the control device 16_2 transmits information to the AV switch 10, the control device 16_2 may also issue a control command to the control component 30, instructing the control component 30 to switch the AV switch 10 such that the audio and/or video signals provided by the control device 16_2 are fed to the output device 18. The control component 30 can switch the AV switch 10 back to the audio and/or video signal selected by the user after the user views/listens to the message. In one instance, the control device 16_2 releases control of the AV switch 10 back to the control component 30 after a predetermined lapse of time, upon receiving an acknowledgement (e.g., electronic, audible, tactile, etc.) from the user that the message has been received, etc.

In another instance, video and/or audio messages from the control device 16_2 can be concurrently presented with video and/or audio from at least one other source device 16. A processing component 38 is used to combine the signals. For example, a video message from the control device 16_2 may be superimposed as an overlay, a banner, a semi-transparent image (with a programmable and adjustable translucence), or the like over the video from one of the other source devices 16 by the processing component 38, and the superimposed video may be provided to the output device 18. The audio, if any, associated with the video from the control device 16_2 may override the audio associated with the video from the other source device 16. In this manner, the video and/or audio message from the control device 16_2 can be presented to the user along with another video signal. In other instances, the processing component 38 is not used to combine signals and the video and/or audio messages from the control device 16_2 are provided to the output device 18 instead video and/or audio from one of the other source devices 16.

The different messages transmitted by the control device 16_2 may be associated with a similar or different priority levels such that some messages (e.g., an urgent announcement, etc.) are associated with relatively high priority levels that secure a transmission path dedicated for displaying messages at the output device 18, some messages (e.g., a general announcement, etc.) are associated with lower priority levels in

which superimposed signals are conveyed to the output device 18, and some messages are associated with relatively low priority levels in which the user is prompted as to whether the message should be displayed now or stored for later retrieval.

In another instance, one of the source devices 16 is a digital video recorder (DVR), a personal video recorder (PVR), or other device that allows a user to capture television or other video to a storage device for later viewing, or the like. When the user turns on the TV to watch a selected program, the control device causes a message recorded while the TV was off to be played while the selected program is recorded. After the message, the selected program is played back. As briefly discussed above, the control device 16_2 may also provide power signals to the AV switch 10 through the USB connection between the USB port 22 and the USB port 32. The power signals may be used to power at least a portion of the AV switch 10. In another instance, the AV switch is additionally or alternatively powered by a power supply (e.g., an AC to DC converter plugged into a wall mounted AV receptacle), a battery, a photocell, or the like. The control device 16_2 may also control power to the output device to turn the output device on, e.g., when a time-sensitive message is received.

FIGURE 2 illustrates an embodiment of the system 8 in which the AV switch 10 further includes input and output mechanisms that facilitate controlling the signal(s) passed through the AV switch 10 and monitoring the status of the AV switch 10. In FIGURE 2, common components with FIGURE 1 are denoted with the same reference number.

As depicted, the AV switch 10 can include a wireless communications port 40. A user of the AV switch 10 can provide control commands to the control component 30 through this port. The control component 30 interprets such commands and controls the signal router 26 accordingly. For example, the user may use a remote control or other wireless transmitter that is programmed to communicate with the port 40 of the AV switch 10. The user can use the remote control to send various control commands such as a command that changes the input 12 that is fed to the output 14_j connected to the output device 18. The communications port 40 and the remote control can include various communication technologies such as radio frequency (RF), infrared (IR), optical, etc. for communication.

The AV 10 may additionally or alternatively include a control panel 42 with which the user can manually communicate with the AV switch 10. For instance, the user can

enter a control command through the control panel 42 to select the input 12 whose signal will pass through the AV switch 10. The control panel 42 may be associated with various input/output mechanisms such as an alpha-numeric panel, a touch-screen, a mouse, a roller ball, a touch pad, a digital pen, a microphone to receive audible instructions, various sensors (e.g., RF, IR, optical, pressure, etc.), and/or other input devices.

The AV 10 may additionally or alternatively include a presentation component 44, which provides visual and/or audible feedback. For example, the presentation component 44 can include a plurality of light emitting diodes (LEDs) and/or seven segment displays that indicate one or more of whether the AV switch 10 is receiving power, the type of power (e.g., converted AC, battery or photocell), whether data is being received at each of the inputs 12, whether data is being conveyed through each of the outputs 14, error codes, whether the AV switch 10 is being controlled by the control device 16_2, whether a message is being broadcast through the output device 18, etc. The presentation component 44 can additionally or alternatively include a flat panel display, a liquid crystal display (LCD), and the like to provide graphical and textual information, a speaker to provide audible information, etc.

FIGURE 3 illustrates an embodiment of the system 8 in which video and/or audio signals received by two or more of the inputs 12 and passed to two or more of the outputs 14 are provided to the output device 18. For explanatory purposes, the inputs 12_1, 12_I and 12_N are illustrated as receiving audio and/or video signals. The output device 18 may include picture-in-picture or similar technology in which the display area of the output device 18 can be delineated into two or more independent regions, each that can present different video. For explanatory purposes, only two of the outputs 14_1 and 14_J provide similar or different video and/or audio signals to the output device 18.

The control component 30 can control the signal router 26 to pass one audio and/or video signal to one of the outputs 14, a different audio and/or video signal to a different one of the outputs 14, etc. Typically, one of the viewing regions of the output device 18 is designated as a primary region and audio associated with this region is provided to the user. The other region(s) are deemed secondary and typically present video without the audio. With conventional picture-in-picture technology, the user can change which region is the current primary region. Some systems allow the user to select the audio output such

that audio from a secondary viewing region is broadcast. Both of these changes can be achieved via the AV switch 10 through the communications interfaces 40 and/or 42.

The control device 16_2 can also change which audio and/or video signal is conveyed to the primary region and which audio and/or video signals are conveyed to the other regions of the output device 18. In one instance, the change is initiated by the reception of a message, which is subsequently provided to the user, by the control device 16_2. For instance, a health care provider/facility may send a message to the user that informs the user that it is time to take a medication. Upon receiving such message, the control device 16_2 may provide the message (e.g., which may include audio and/or video) to the input 12_1 of the AV switch 10. Concurrently, another source device 16 may be providing audio and/or video to another of the inputs 12. The control device 16_2 may send a control command to the AV switch 10 via the USB connection between the USB port 22 and the USB port 32. The control command is conveyed to the control component 30, which controls the signal router 26 to pass the message to the output 14 that feeds either the primary region or one of the secondary regions of the output device 18. The other audio and/or video signal can be displayed in another viewing region.

In instances where the message is deemed critical, the message may be sent to the output 14 feeding the primary region and the picture-in-picture feature can be either disabled or the signal feeding the display regions can be temporarily terminated, or the message may be sent to the outputs feeding the primary and the secondary regions of the output device 18 such that all display regions present the message. In instances where the message is deemed less important, the message may be sent to the output 14 feeding the primary region or an output 14 feeding a secondary region of the output device 18. For example, the selected programming can be switched to the secondary region which the message is displayed in the primary region. After the message is displayed, or, if the message calls for user action such as taking medication, after the signals that the action has been completed, the selected programming is returned to the primary region. In instances where picture-in-picture is not enabled, the control component 30 can enable it to present the message within the primary or a secondary region.

In another instance, the user initiates the change in primary region through the communications port 24 and/or manual controls such as buttons, switches, knobs, etc. residing on the control device 16_2.

The invention has been described with reference to the preferred embodiments. Modifications and alterations may occur to others upon reading and understanding the preceding detailed description. It is intended that the invention be constructed as including all such modifications and alterations insofar as they come within the scope of the appended claims or the equivalents thereof.

IPR2024-01015 Page 00538

CLAIMS

1. A telemedicine system (8), comprising:

an audio-video switch (10) that receives at least two different signals and conveys at least one of the at least two different signals to an output device (18) that presents it to a user; and

a control device (16_2) that monitors a position of the audio-video switch (10) and automatically controls the position to select which of the at least two different signals is passed through the audio-video switch (10) to the output device (18) in response to a signal from a remote source (34).

2. The telemedicine system (8) as set forth in claim 1, wherein the control device (16_2) includes a control port (22) and the audio-video switch (10) includes a port (32) and the control device (16_2) monitors and controls the audio-video switch (10) through signals conveyed from the control port (22) to the port (32).

3. The telemedicine system (8) as set forth in claim 2, wherein the control commands determines which signal is passed through the audio-video switch (10) to the output device (18) by setting the relative position of audio-video switch (10).

4. The telemedicine system (8) as set forth in claim 2, wherein the control port (22) and the port (32) are USB ports.

5. The telemedicine system (8) as set forth in claim 2, wherein the control device (16_2) obtains at least one of a unique identification, a serial number, a software versions, and a firmware version of the audio-video switch (10) via the control port (22) and the port (32).

6. The telemedicine system (8) as set forth in claim 1, wherein the control device (16_2) powers the audio-video switch (10) through a USB connection between the control device (16_2) and the audio-video switch (10).

IPR2024-01015 Page 00539

7. The telemedicine system (8) as set forth in claim 1, wherein the control device (16_2) is one of a telemedicine receiver, set top box, and a computer.

8. The telemedicine system (8) as set forth in claim 1, wherein the output device (18) is one of a television and a computer.

9. The telemedicine system (8) as set forth in claim 1, wherein at least one of the at least two different signals is provided by the control device (16_2) and the at least one other signal is provided by one of a cable television receiver, a satellite receiver, a DVD player, a VCR player, a CD player, a computer, a digital camera, and a digital video recorder.

10. The telemedicine system (8) as set forth in claim 1, wherein the audio-video switch (10) includes a plurality of inputs (12) that each receive one of the at least two different signals and at least one output (14), each of the a plurality of inputs (12) and the at least one output (14) is associated with one of composite video, s-video, RGB, component video, and other formatted video.

11. The telemedicine system (8) as set forth in claim 10, wherein the audio-video switch (10) includes a signal router (26) having switching circuitry (28), and wherein the at least two different signals are received by the plurality of inputs (12), fed to a signal router (26), and switched by the switching circuitry (28) based on a control command from the control device (16_2).

12. The telemedicine system (8) as set forth in claim 1, wherein the audio-video switch (10) includes at least one of a wireless communications port (40) for controlling the audio-video switch (10) with a remote wireless device and control panel (42) for manual control of the audio-video switch (10).

13. The telemedicine system (8) as set forth in claim 1, wherein the audio-video switch (10) includes a processing component (38) that combines at least two input signals, the two signals being combined by one of overlaying one of the signals over the other

signal, the overlayed signals being passed to the output device (18) to display both signals concurrently.

14. The telemedicine system (8) as set forth in claim 1, wherein the control device (16_2) automatically switches the audio-video switch (10) in order to present a message to a user via the output device (18) and the audio-video switch (10) automatically switches back to an input device (16) selected by the user after the user views/listens to the message.

15. The telemedicine system (8) as set forth in claim 1, wherein at least one of the at least two different signals is associated with a tag that indicates a priority of the signal relative to other signals and the priority is used to determine at least one of when and how to present the signal to the user.

16. The telemedicine system (8) as set forth in claim 1, wherein control over the audio-video switch (10) by the control device (16_2) overrides control by a user.

17. The telemedicine system (8) as set forth in claim 1, further including:

a recording device (16) connected with the audio-video switch (10), the control device (16_2) causing the recording device to record audio/video signal and causing the recorded audio/video signal to be passed through the audio-video switch (10) to the output device (18) in accordance with instruction signals received from the remote source (34).

18 The telemedicine system (8) as set forth in claim 17, wherein the recorded audio/video signal is received over a low bandwidth connection and converted into high quality outputs.

19. The telemedicine system (8) as set forth in claim 17, wherein the recorded audio/video signal is reduced in size prior to one of storing and playing the audio/video signal.

IPR2024-01015 Page 00541

20. The telemedicine system (8) as set forth in claim 1, wherein the output device (18) concurrently displays more than one video signal, each within a different display region of the output device (18), and the an audio-video switch (10) passes at least two signals to an output device (18), wherein the control device (16_2) controls the position of the audio-video switch (10) to select which of the at least two signals is displayed in each of the displays regions in response to the signal from the remote source (34).

21. The telemedicine system (8) as set forth in claim 1, wherein the output device (18) includes an audio/visual display and the signals from the remote source (34) include audio/video signals conveying at least one of instructional programming relating to a disease or physiological condition, reminders to take medicines, reminders of medical appointments, requests to monitor physiological conditions, motivational programming, requests to call a doctor's office, instructions associated with a scheduled medical procedure, test results, and real-time communications from medical personnel, and control signals which cause the control device (16_2) to control the audio-video switch (10) to pass the audio-video signals from the remote source (34) to the audio/visual display.

22. The telemedicine system (8) as set forth in claim 1, wherein at least one of the control device (16_2) and the audio-video switch (10) validates parties attempting to communicate therewith and only authorized parties are allowed to communicate with the at least one of the control device (16_2) and the audio-video switch (10).

23. A telemedicine method comprising:

receiving at least two different signals with an audio-video switch (10) that conveys at least one of the received signals to an output device (18);

monitoring a position of the audio-video switch (10); and

controlling the audio-video switch (10) to select which of the received signals is passed to the output device (18) in accordance with signals from a remote source (34).

24. The method as set forth in claim 23, wherein the signals received from the remote source (34) includes medically related audio visual message signals which are input

IPR2024-01015 Page 00542

to the audio-video switch (10) and a control signal which causes the audio-video switch (10) to select between passing the medically related audio-visual message signals and other received signals to the output device (18).

25. The method as set forth in claim 23, wherein the monitoring and controlling are performed by a control device (16_2) and further comprising:

conveying electrical power from the control device (16_2) to the audio-video switch (10) over USB ports (22, 32); and

powering the audio-video switch (10) with the electrical power.

26. An audio-video switch (10), comprising:

a plurality of inputs (12) for receiving signals from a plurality of source devices (16);

at least one output (14) for conveying at least one of the received signals to an output device (18); and

a USB port (32) for receiving switch setting control commands from a control component (30) to set the audio-video switch (10) to pass at least one of the received signals to the output device (18) based on the switch setting control commands, and the USB port (32), preferably, receiving electrical power from the control component (30).

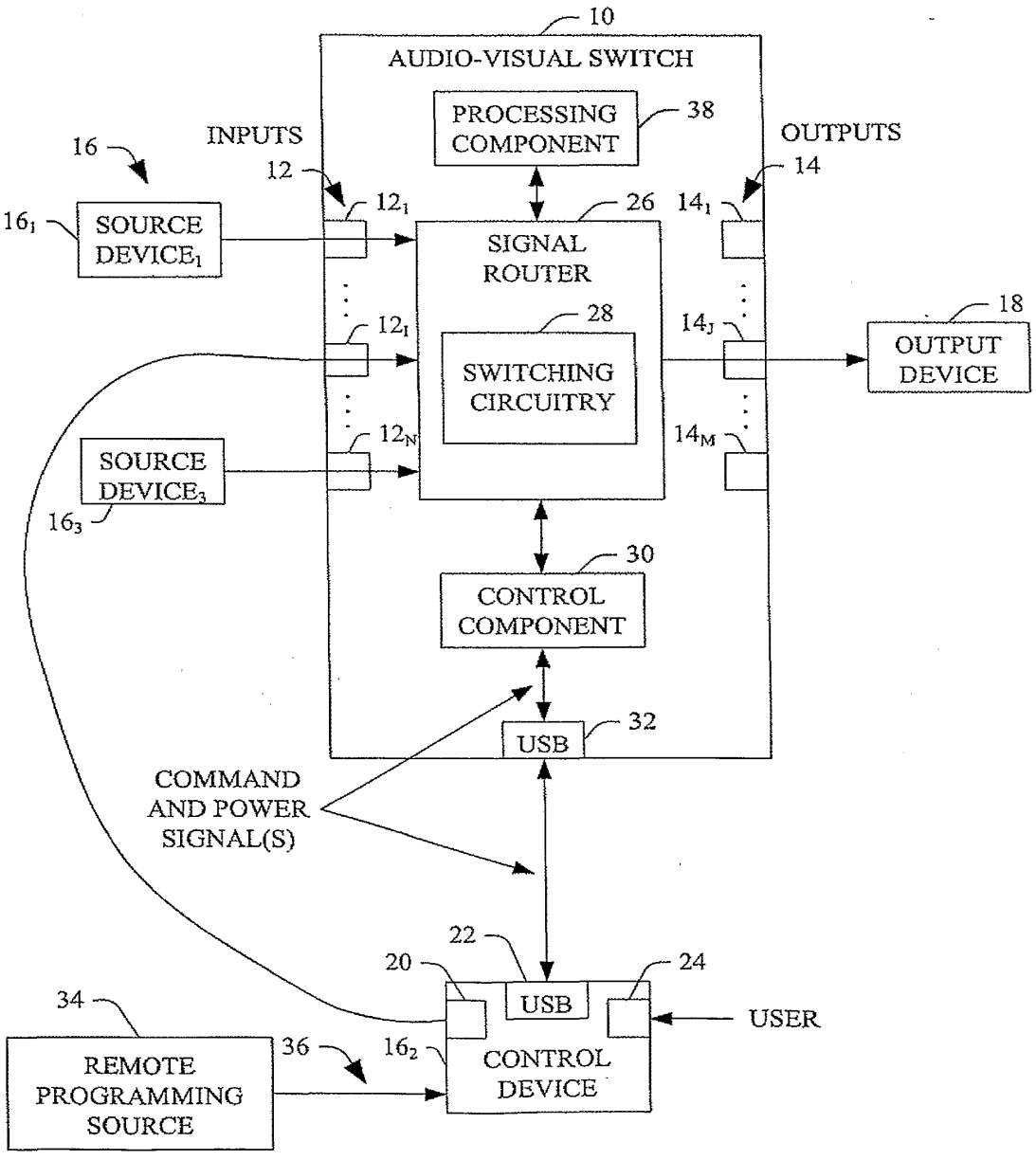

10
AUDIO-VISUAL SWITCH
PROCESSING COMPONENT
38
INPUTS
12
OUTPUTS
14
16
16₁
SOURCE DEVICE₁
12₁
26
14₁
SIGNAL ROUTER
12ᵢ
28
14ⱼ
18
OUTPUT DEVICE
SWITCHING CIRCUITRY
12ₙ
14ₘ
SOURCE DEVICE₃
16₃
30
CONTROL COMPONENT
32
USB
COMMAND AND POWER SIGNAL(S)
22
20
24
USB
USER
34
36
16₂
REMOTE PROGRAMMING SOURCE
CONTROL DEVICE

FIGURE 1

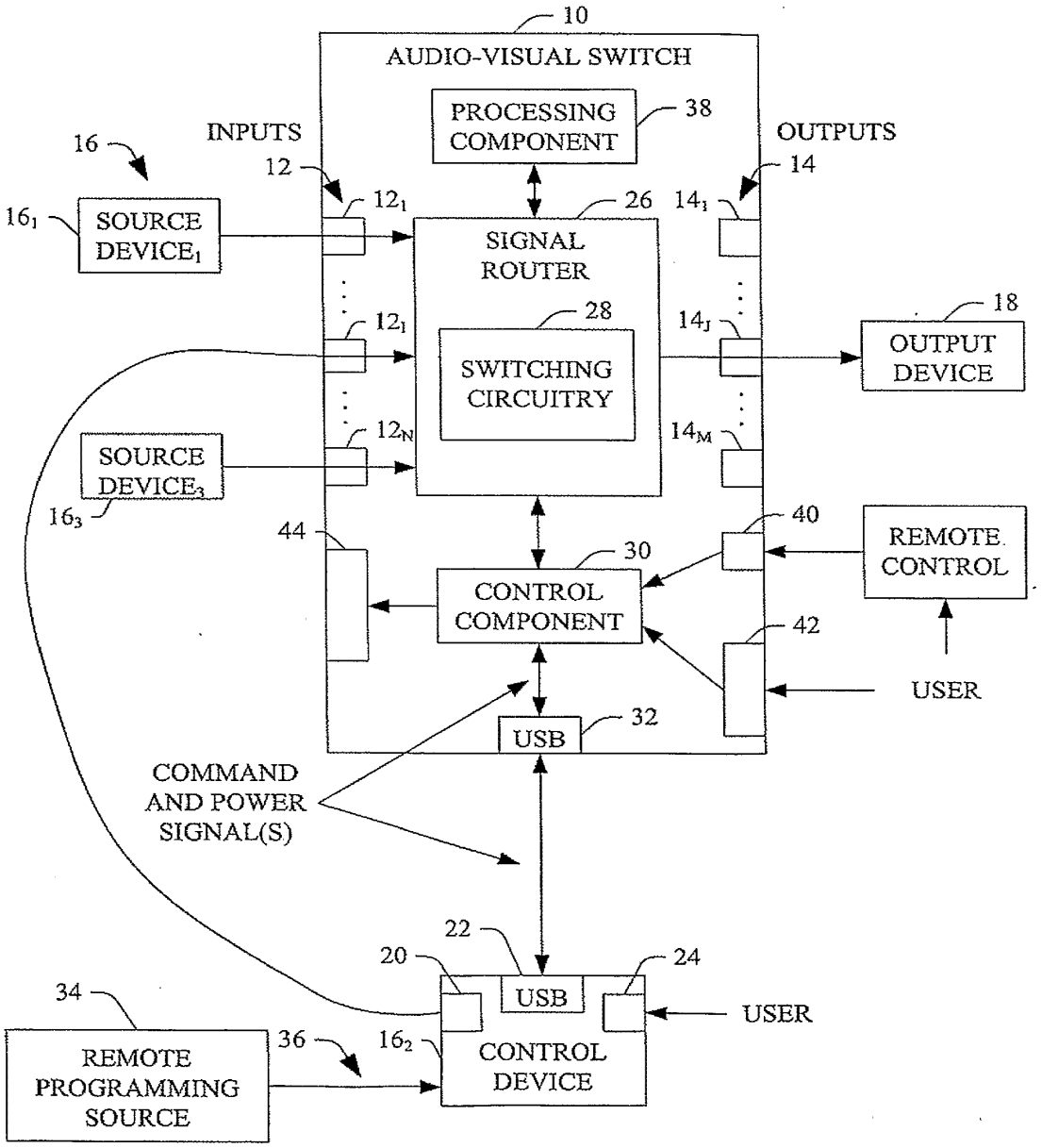

10
AUDIO-VISUAL SWITCH
PROCESSING COMPONENT
38
INPUTS
12
16
OUTPUTS
14
16₁
SOURCE DEVICE₁
12₁
26
14₁
SIGNAL ROUTER
28
12ᵢ
14ⱼ
18
OUTPUT DEVICE
SWITCHING CIRCUITRY
12N
14M
SOURCE DEVICE₃
16₃
44
40
REMOTE CONTROL
30
CONTROL COMPONENT
42
USER
USB
32
COMMAND AND POWER SIGNAL(S)
22
20
24
USB
USER
34
36
16₂
REMOTE PROGRAMMING SOURCE
CONTROL DEVICE

FIGURE 2

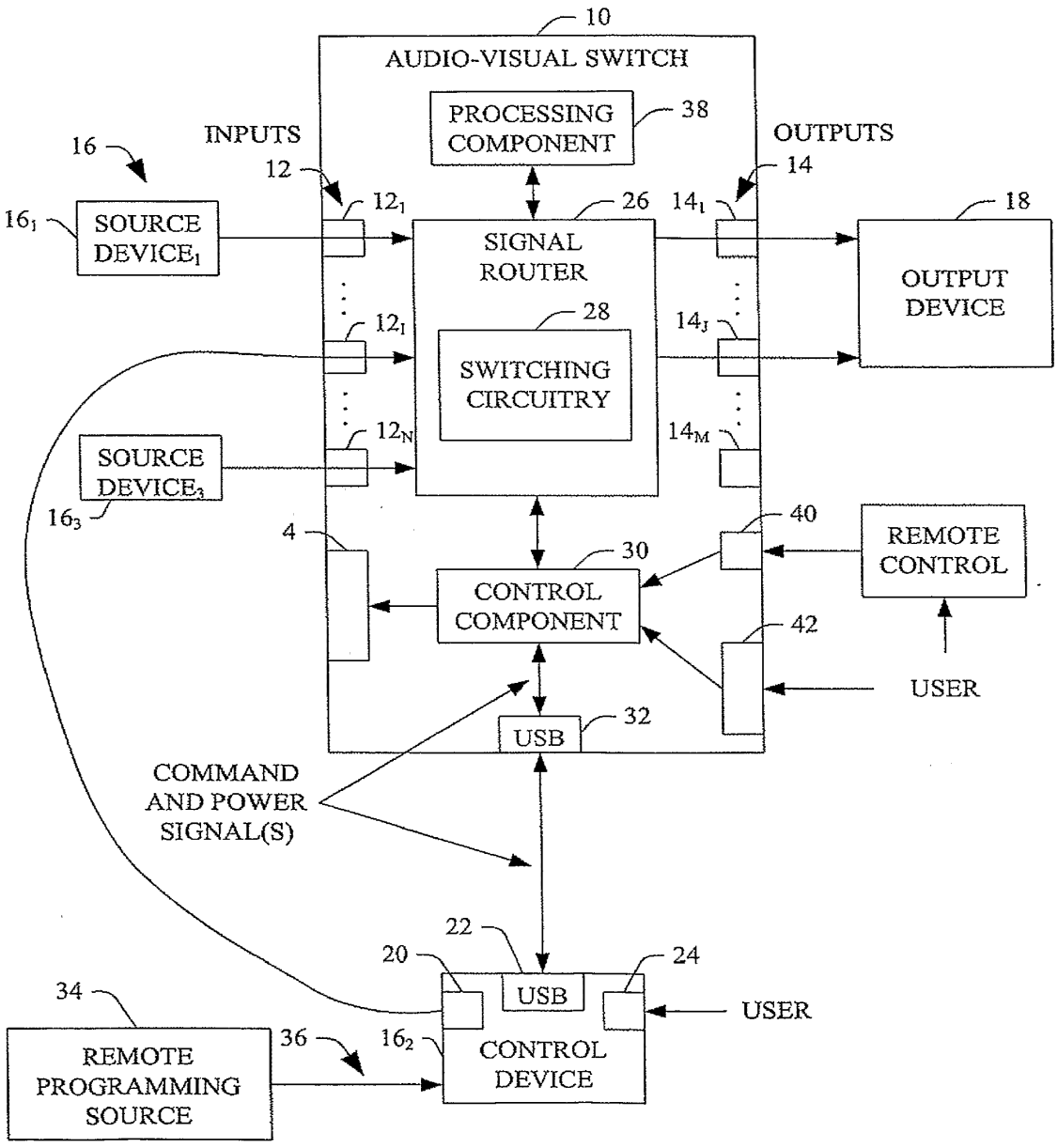

10
AUDIO-VISUAL SWITCH
PROCESSING COMPONENT
38
INPUTS
12
OUTPUTS
14
16
16₁
SOURCE DEVICE₁
12₁
26
14₁
18
SIGNAL ROUTER
OUTPUT DEVICE
28
12ᵢ
14ⱼ
SWITCHING CIRCUITRY
12ₙ
14ₘ
SOURCE DEVICE₃
16₃
4
40
REMOTE CONTROL
30
CONTROL COMPONENT
42
USER
USB
32
COMMAND AND POWER SIGNAL(S)
22
20
24
USB
USER
34
36
16₂
REMOTE PROGRAMMING SOURCE
CONTROL DEVICE

FIGURE 3

INTERNATIONAL SEARCH REPORT

International application No
PCT/US2007/060620

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04N7/18 G06F19/00

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)
H04N G06F

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	EP 1 126 715 A2 (KATZ RONALD ALAN [US] TELEBUYER LLC [US]) 22 August 2001 (2001-08-22) page 3, paragraph 13 page 9, paragraph 70 page 16, paragraph 134 - page 17, paragraph 151; figures 1-5	1-26
X	US 2005/052527 A1 (REMY CHRISTOPHE [US] ET AL) 10 March 2005 (2005-03-10) page 3, paragraph 30 - paragraph 32; figure 3 page 5, paragraph 41 - paragraph 42; figures 7,8	1-26

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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
3 April 2007	13/04/2007

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, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Schinnerl, Arnold

Form PCT/ISA/210 (second sheet) (April 2005)

INTERNATIONAL SEARCH REPORT

International application No
PCT/US2007/060620

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 6 717 609 B2 (SUGIMOTO HIDEO [JP] ET AL) 6 April 2004 (2004-04-06) column 7, line 19 - column 8, line 42 abstract; figures 1-5	1-26
X	US 5 553 609 A (CHEN YAOBIN [US] ET AL) 10 September 1996 (1996-09-10) column 9, line 57 - column 12, line 53; figures 5-9	1,23,26
A	EP 1 343 326 A1 (MACROSYSTEM DIGITAL VIDEO AG [DE]) 10 September 2003 (2003-09-10) abstract; figure 1 column 6, line 22 - line 58	1-26

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/US2007/060620

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 1126715 A2	22-08-2001	NONE	
US 2005052527 A1	10-03-2005	US 2006119701 A1	08-06-2006
US 6717609 B2	06-04-2004	DE 10101064 A1	19-07-2001
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		DE 50200955 D1	14-10-2004
		US 2003193577 A1	16-10-2003

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.

[0001] BACKGROUND OF THE INVENTION 1. Field of the Invention The present invention relates to a self-speech detection device, a voice input device, and a hearing aid having a function of distinguishing a voice uttered by a wearer from a voice coming from the outside world.

[0002] 2. Description of the Related Art Conventionally, a device using a bone conduction microphone has been put into practical use as a device for separating and identifying self-utterance, which is a voice uttered by a wearer, from an outside voice.

[0003] The bone conduction microphone detects the vibration of the skull caused by the wearer's voice through the attachment part that is in close contact with the wearer's ear canal.There is no sensitivity to external noise, which is air vibration, and the wearer's self-speech Only the sound can be picked up.

[0004] However, the bone conduction microphone is required to have close contact between the mounting portion and the ear canal in order to pick up the self-speech sound with high sensitivity. For this reason, it is necessary to match the shape of the wearing part with the shape of the ear canal of each wearer, and in the case of a cell type body in which the main body is integrated with the wearing part, it is necessary to replace the whole cell, so each wearer There was a problem that the fitting of was complicated.

[0005] An object of the present invention is to provide a self-speech detection device, a voice input device, and a hearing aid that can detect self-speech with a simpler configuration.

[0006] According to the invention of claim 1 of the present application, a mounting portion that is inserted into the ear canal of a wearer to insulate between the outside world and the ear canal, and a first microphone provided toward the outside world. A second microphone provided toward the ear canal; a delay unit that delays the signal of the first microphone for a predetermined time; a receiver that outputs the signal delayed by the delay unit into the ear canal; and a first microphone. And a signal processing unit that detects the self-utterance of the wearer based on the correlation between the signal waveform of (1) and the signal waveform of the second microphone.

[0007] In the invention of claim 2 of this application, in the invention of claim 1, the signal processing unit calculates a difference between the signal waveform of the first microphone and the signal waveform of the second microphone, and It is a processing unit that detects the wearer's self-utterance based on the correlation with the signal waveform of the microphone 1.

[0008] According to a third aspect of the present application, in the first aspect of the invention, the signal processing unit propagates the signal waveform of the first microphone delayed by the delay means including the ear canal from the receiver to the second microphone. The convolution calculation is performed by the transfer function of the road, the difference between the convolved signal waveform and the signal waveform of the second microphone is calculated, and the wearer's self is calculated based on the correlation between the difference signal waveform and the signal waveform of the first microphone. It is characterized in that it is a processing unit for detecting an utterance.

[0009] According to the invention of claim 4 of this application, in the invention of claim 3, a sound source for generating a test sound signal and inputting it to the delay means, a direct waveform of the

test sound signal and a second microphone output from the receiver are used. It is characterized in that it further comprises means for calculating a transfer function of the propagation path based on the propagation waveform of the received test sound signal.

[0010] According to the invention of claim 5 of this application, a mounting portion that is inserted into the ear canal of the wearer to insulate between the outside world and the ear canal, the first microphone provided toward the outside world, and the inside of the ear canal are provided. A second microphone, a delay unit that delays the signal received by the first microphone for a predetermined time, a receiver that outputs the signal delayed by the delay unit into the ear canal, and a first microphone that is delayed by the delay unit. The signal waveform is convolved with the transfer function of the propagation path including the ear canal from the receiver to the second microphone, the difference signal waveform between this convolutional signal waveform and the signal waveform of the second microphone is extracted, and this difference signal waveform is And a signal processing unit for external output.

[0011] According to the invention of claim 6 of this application, in the invention of claim 5, the sound source for generating a test sound signal and inputting it to the delay means, the direct waveform of the test sound signal and the second microphone output from the receiver are used. It is characterized in that it further comprises means for calculating a transfer function of the propagation path based on the propagation waveform of the received test sound signal.

[0012] According to the invention of claim 7 of this application, a mounting portion that is inserted into the ear canal of the wearer to insulate between the outside world and the ear canal, a first microphone provided toward the outside world, and the inside of the ear canal are provided. The second microphone, the voice speed conversion means for extending the signal received by the first microphone on the time axis, the receiver for outputting the signal delayed by the delay means into the ear canal, and the voice speed conversion means. The signal waveform of the first microphone is convolved with the transfer function of the propagation path including the ear canal from the receiver to the second microphone, and the difference signal waveform between this convolutional signal waveform and the signal waveform of the second microphone and the first And a signal processing unit that detects the self-utterance of the wearer based on the correlation with the signal waveform of the microphone and prohibits the operation of the speech speed conversion means when the self-utterance is detected.

[0013] According to the invention of claim 8 of this application, in the invention of claim 7, a sound source for generating a test sound signal and inputting it to the delay means, a direct waveform of the test sound signal and a second microphone output from the receiver are used. It is characterized in that it further comprises means for calculating a transfer function of the propagation path based on the propagation waveform of the received test sound signal.

[0014] In the present invention, the second microphone receives not only the self-speech sound but also the external voice output from the receiver. On the other hand, the first microphone receives the foreign voice. By canceling the component of the external voice by correlating the signal waveform of the second microphone and the first signal waveform, only the self-speech sound component can be separated and extracted. The foreign voice received by the first microphone includes the self-speech sound uttered by the mouth, but the same processing may be performed.

[0015] Since the signal processing unit can be configured by a DSP or the like, the structure can be simplified even when performing complicated signal processing.

[0016] Therefore, according to the present invention, it is possible to separate and extract only the self-speech sound with a configuration of the mounting portion that is much simpler than that of the bone conduction microphone that only shields the outside world from the ear canal.

[0017] BEST MODE FOR CARRYING OUT THE INVENTION Embodiments of the present invention will be described with reference to the drawings. FIG. 1 is an external view of a self-speech detection device to which the present invention is applied. This self-speech detection device is used for a hearing aid and a voice input device. An earplug 2 which is a mounting portion to be inserted into the ear canal 4 is provided at the tip of the cell 1. The earplug 2 is made of an umbrella-shaped flexible resin, and is in close contact with the external auditory meatus wall 3 to block the external auditory meatus 4 from the outside world and prohibit the propagation of sound between them.

[0018] In the cell 1, a microphone 5 for detecting a sound coming from the outside world, a receiver 6 for outputting a sound to the ear canal 4, a microphone 7 for detecting a sound inside the ear canal 4, and a receiver 6, A pipe hole 8 that spatially connects with the microphone 7 is provided. The microphone 5, the receiver 6 and the microphone 7 are connected to the signal processing unit 9. The signal processing unit 9 is composed of an electronic circuit such as a DSP, analyzes and processes the audio signal input from the microphone 5 and the microphone 7, and outputs the processed audio signal from the receiver 6. Hereinafter, examples of various configurations of the signal processing unit 9 and their operations will be described.

[0019] FIG. 2 shows an example of a self-speech detection device in which the signal processing unit 9 is composed of a delay device 10, a correlation calculation device 11, and a speech determination device 12. In FIG. 2A, the external voice received by the microphone 5 is input to the delay device 10 and the correlation calculation device 11. The delay device 10 delays the input audio signal by a minute time t and outputs it to the receiver 6. The receiver 6 converts this voice signal into voice (air vibration) and emits it into the ear canal 4. The microphone 7 receives both the voice output by the receiver 6 and the voice of the wearer transmitted from the ear canal 4. The audio signal received by the microphone 7 is input to the correlation calculation device 11. The correlation calculation device 11 calculates a correlation value between the audio signal input from the microphone 5 and the audio signal input from the microphone 7.

[0020] With reference to FIGS. 2B and 2C, the signal waveform and the correlation when the external voice arrives will be described. The time when the microphone 5 receives the foreign voice is 0 sec. The input point A waveform of the microphone 5 has a waveform as shown in the upper part of FIG. The delay device 10 delays this audio signal for tsec and outputs it to the receiver 6. The pointB waveform which is the waveform supplied to the receiver 6 is a waveform as shown in the middle part of FIG. 7B in which the waveform of pointA is delayed by tsec. The receiver 6 outputs the delayed audio signal to the ear canal 4. The sound output to the ear canal 4 is also received by the microphone 7. The pointC waveform which is the input of the microphone 7 has a waveform similar to the pointB waveform as shown in the lower part of FIG. The correlation calculation device 11 is a device that in time series examines how similar the point A waveform is to the reference waveform, using the point A waveform as the reference waveform, and outputs a correlation value proportional to the degree of the correlation. In the same figure (C), since there is no voice signal in the ear canal 4 at the time of 0 sec

and the waveform of pointC is flat, the correlation value is 0, but the waveform of the voice delayed to pointC appears at time t. The correlation value at this time t becomes maximum.

[0021] 8B and 8C are diagrams in which the level of the input waveform is modeled, but actually the waveform as shown in FIG. 8 is shown. The same figure (A) shows the waveform of point A, and the same figure (B) shows the waveform of point C. And the same figure (C) is a correlation value. As described above, in the case of an analog waveform, the correlation value does not have a pulse shape, but the peak value also changes after tsec.

[0022] On the other hand, with reference to FIGS. 2D and 2E, the signal waveform and the correlation when self-speech is performed will be described. When the wearer utters, the sound signal is emitted from the mouth to the outside and also propagates into the ear canal 4 through the body propagation path. The sound emitted to the outside world is received by the microphone 5, and the sound propagated to the ear canal 4 is received by the microphone 7. Since the propagation distances of both are extremely short, they are received almost at the same time (0 sec). Therefore, at time 0 sec, a waveform as shown in the upper part of FIG. 9D appears at the point A which is the input of the microphone 5, and at the time 0 sec, a waveform similar to the point A waveform also appears at the time 0 sec. Further, the pointA waveform is delayed by the delay device 10 for tsec (pointB waveform) and then output from the receiver 6 to the ear canal 4, so that the microphone 7 also receives this sound, and the microphone 7 causes the wearer generated in the ear canal. Both the own voice and the delayed voice output from the receiver 6 are received, and the received pointC waveform is a synthesized waveform as shown in the lower part of FIG. At time 0 sec, the waveforms at point A and point C are almost the same, so the correlation calculation device 11 outputs a high correlation value at time 0 sec as shown in FIG.

[0023] 9D and 9E are diagrams in which the level of the input waveform is modeled, the waveforms shown in FIG. 9 are actually shown. The same figure (A) shows the waveform of point A, and the same figure (B) shows the waveform of point C. And the same figure (C) is a correlation value. As described above, in the case of an analog waveform, the correlation value does not have a pulse shape, but it also has a peak value at time 0 and is gradually attenuated.

[0024] Based on the difference in the correlation values described above, when the peak of the correlation value is output at 0 sec, the speech determination device 12 determines that the correlated waveform is due to self-speech, and the peak of the correlation value is output after tsec. Then, it is determined that the waveform is due to the foreign voice.

[0025] FIG. 3 shows an example of a self-speech detection device in which the signal processing unit 9 is composed of a delay device 10, a correlation calculation device 11, a speech determination device 12, and a difference processing device 13. In FIG. 3A, the external voice received by the microphone 5 is input to the delay device 10 and the correlation calculation device 11. The delay device 10 delays the input audio signal by tsec and outputs it to the receiver 6. The receiver 6 converts this voice signal into voice (air vibration) and emits it into the ear canal 4. The signal (pointB) delayed by the delay device 10 is also input to the difference processing device 13. On the other hand, the microphone 7 receives the sound generated in the external auditory meatus 4 and inputs it to the difference processing device 13. The microphone 7 receives both the voice output by the receiver 6 and the voice of the wearer transmitted from the ear canal 4. The difference processing unit 13 cancels the foreign voice component by subtracting the same voice signal waveform as that output from the delay unit 10 to the receiver 6

from the voice signal waveform input from the microphone 7. The speech signal waveform in which the foreign speech component is canceled is input to the correlation calculation device 11. The correlation calculation device 11 calculates a correlation value between the voice signal input from the microphone 5 and the voice signal input from the difference processing device 13.

[0026] With reference to FIGS. 3B and 3C, the signal waveform and the correlation when the foreign voice arrives will be described. The time when the microphone 5 receives the foreign voice is set to 0 sec. Point A, which is the input to the microphone 5, has a waveform as shown in the first stage of FIG. The delay device 10 delays this audio signal for tsec and outputs it to the receiver 6. Since the pointB waveform supplied to the receiver 6 is the pointA waveform delayed by tsec, it has a waveform as shown in the second stage of FIG. The receiver 6 outputs the delayed audio signal to the ear canal 4. The sound output to the ear canal 4 is also received by the microphone 7. The pointC waveform, which is the input signal waveform of the microphone 7, has a waveform similar to the pointB waveform as shown in the third stage of FIG. The difference processing device 13 subtracts the pointC waveform, which is the audio signal waveform input from the microphone 7, from the pointB waveform, which is the audio signal waveform input from the delay device 10. In the case of only foreign voice, since there is no internally generated voice signal, the pointB waveform and the pointC waveform almost match, and the pointD waveform which is the output of the difference processing device 13 is as shown in the fourth stage of FIG. It becomes a flat signal of almost 0 level. The correlation calculation device 11 is a device that in time series examines how similar the point A waveform is to the reference waveform, using the point A waveform as the reference waveform, and outputs a correlation value proportional to the degree of the correlation. In this example, since the waveform of pointD is almost flat, there is almost no correlation with the reference waveform of pointA, and a large correlation value is not output. The utterance determination device 12 determines that there is no self-utterance because a large correlation value is not input.

[0027] On the other hand, a signal waveform and a correlation when a self-speech is performed will be described with reference to FIGS. When the wearer utters, the voice signal is emitted from the mouth to the outside world and also propagates into the ear canal 4 through the body propagation path. The sound emitted to the outside world is received by the microphone 5, and the sound transmitted to the external auditory meatus 4 is received by the microphone 7. Since the propagation distances of both are extremely short, they are received almost at the same time (0 sec). Therefore, the waveform of the point A appears at the microphone 5 at the time 0 sec, and the waveform similar to the point A also appears at the microphone 7 at the time 0 sec. Further, the waveform of the point A is delayed by the delay device 10 for tsec and then emitted from the receiver 6 to the ear canal 4, so that the microphone 7 also receives this voice. That is, the microphone 7 receives both the voice of the wearer himself/herself generated in the ear canal and the delayed voice emitted from the receiver 6, and the synthesized waveform thereof is as shown in the third row of FIG. It becomes the waveform of pointC. The difference processing device 13 subtracts the pointC waveform which is the voice signal waveform input from the microphone 7 and the pointB waveform which is the voice signal waveform input from the delay device 10. Since the pointC waveform is a composite signal of the internally transmitted waveform and the voice waveform output from the receiver 6, when the pointB waveform is subtracted from this, the waveform component of the external voice output from the receiver 6 is canceled, and) As shown in the fourth row, the waveform becomes similar to pointA. The correlation calculation device 11 is a device that in time series examines how similar the point A waveform is to the reference

waveform, using the point A waveform as the reference waveform, and outputs a correlation value proportional to the degree of the correlation. In this example, since the waveform of pointD almost coincides with the waveform of pointA, a large correlation value is output at the timing of 0 sec. The utterance determination device 12 determines that there is a self-utterance started at the timing of 0 sec due to the input of the large correlation value. As described above, in the example of FIG. 3, since the extra signal is eliminated by the difference processing device 13, the correlation value is improved and the error determination can be expected to be reduced.

[0028] The examples of FIGS. 2 and 3 have been described for the case where the signal path is ideal and the frequency characteristics and phase characteristics of the path have almost no effect, but consider the case where the propagation characteristics of the signal path affect the waveform of the audio signal. An example is shown in FIG.

[0029] In the example of FIG. 4, the signal processing unit 9 includes a delay device 10, a correlation calculation device 11, a speech determination device 12, a difference processing device 13, and a transfer function correction device 14.

[0030] In FIG. 4A, the external voice received by the microphone 5 is input to the delay device 10 and the correlation calculation device 11. The delay device 10 delays the input audio signal by tsec and outputs it to the receiver 6. The receiver 6 converts this voice signal into voice (air vibration) and emits it into the ear canal 4. The pointB waveform delayed by the delay device 10 is also input to the transfer function correction device 14. The transfer function correction device 14 filters the input signal and inputs it to the difference processing device 13. This filter is a filter that simulates the impulse response of the ear canal 4 when the device is inserted into the ear canal 4, so that the signal passing through this filter has the same waveform as the signal propagating in the ear canal 4. Is designed. That is, by passing through this filter, the transfer function of the propagation path is convoluted with respect to the signal waveform.

[0031] On the other hand, the microphone 7 receives the sound generated in the external auditory meatus 4 and inputs it to the difference processing device 13. The microphone 7 receives both the voice output by the receiver 6 and the voice of the wearer transmitted from the ear canal 4. The difference processing device 13 subtracts the audio signal waveform input from the transfer function correction device 14 from the audio signal waveform input from the microphone 7 to cancel this component. The audio signal waveform from which the output of the receiver 6 has been canceled is input to the correlation calculation device 11. The correlation calculation device 11 calculates a correlation value between the voice signal input from the microphone 5 and the voice signal input from the difference processing device 13.

[0032] With reference to FIG. 4(B), the signal waveform and the correlation when the foreign voice arrives will be described. The time when the microphone 5 receives the foreign voice is set to 0 sec. Point A, which is the input to the microphone 5, has a waveform as shown in the first stage of FIG. The delay device 10 delays this audio signal for tsec and outputs it to the receiver 6. The pointB, which is the waveform applied to the receiver 6, has a waveform as shown in the second stage of FIG. 7B, which is obtained by delaying the waveform of the pointA by tsec. The receiver 6 outputs the delayed audio signal to the ear canal 4. The sound output to the ear canal 4 is also received by the microphone 7. The point C, which is the input of the microphone 7, has almost the same timing as the point B, but the delay component is superimposed due to the propagation through the external auditory meatus 4, and the

waveform is as shown in the third stage of FIG. Since the waveform obtained by correcting the pointB waveform output from the delay device 10 by the transfer function correction device 14 is input to the difference processing device 13, this waveform is subtracted from the point C which is the audio signal waveform input from the microphone 7. .. Since the waveform of pointB corrected by the transfer function correction device 14 is almost the same as the waveform of pointC, these waveforms almost match and the pointE waveform which is the output of the difference processing device 13 is shown in FIG. As shown in the fifth stage, the signal becomes a flat signal of almost 0 level. The correlation calculating device 11 is a device for checking in time series how similar the point A waveform is to the reference waveform with the point A waveform as the reference waveform, and outputs a correlation value proportional to the degree of the correlation. In this example, since the waveform of point E is almost flat, there is almost no correlation with the reference waveform of point A, and a large correlation value is not output. The utterance determination device 12 determines that there is no self-utterance because a large correlation value is not input.

[0033] On the other hand, with reference to FIG. 4C, the signal waveform and the correlation when self-utterance is performed will be described. When the wearer utters, the sound signal is emitted from the mouth to the outside and also propagates into the ear canal 4 through the body propagation path. The sound emitted to the outside world is received by the microphone 5, and the sound transmitted to the external auditory meatus 4 is received by the microphone 7. Since the propagation distances of both are extremely short, they are received almost at the same time (0 sec). Therefore, the waveform of the point A appears at the microphone 5 at the time 0 sec, and the waveform similar to the point A waveform also appears at the microphone 7 at the time 0 sec. Further, the waveform of the point A is delayed by the delay device 10 for tsec and then emitted from the receiver 6 to the ear canal 4, so that the microphone 7 also receives this voice. The signal emitted from the receiver 6 and received by the microphone 7 is a signal that propagates through the external auditory meatus 4 and is input, so that the reverberation sound component of the external auditory meatus 4 is superimposed. Therefore, the synthesized waveform of the direct sound of the self-utterance received by the microphone 7 and the sound output from the receiver 6 becomes the pointC waveform shown in the third stage of FIG. The sound signal emitted from the receiver 6 and propagated through the external auditory meatus 4 has a gradual rise and fall of the waveform because the reverberation component is superimposed due to the propagation characteristics of the external auditory meatus 4 as described above. Since the waveform output from the transfer function correction device 14 also passes through a filter having the transfer characteristic of the external auditory meatus 4 as a filter coefficient, similar reverberation components are superposed, and this signal is input from the input signal of the microphone 7. By subtracting, only the self-speech sound can be separated (pointE waveform).

[0034] The correlation calculation device 11 is a device that in time series examines how similar the point A waveform is to the reference waveform, using the point A waveform as the reference waveform, and outputs a correlation value proportional to the degree of the correlation. In this example, since the waveform of pointD almost coincides with the waveform of pointA, a large correlation value is output at the timing of 0 sec. The utterance determination device 12 determines that there is a self utterance started at the timing of 0 sec due to the input of the large correlation value.

[0035] In FIG. 4A, if the output of the difference processing device 13 is taken out as it is, it can function as a self-speech detecting microphone.

[0036] Although the transfer function correction device 14 is inserted between the delay device 10 and the difference processing device 13 in the device of FIG. 4A, it may be placed between the microphone 7 and the difference processing device 13 as shown in FIG. Can also In this case, since the transfer function is also convoluted with the self-speech voice generated in the ear canal, the transfer function correction device that restores the waveform of the self-speech sound after the difference processing is the difference processing device 14 and the correlation calculation device 11. To be installed between and.

[0037] Note that, in the examples of FIGS. 3 and 4, the correlation calculation device 11 may be a simple comparison calculation device because it only compares two audio signal waveforms that occur simultaneously. Further, the delay device 10 of FIGS. 4 and 5 may be a signal processing device that digitally delays a signal.

[0038] FIG. 6 is a diagram showing an embodiment in which the present invention is applied to a voice input/output device of a communication terminal. A control device 30 for controlling this device controls the device in a transfer function calculation mode and a normal operation mode. The indicator device 33 includes a switch for the wearer to instruct the controller 30 to turn on/off the device.

[0039] 6, as a system of the microphone 5-delay device 10-receiver 6 of FIGS. 1 and 4A, a microphone 5 (this device also has the same external appearance as that of FIG. 1; the same applies hereinafter), an amplifier 36, An A/D converter 37, a delay device 38, a D/A converter 39, an amplifier 40 and a receiver 6 are provided. A microphone 7-amplifier 41 and an A/D converter 42 are provided as a system of the microphone 7 in FIGS. 1 and 4A. Further, the functions of the transfer function correction device 14, the difference processing device 13, and the comparison operation device 11 of FIG. 4A are implemented by software by the control device 30 and the operation processing device 35. Further, this apparatus has a built-in sound source 34 for generating a test sound signal in a transfer function calculation mode for calculating the transfer function of the ear canal 4. The audio signal formed by the sound source 34 is input to the A/D converter 37 as a test sound signal instead of being input to the microphone 5.

[0040] When the instruction device 33 gives an instruction to turn on, the control unit 30 first sets the transfer function mode, the control device 30 turns on the sound source 34, and instructs the arithmetic processing device 35 to perform a transfer function arithmetic operation. The test sound signal formed by the sound source 34 may be white noise or pink noise, but may be another signal, for example, a waveform of an impulse or a low-frequency signal having a fixed frequency. The test sound signal output from the sound source 34 is emitted into the external auditory meatus 4 via the A/D converter 37, the delay device 38, the D/A converter 39, the amplifier 40, and the receiver 6. The test sound emitted into the ear canal 4 is detected by the microphone 7 provided toward the ear canal 4. The output of the microphone 7 is input to the arithmetic processing unit 35 via the amplifier 41 and the A/D converter 42. The arithmetic processing unit 35 may be configured by a DSP or the like. On the other hand, the arithmetic processing unit 35 is also supplied with a signal obtained by converting the test sound signal formed by the sound source 34 into a digital signal from the A/D converter 37. The arithmetic processing unit 35 compares the pointA signal (the signal input from the A/D converter 37) and the pointB signal (the signal input from the A/D converter 42) as the transfer function operation and compares these signals. A correction coefficient is calculated according to the difference between There are various methods for obtaining the correction coefficient. Then, a function G(k) such that

[0041] $G(k)\cdot A(k)-B(k)=E(k)$ Once obtained, the function G(k) is in a relatively stable state unless the device is removed from the ear canal. You just need to calculate. When the calculation is completed, the

processing unit 35 goes into a standby state. When the control device 30 detects that the arithmetic processing device 35 is in the standby state, the control device 30 stops the sound source 34 and instructs the arithmetic processing device 35 to perform a signal processing operation to switch to the normal operation mode.

[0042] In the normal operation mode, when an external sound is input from the microphone 5, it passes through the amplifier 36, the A/D converter 37, the delay device 38, the D/A converter 39, and the amplifier 40, and after tsec from the receiver 6 to the ear canal 4. Is output. The external sound output to the external auditory meatus 4 is received by the microphone 7 provided toward the external auditory meatus 4, and is input to the arithmetic processing unit 35 via the amplifier 41 and the A/D converter 42. The arithmetic processing unit 35 in which the signal processing operation is set, after convolving the transfer function G(k) obtained above into the pointA signal input from the A/D converter 37, that is, the direct foreign voice signal, The difference calculation processing is performed with the pointB signal, that is, the signal including the external audio signal transmitted through the external auditory meatus 4. By this processing, the foreign voice component is deleted from the pointB signal and only the self-speech voice is extracted. If the extracted self-speech voice is connected to the voice input terminal of the communication device, only a voice signal with little noise can be picked up and sent to the other party even in a place where environmental noise is large.

[0043] In the above example, the output of the A/D converter 37 (pointA signal) is input to the arithmetic processing device 35 as the reference signal, but the output of the delay device 38 may be input.

[0044] FIG. 7 is a diagram showing an embodiment in which the present invention is applied to a hearing aid having a speech speed conversion function. The control device 50 for controlling this device controls the device in the transfer function calculation mode and the normal operation mode. The indicator device 53 includes a switch for the wearer to instruct the controller 50 to turn on/off the device.

[0045] In FIG. 7, as a system of the microphone 5-delay device 10-receiver 6 of FIGS. 1 and 4A, a microphone 5 (this device also has the same external appearance as that of FIG. 1; the same applies hereinafter), an amplifier 56, An A/D converter 57, a speech speed conversion/gain control device 58, a D/A converter 59, an amplifier 60 and a receiver 6 are provided. A microphone 7-amplifier 61 and an A/D converter 62 are provided as a system of the microphone 7 in FIGS. 1 and 4(A). Further, the functions of the transfer function correction device 14, the difference processing device 13, and the comparison operation device 11 of FIG. 4A are realized by software by the control device 50 and the operation processing device 55. Further, this apparatus has a built-in sound source 54 for generating a test sound signal in a transfer function calculation mode for calculating the transfer function of the ear canal 4. The audio signal formed by the sound source 54 is output by the receiver 9 provided outside and is received by the microphone 5.

[0046] When the instruction device 53 gives an instruction to turn on, the control unit 50 first sets the transfer function mode, the control device 50 turns on the sound source 54, and instructs the arithmetic processing device 55 to perform a transfer function arithmetic operation. The test sound signal formed by the sound source 54 may be white noise or pink noise, but may be formed of another signal, for example, an impulse or a waveform of a low frequency signal having a fixed frequency. The test sound signal output from the sound source 54 is output from the receiver 9 and received by the microphone 5 and converted into an electric signal, and then the A/D converter 57, the speech speed conversion/gain

control device 58, and the D/A conversion. It is emitted into the ear canal 4 through the container 59, the amplifier 60, and the receiver 6. The test sound emitted into the ear canal 4 is detected by the microphone 7 provided toward the ear canal 4. The output of the microphone 7 is input to the arithmetic processing unit 55 via the amplifier 61 and the A/D converter 62. The arithmetic processing unit 55 may be composed of a DSP or the like. On the other hand, the arithmetic processing unit 55 is also supplied with the voice source 54 formed by the voice speed conversion/gain control unit 58 and the voice speed converted from the test sound signal output from the receiver 9. The arithmetic processing unit 35 compares the point C signal (the signal input from the speech speed conversion/gain control unit 58) and the point B signal (the signal input from the A/D converter 62) as the transfer function arithmetic operation, and The correction coefficient is calculated according to the difference in the signal of. There are various methods for obtaining the correction coefficient, but for example, the point A signal is A(k), the point B signal is B(k), and the error E(k) between A(k) and B(k) is zero as shown in the following equation. Then, a function G(k) such that

[0047] $G(k)\cdot A(k)-B(k)=E(k)$ Once obtained, the function G(k) is in a relatively stable state unless the device is removed from the ear canal. You just need to calculate. When the calculation is completed, the calculation processing device 55 enters the standby state. When the control device 50 detects that the arithmetic processing device 55 is in the standby state, the control device 50 stops the sound source 54 and instructs the arithmetic processing device 55 to perform a signal processing operation to switch to the normal operation mode.

[0048] In the normal operation mode, when an external sound is input from the microphone 5, it passes through the amplifier 56, the A/D converter 57, the speech speed conversion/gain control device 58, the D/A converter 59, the amplifier 60, and the receiver after tsec. 6 is output to the ear canal 4. The external sound output to the external auditory meatus 4 is received by the microphone 7 provided toward the external auditory meatus 4, and is input to the arithmetic processing unit 55 via the amplifier 61 and the A/D converter 62. The arithmetic processing unit 55 to which the signal processing operation is set convolves the transfer function G(k) obtained above with the pointC signal input from the speech speed conversion/gain control unit 58, that is, the direct foreign voice signal. After that, a difference calculation process is performed with the pointB signal, that is, the signal including the external audio signal transmitted through the external auditory meatus 4. By this processing, the foreign voice component is deleted from the pointB signal and only the self-speech voice component is extracted. When the microphone 5 receives only the foreign voice, the self-speech voice component is 0, but when self-speaking, the component of that level is extracted. When the arithmetic processing device 55 extracts the self-speech voice component, the control device 50 prohibits the voice speed conversion/gain control device 58 from performing the voice speed conversion and controls the gain to be small. As a result, the speech speed conversion for the self-speech is prohibited, and the self-speech is smoothly performed.

[0049] In the above example, the output of the A/D converter 57 (pointA signal) may be input to the arithmetic processing device 55 as a reference signal.

[0050] As a result, the speech speed conversion is desired to be applied to the foreign voice, but not to the vocalized voice. When a person speaks, since the person controls the utterance while listening to his/her own voice, there arises a problem that the person cannot speak well if the uttered speech is delayed. Therefore, it is necessary to detect the utterance section and not apply the speech speed

conversion process to the self-utterance. In addition, since the self-speech voice sounds louder than the foreign voice, it is necessary to control the gain so as to have an appropriate volume.

[0051] In this way, since it is possible to separate the self-speech and the external sound with a simple configuration, when this is used in a speech speed conversion hearing aid, it is used in the same cell regardless of the shape of the ear canal of the wearer. be able to.

[0052] According to the present invention, since the external voice signal is canceled from the input signal of the second microphone provided toward the ear canal to extract only the self-speech sound, the structure is complicated. It is possible to extract only the self-speech voice from which the external voice such as environmental noise is removed without using a bone conduction microphone whose setting is delicate for each wearer.

[0053] Also, if this is used as a hearing aid, the speech speed conversion can be prohibited only in the section of the self-speech voice, and a hearing aid that can easily speak can be configured.

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Abstract

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PROBLEM TO BE SOLVED: To detect one's own uttered sound with simple constitution. SOLUTION: An external environment and an external auditory meatus are cut off and a signal received at the external environment is delayed by a prescribed time and outputted from a receiver 6 of the external auditory meatus. The external auditory meatus is provided with a microphone 7, which picks up a signal outputted from the receiver 6 and a voice signal that is uttered by a wearing person and propagated internally. The external voice signal component is cancelled by subtracting the signal component picked up by the microphone 5 out of the signal received by the microphone 7 so as to detect and extract only one's own uttered voice component.

IPR2024-01015 Page 00561

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(54)【発明の名称】　自己発話検出装置、音声入力装置および補聴器

(57)【特許請求の範囲】
【請求項１】　装用者の外耳道に挿入され、外界と外耳道との間を遮音する装着部と、
外界に向けて設けられた第１のマイクと、
外耳道内に向けて設けられた第２のマイクと、
第１のマイクの信号を所定時間遅延する遅延手段と、
遅延手段が遅延した信号を外耳道内に出力するレシーバと、
第１のマイクの信号波形と第２のマイクの信号波形との相関に基づいて装用者の自己発話を検出する信号処理部と、
を備えたことを特徴とする自己発話検出装置。
【請求項２】　前記信号処理部は、第１のマイクの信号波形と第２のマイクの信号波形の差分を計算し、この差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出する処理部である請求項１に記載の自己発話検出装置。
【請求項３】　前記信号処理部は、遅延手段で遅延された第１のマイクの信号波形を、レシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分を計算し、この差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出する処理部である請求項１に記載の自己発話検出装置。
【請求項４】　テスト音信号を発生して前記遅延手段に入力する音源と、
該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする請求項３に記載の自己発話検出装

置。
【請求項５】　装用者の外耳道に挿入され、外界と外耳道との間を遮音する装着部と、
外界に向けて設けられた第１のマイクと、
外耳道内に向けて設けられた第２のマイクと、
第１のマイクが受信した信号を所定時間遅延する遅延手段と、
遅延手段が遅延した信号を外耳道内に出力するレシーバと、
遅延手段で遅延された第１のマイクの信号波形を、レシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分信号波形を抽出し、この差分信号波形を外部出力する信号処理部と、
を備えたことを特徴とする音声入力装置。
【請求項６】　テスト音信号を発生して前記遅延手段に入力する音源と、
該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする請求項５に記載の音声入力装置。
【請求項７】　装用者の外耳道に挿入され、外界と外耳道との間を遮音する装着部と、
外界に向けて設けられた第１のマイクと、
外耳道内に向けて設けられた第２のマイクと、
第１のマイクが受信した信号を時間軸に伸長する話速変換手段と、
遅延手段が遅延した信号を外耳道内に出力するレシーバと、
話速変換手段で伸長された第１のマイクの信号波形を、レシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出し、自己発話が検出されたとき、前記話速変換手段の動作を禁止する信号処理部と、
を備えたことを特徴とする補聴器。
【請求項８】　テスト音信号を発生して前記話速変換手段に入力する音源と、
該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする請求項７に記載の補聴器。
【発明の詳細な説明】
【０００１】
【発明の属する技術分野】この発明は、装用者が発した音声と外界から到来した音声とを識別する機能を備えた自己発話検出装置、音声入力装置および補聴器に関する。
【０００２】
【従来の技術】従来、装用者が自ら発声した音声である自己発話を外来の音声から分離識別して検出するための装置として骨伝導マイクを用いたものが実用化されている。
【０００３】骨伝導マイクは、装用者の外耳道に密着させた装着部を介して装用者の発声による頭蓋骨の振動を検出するものであり、空気振動である外来音には感度がなく、装用者の自己発話音のみをピックアップできるものである。
【０００４】
【発明が解決しようとする課題】しかし、骨伝導マイクは、自己発話音を感度よくピックアップするためには装着部と外耳道との密着性が要求される。このため、装着部の形状を各装用者の外耳道の形状に合わせる必要があり、本体が装着部に一体になったセル型のものの場合にはセル全体の交換が必要になるため、装用者毎のフィッティングが面倒であるという問題点があった。
【０００５】この発明は、より簡略な構成で自己発話を検出することができる自己発話検出装置、音声入力装置および補聴器を提供することを目的とする。
【０００６】
【課題を解決するための手段】この出願の請求項１の発明は、装用者の外耳道に挿入され外界と外耳道との間を遮音する装着部と、外界に向けて設けられた第１のマイクと、外耳道内に向けて設けられた第２のマイクと、第１のマイクの信号を所定時間遅延する遅延手段と、遅延手段が遅延した信号を外耳道内に出力するレシーバと、第１のマイクの信号波形と第２のマイクの信号波形との相関に基づいて装用者の自己発話を検出する信号処理部と、を備えたことを特徴とする。
【０００７】この出願の請求項２の発明は、請求項１の発明において、前記信号処理部は、第１のマイクの信号波形と第２のマイクの信号波形の差分を計算し、この差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出する処理部であることを特徴とする。
【０００８】この出願の請求項３の発明は、請求項１の発明において、前記信号処理部は、遅延手段で遅延された第１のマイクの信号波形を、レシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分を計算し、この差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出する処理部であることを特徴とする。
【０００９】この出願の請求項４の発明は、請求項３の発明において、テスト音信号を発生して前記遅延手段に入力する音源と、該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする。

【００１０】この出願の請求項５の発明は、装用者の外耳道に挿入され外界と外耳道との間を遮音する装着部と、外界に向けて設けられた第１のマイクと、外耳道内に向けて設けられた第２のマイクと、第１のマイクが受信した信号を所定時間遅延する遅延手段と、遅延手段が遅延した信号を外耳道内に出力するレシーバと、遅延手段で遅延された第１のマイクの信号波形をレシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分信号波形を抽出し、この差分信号波形を外部出力する信号処理部と、を備えたことを特徴とする。

【００１１】この出願の請求項６の発明は、請求項５の発明において、テスト音信号を発生して前記遅延手段に入力する音源と、該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする。

【００１２】この出願の請求項７の発明は、装用者の外耳道に挿入され外界と外耳道との間を遮音する装着部と、外界に向けて設けられた第１のマイクと、外耳道内に向けて設けられた第２のマイクと、第１のマイクが受信した信号を時間軸に伸長する話速変換手段と、遅延手段が遅延した信号を外耳道内に出力するレシーバと、話速変換手段で伸長された第１のマイクの信号波形をレシーバから第２のマイクに至る外耳道を含む伝搬路の伝達関数で畳み込み演算し、この畳込信号波形と第２のマイクの信号波形の差分信号波形と第１のマイクの信号波形との相関に基づいて装用者の自己発話を検出し、自己発話が検出されたとき前記話速変換手段の動作を禁止する信号処理部と、を備えたことを特徴とする。

【００１３】この出願の請求項８の発明は、請求項７の発明において、テスト音信号を発生して前記遅延手段に入力する音源と、該テスト音信号の直接波形およびレシーバから出力され第２のマイクによって受信されたテスト音信号の伝搬波形に基づいて前記伝搬路の伝達関数を割り出す手段を備えたことを特徴とする。

【００１４】この発明では、第２のマイクは自己発話音以外にレシーバから出力された外来音声も受信する。一方、第１のマイクは外来音声を受信する。第２のマイクの信号波形と第１の信号波形の相関をとることによって外来音声の成分をキャンセルすることにより、自己発話音成分のみを分離抽出することができる。なお、第１のマイクが受信する外来音声には口から発声された自己発話音も含まれるが同様の処理でよい。

【００１５】信号処理部はＤＳＰなどで構成することができるため、複雑な信号処理をする場合でも構造的には簡略なものにすることができる。

【００１６】したがって、この発明によれば、外界と外耳道とを遮音するのみの骨伝導マイクよりも遙に簡略な装着部の構成で自己発話音のみを分離抽出することができる。

【００１７】

【発明の実施の形態】図面を参照してこの発明の実施形態について説明する。図１は、この発明が適用される自己発話検出装置の外観図である。この自己発話検出装置は補聴器や音声入力装置に用いられる。セル１の先端に外耳道４に挿入される装着部である耳栓２が設けられている。耳栓２は傘形の柔軟な樹脂で構成されており、外耳道壁３に密着して外耳道４と外界とを遮断し、相互間の音の伝搬を禁止する。

【００１８】セル１内には、外界から到来する音声を検出するためのマイク５、外耳道４に音響を出力するためのレシーバ６、外耳道４の内部の音声を検出するためのマイク７、および、レシーバ６，マイク７とを空間的に結ぶパイプ孔８が設けられている。また、マイク５，レシーバ６およびマイク７は信号処理部９に接続されている。信号処理部９はＤＳＰなどの電子回路で構成され、マイク５，マイク７から入力された音声信号を分析・処理し、処理された音声信号をレシーバ６から出力する。以下、信号処理部９を種々に構成した例とその動作について説明する。

【００１９】図２は信号処理部９を遅延装置１０、相関演算装置１１、発話判定装置１２で構成した自己発話検出装置の例を示している。図２（Ａ）において、マイク５が受信した外来音声は、遅延装置１０および相関演算装置１１に入力される。遅延装置１０は入力された音声信号を微小時間ｔだけ遅延させてレシーバ６に出力する。レシーバ６はこの音声信号を音声（空気振動）に変換して外耳道４内に放出する。マイク７は、レシーバ６が出力した音声および外耳道４から伝達された装用者の音声の両方を受信する。マイク７が受信した音声信号は相関演算装置１１に入力される。相関演算装置１１は、マイク５から入力された音声信号とマイク７から入力された音声信号の相関値を計算する。

【００２０】図２（Ｂ）、（Ｃ）を参照して外来音声が到来したときの信号波形と相関について説明する。マイク５が外来音声を受信した時刻を０ｓｅｃとする。マイク５の入力であるｐｏｉｎｔＡ波形は、同図（Ｂ）上段のような波形となる。遅延装置１０はこの音声信号をｔｓｅｃ遅延させてレシーバ６に出力する。レシーバ６に供給される波形であるｐｏｉｎｔＢ波形は、ｐｏｉｎｔＡの波形をｔｓｅｃ遅延させた同図（Ｂ）中段のような波形となる。レシーバ６はこの遅延された音声信号を外耳道４に出力する。外耳道４に出力された音声はマイク７にも受信される。マイク７の入力であるｐｏｉｎｔＣ波形は、ｐｏｉｎｔＢ波形とほぼ同様の同図（Ｂ）下段のような波形となる。相関演算装置１１は、ｐｏｉｎｔＡの波形を基準波形とし、ｐｏｉｎｔＣの波形が基準波形にどの程度類似しているかを時系列に検査する装置であり、その相関の程度に比例した相関値を出力する。同

図（Ｃ）において、０ｓｅｃの時点では外耳道４内には音声信号がなくｐｏｉｎｔＣの波形はフラットであるので相関値は0であるが、時刻ｔにｐｏｉｎｔＣに遅延された音声の波形が現れるのでこの時刻ｔの相関値が最大となる。

【００２１】なお、同図（Ｂ），（Ｃ）は入力波形のレベルをモデル化した図であるが、実際には図8のような波形を示す。同図（Ａ）がｐｏｉｎｔＡの波形であり、同図（Ｂ）がｐｏｉｎｔＣの波形である。そして同図（Ｃ）が相関値である。このようにアナログ波形の場合、相関値はパルス状にはならないが、やはりｔｓｅｃ後にピーク値をとるように推移する。

【００２２】一方、図２（Ｄ）、（Ｅ）を参照して自己発話が行われたときの信号波形と相関について説明する。装用者が発声すると、その音声信号は口から外界に放出されるとともに体内伝搬経路によって外耳道４内にも伝搬する。外界に放出された音声はマイク５に受信され、外耳道４に伝搬した音声はマイク７に受信される。双方とも伝搬距離が極めて短いため、ほぼ同時（０ｓｅｃ）に受信される。したがって、時刻０ｓｅｃにマイク５の入力であるｐｏｉｎｔＡには同図（Ｄ）上段に示すような波形が現れ、マイク７にも時刻０ｓｅｃにｐｏｉｎｔＡ波形と類似した波形が現れる。また、ｐｏｉｎｔＡ波形は遅延装置１０でｔｓｅｃ遅延されたのち（ｐｏｉｎｔＢ波形）、レシーバ６から外耳道４に出力されるので、マイク７はこの音声も受信し、マイク７は、外耳道内に発生した装用者自身の音声とレシーバ6から出力される遅延された音声の両方を受信し、受信した波形であるｐｏｉｎｔＣ波形は同図（Ｄ）下段のような合成波形となる。時刻０ｓｅｃではポイントＡとポイントＣの波形はほぼ同様であるため、相関演算装置１１は、同図（Ｅ）のように時刻０ｓｅｃに高い相関値を出力する。

【００２３】なお、同図（Ｄ），（Ｅ）は入力波形のレベルをモデル化した図であるが、実際には図９のような波形を示す。同図（Ａ）がｐｏｉｎｔＡの波形であり、同図（Ｂ）がｐｏｉｎｔＣの波形である。そして同図（Ｃ）が相関値である。このようにアナログ波形の場合、相関値はパルス状にはならないが、やはり時刻０でピーク値をとり、徐々に減衰している。

【００２４】以上説明した相関値の相違に基づき、発話判定装置１２は、０ｓｅｃに相関値のピークが出力されたとき、その相関する波形は自己発話によるものと判定し、ｔｓｅｃ後に相関値のピークが出力されたとき、その波形は外来音声によるものと判定する。

【００２５】図３は信号処理部９を遅延装置１０、相関演算装置１１、発話判定装置１２および差分処理装置１３で構成した自己発話検出装置の例を示す。図３（Ａ）において、マイク５が受信した外来音声は遅延装置１０および相関演算装置１１に入力される。遅延装置１０は入力された音声信号をｔｓｅｃだけ遅延させてレシーバ６に出力する。レシーバ6はこの音声信号を音声（空気振動）に変換して外耳道４内に放出する。遅延装置１０が遅延した信号（ｐｏｉｎｔＢ）は差分処理装置１３にも入力される。一方、マイク７は、外耳道４内に発生した音声を受信して差分処理装置１３に入力する。マイク７は、レシーバ６が出力した音声および外耳道４から伝達された装用者の音声の両方を受信する。差分処理装置１３はマイク７から入力された音声信号波形から遅延装置１０がレシーバ６に出力したものと同じ音声信号波形を減算して外来音声成分をキャンセルする。この外来音声成分がキャンセルされた音声信号波形を相関演算装置１１に入力する。相関演算装置１１は、マイク５から入力された音声信号と差分処理装置１３から入力された音声信号の相関値を計算する。

【００２６】図３（Ｂ）、（Ｃ）を参照して外来音声が到来したときの信号波形と相関について説明する。マイク５が外来音声を受信した時刻を０ｓｅｃとする。マイク５の入力であるｐｏｉｎｔＡは、同図（Ｂ）第１段のような波形となる。遅延装置１０はこの音声信号をｔｓｅｃ遅延させてレシーバ６に出力する。レシーバ６に供給されるｐｏｉｎｔＢ波形は、ｐｏｉｎｔＡ波形をｔｓｅｃ遅延させたものであるため同図（Ｂ）第２段のような波形となる。レシーバ６はこの遅延された音声信号を外耳道４に出力する。外耳道４に出力された音声はマイク７にも受信される。マイク７の入力信号波形であるｐｏｉｎｔＣ波形は、ｐｏｉｎｔＢ波形とほぼ同様の同図（Ｂ）第３段のような波形となる。差分処理装置１３はマイク７から入力された音声信号波形であるｐｏｉｎｔＣ波形と遅延装置１０から入力された音声信号波形であるｐｏｉｎｔＢ波形とを減算する。外来音声のみの場合、内部で発生した音声信号がないため、ｐｏｉｎｔＢ波形とｐｏｉｎｔＣ波形はほとんど一致し、差分処理装置１３の出力であるｐｏｉｎｔＤ波形は、同図（Ｂ）第４段に示すように殆ど０レベルのフラットな信号となる。相関演算装置１１は、ｐｏｉｎｔＡの波形を基準波形とし、ｐｏｉｎｔＤの波形が基準波形にどの程度類似しているかを時系列に検査する装置であり、その相関の程度に比例した相関値を出力する。この例では、ｐｏｉｎｔＤの波形は殆どフラットであるためｐｏｉｎｔＡの基準波形と殆ど相関性がなく、大きな相関値が出力されない。発話判定装置１２は大きな相関値が入力されないことにより、自己発話がないことを判断する。

【００２７】一方、図３（Ｄ）、（Ｅ）を参照して自己発話が行われたときの信号波形と相関について説明する。装用者が発声すると、その音声信号は口から外界に放出されるとともに体内伝搬経路によって外耳道４内にも伝搬する。外界に放出された音声はマイク５に受信され、外耳道４に伝搬した音声はマイク７に受信される。双方とも伝搬距離が極めて短いため、ほぼ同時（０ｓｅｃ）に受信される。したがって、マイク５には時刻０ｓ

ｅｃにｐｏｉｎｔＡの波形が現れ、マイク７にも時刻０ｓｅｃにｐｏｉｎｔＡと類似した波形が現れる。また、ｐｏｉｎｔＡの波形は遅延装置１０でｔｓｅｃ遅延されたのちレシーバ６から外耳道４に放出されるので、マイク７はこの音声も受信する。すなわち、マイク７は、外耳道内に発生した装用者自身の音声とレシーバ６から放出される遅延された音声の両方を受信し、その合成波形は、同図（Ｄ）第３段に示すようなｐｏｉｎｔＣの波形となる。差分処理装置１３は、マイク７から入力された音声信号波形であるｐｏｉｎｔＣ波形と遅延装置１０から入力された音声信号波形であるｐｏｉｎｔＢ波形とを減算する。ｐｏｉｎｔＣ波形は、内部伝達した波形とレシーバ６から出力された音声波形の合成信号であるため、これからｐｏｉｎｔＢ波形を減算すると、レシーバ６から出力された外来音声の波形成分がキャンセルされ、同図（Ｂ）第４段に示すようにｐｏｉｎｔＡと類似した波形となる。相関演算装置１１は、ｐｏｉｎｔＡの波形を基準波形とし、ｐｏｉｎｔＤの波形が基準波形にどの程度類似しているかを時系列に検査する装置であり、その相関の程度に比例した相関値を出力する。この例では、ｐｏｉｎｔＤの波形は殆どｐｏｉｎｔＡの波形と一致しているため、０ｓｅｃのタイミングに大きな相関値が出力される。発話判定装置１２は、大きな相関値が入力されたことにより、０ｓｅｃのタイミングに開始した自己発話があったことを判断する。このように、図３の例では、差分処理装置１３によって余分な信号を排除しているので相関値が改善され誤り判定の減少が期待できる。

【００２８】図２，図３の例は、信号経路が理想的で経路の周波数特性や位相特性の影響がほとんど無い場合について説明したが、信号経路の伝搬特性が音声信号の波形に影響を与える場合を考慮した例を図４に示す。

【００２９】図４の例では、信号処理部９を遅延装置１０、相関演算装置１１、発話判定装置１２、差分処理装置１３に加えて伝達関数補正装置１４で構成している。

【００３０】図４（Ａ）において、マイク５が受信した外来音声は遅延装置１０および相関演算装置１１に入力される。遅延装置１０は入力された音声信号をｔｓｅｃだけ遅延させてレシーバ６に出力する。レシーバ６はこの音声信号を音声（空気振動）に変換して外耳道４内に放出する。遅延装置１０が遅延したｐｏｉｎｔＢ波形は伝達関数補正装置１４にも入力される。伝達関数補正装置１４は入力された信号をフィルタ処理して差分処理装置１３に入力する。このフィルタは、この装置が外耳道４に挿入されているときの外耳道４のインパルス応答をシミュレートしたフィルタであり、このフィルタを通過した信号が外耳道４内を伝搬した信号と同じ波形になるように設計されている。すなわち、このフィルタを通過させることにより、信号波形に対して伝搬路の伝達関数が畳み込み演算される。

【００３１】一方、マイク７は、外耳道４内に発生した音声を受信して差分処理装置１３に入力する。マイク７は、レシーバ６が出力した音声および外耳道４から伝達された装用者の音声の両方を受信する。差分処理装置１３はマイク７から入力された音声信号波形から伝達関数補正装置１４から入力された音声信号波形を減算してこの成分をキャンセルする。このレシーバ６の出力分がキャンセルされた音声信号波形を相関演算装置１１に入力する。相関演算装置１１は、マイク５から入力された音声信号と差分処理装置１３から入力された音声信号の相関値を計算する。

【００３２】図４（Ｂ）を参照して外来音声が到来したときの信号波形と相関について説明する。マイク５が外来音声を受信した時刻を０ｓｅｃとする。マイク５の入力であるｐｏｉｎｔＡは、同図（Ｂ）第１段のような波形となる。遅延装置１０はこの音声信号をｔｓｅｃ遅延させてレシーバ６に出力する。レシーバ６にされる波形であるｐｏｉｎｔＢは、ｐｏｉｎｔＡの波形をｔｓｅｃ遅延させた同図（Ｂ）第２段のような波形となる。レシーバ６はこの遅延された音声信号を外耳道４に出力する。外耳道４に出力された音声はマイク７にも受信される。マイク７の入力であるｐｏｉｎｔＣは、ｐｏｉｎｔＢとタイミングはほぼ同じであるが、外耳道４を伝搬したことによって遅延成分が重畳されており同図（Ｂ）第３段のような波形となる。差分処理装置１３には、遅延装置１０から出力されるｐｏｉｎｔＢ波形を伝達関数補正装置１４で補正した波形が入力されるため、この波形をマイク７から入力された音声信号波形であるｐｏｉｎｔＣから減算する。ｐｏｉｎｔＢの波形を伝達関数補正装置１４で補正したものはほぼｐｏｉｎｔＣの波形と同じになっているため、これらの波形はほとんど一致し差分処理装置１３の出力であるｐｏｉｎｔＥ波形は、同図（Ｂ）第５段に示すように殆ど０レベルのフラットな信号となる。相関演算装置１１は、ｐｏｉｎｔＡの波形を基準波形とし、ｐｏｉｎｔＥの波形が基準波形にどの程度類似しているかを時系列に検査する装置であり、その相関の程度に比例した相関値を出力する。この例では、ｐｏｉｎｔＥの波形は殆どフラットであるためｐｏｉｎｔＡの基準波形と殆ど相関性がなく、大きな相関値が出力されない。発話判定装置１２は大きな相関値が入力されないことにより、自己発話がないことを判断する。

【００３３】一方、図４（Ｃ）を参照して自己発話が行われたときの信号波形と相関について説明する。装用者が発声すると、その音声信号は口から外界に放出されるとともに体内伝搬経路によって外耳道４内にも伝搬する。外界に放出された音声はマイク５に受信され、外耳道４に伝搬した音声はマイク７に受信される。双方とも伝搬距離が極めて短いため、ほぼ同時（０ｓｅｃ）に受信される。したがって、マイク５には時刻０ｓｅｃにｐｏｉｎｔＡの波形が現れ、マイク７にも時刻０ｓｅｃに

ｐｏｉｎｔＡ波形と類似した波形が現れる。また、ｐｏｉｎｔＡの波形は遅延装置１０でｔｓｅｃ遅延されたのちレシーバ６から外耳道４に放出されるので、マイク７はこの音声も受信する。レシーバ６から放出されマイク７が受信する信号は外耳道４を伝搬して入力される信号であるため、外耳道４による残響音成分が重畳されたものになっている。このためマイク７が受信する自己発話の直接音とレシーバ６から出力された音の合成波形は、同図（Ｃ）第３段に示すｐｏｉｎｔＣ波形となる。レシーバ６から放出され外耳道４を伝搬した音声信号は、上述したように外耳道４の伝搬特性によって残響分が重畳されているため、波形の立ち上がりと立ち下がりが緩やかになっている。伝達関数補正装置１４から出力された波形も外耳道４の伝達特性をフィルタ係数として有するフィルタを通過したものであるため、同じような残響成分が重畳されており、この信号をマイク７の入力信号から減算することにより、自己発話音のみを分離することができる（ｐｏｉｎｔＥ波形）。

【００３４】相関演算装置１１は、ｐｏｉｎｔＡの波形を基準波形とし、ｐｏｉｎｔＤの波形が基準波形にどの程度類似しているかを時系列に検査する装置であり、その相関の程度に比例した相関値を出力する。この例では、ｐｏｉｎｔＤの波形は殆どｐｏｉｎｔＡの波形と一致しているため、０ｓｅｃのタイミングに大きな相関値が出力される。発話判定装置１２は、大きな相関値が入力されたことにより、０ｓｅｃのタイミングに開始した自己発話があったことを判断する。

【００３５】なお、図４（Ａ）において、差分処理装置１３の出力をそのまま取り出せば自己発話検出マイクとして機能させることができる。

【００３６】なお、図４（Ａ）の装置では、伝達関数補正装置１４を遅延装置１０と差分処理装置１３の間に挿入したが、図５のように、マイク７と差分処理装置１３の間におくこともできる。この場合、伝達関数が外耳道内に発生した自己発話音声にも畳み込まれてしまうので、差分処理後自己発話音の波形をもとに戻す伝達関数修正装置を差分処理装置１４と相関演算装置１１との間に設ける。

【００３７】なお、図３および図４の例において、相関演算装置１１は、同時に発生する２つの音声信号波形を比較するのみであるため、単純な比較演算装置でもよい。さらに、図４，図５の遅延装置１０は、ディジタル的に信号を遅延させる信号処理装置であってもよい。

【００３８】図６は、この発明を通信端末の音声入出力装置に適用した実施形態を示す図である。この装置を制御する制御装置３０は伝達関数演算モードと通常動作モードで装置を制御する。指示装置３３は装用者がこの装置のオン／オフを制御装置３０に指示するためのスイッチを含んでいる。

【００３９】図６において、図１および図４（Ａ）のマイク５－遅延装置１０－レシーバ６の系統として、マイク５（この装置も外観は図１のものと同様である：以下同じ）、増幅器３６、Ａ／Ｄ変換器３７、遅延装置３８、Ｄ／Ａ変換器３９、増幅器４０およびレシーバ６が設けられている。また、図１および図４（Ａ）のマイク７の系統として、マイク７－増幅器４１、Ａ／Ｄ変換器４２が設けられている。また、図４（Ａ）の伝達関数補正装置１４、差分処理装置１３、比較演算装置１１の機能は、制御装置３０および演算処理装置３５がソフト的に実現する。また、この装置は、外耳道４の伝達関数を割り出す伝達関数演算モードにテスト音信号を発生するための音源３４を内蔵している。この音源３４が形成したオーディオ信号は、テスト音信号としてマイク５の入力に変えてＡ／Ｄ変換器３７に入力される。

【００４０】指示装置３３によってオンが指示されると、制御部３０は最初に伝達関数モードを設定し、制御装置３０は音源３４をオンするとともに、演算処理装置３５に伝達関数演算動作を指示する。音源３４が形成するテスト音信号は、ホワイトノイズやピンクノイズでよいが、他の信号、例えばインパルスや周波数の定まった低周波信号の波形を形成するようにしてもよい。音源３４から出力されたテスト音信号はＡ／Ｄ変換器３７、遅延装置３８、Ｄ／Ａ変換器３９、増幅器４０、レシーバ６を経て外耳道４内に放出される。外耳道４内に放出されたテスト音は外耳道４に向けて設けられたマイク７で検出される。マイク７の出力は増幅器４１、Ａ／Ｄ変換器４２を経て演算処理装置３５に入力される。演算処理装置３５はＤＳＰなどで構成すればよい。一方、この演算処理装置３５には、Ａ／Ｄ変換器３７から音源３４が形成したテスト音信号をディジタル信号に変換したものも供給される。演算処理装置３５は、伝達関数演算動作として、ｐｏｉｎｔＡ信号（Ａ／Ｄ変換器３７から入力される信号）およびｐｏｉｎｔＢ信号（Ａ／Ｄ変換器４２から入力される信号）を比較し、これらの信号の相違に応じた補正係数を算出する。補正係数を求める手法は色々あるが例えば下式のようにｐｏｉｎｔＡ信号をＡ（ｋ）、ｐｏｉｎｔＢ信号をＢ（ｋ）とし、Ａ（ｋ）とＢ（ｋ）間の誤差Ｅ（ｋ）がゼロになるような関数Ｇ（ｋ）を求める。

【００４１】Ｇ（ｋ）・Ａ（ｋ）－Ｂ（ｋ）＝Ｅ（ｋ）

一度求めた関数Ｇ（ｋ）は装置を外耳道から取り外したりしなければ、比較的安定した状態にあるので、動作の最初に計算するのみでよい。演算が終了すると演算処理装置３５は待機状態になる。制御装置３０は演算処理装置３５が待機状態になったことを検出すると、音源３４を停止するとともに、演算処理装置３５に信号処理動作を指示して通常動作モードに切り換える。

【００４２】通常動作モードにおいて、マイク５から外来音声が入力されると、増幅器３６、Ａ／Ｄ変換器３７、遅延装置３８、Ｄ／Ａ変換器３９、増幅器４０を経

て、ｔｓｅｃ後にレシーバ６から外耳道４に出力される。外耳道４に出力された外来音声は外耳道４に向けて設けられたマイク７によって受信され、増幅器４１、Ａ／Ｄ変換器４２を経て演算処理装置３５に入力される。信号処理動作が設定されている演算処理装置３５は、Ａ／Ｄ変換器３７から入力されるｐｏｉｎｔＡ信号、すなわち直接の外来音声信号に上記で求めた伝達関数Ｇ（ｋ）を畳み込んだのち、ｐｏｉｎｔＢ信号、すなわち外耳道４を伝達した外来音声信号を含む信号と差演算処理を行う。この処理によってｐｏｉｎｔＢ信号から外来音声成分は消去され自己発話音声のみが抽出される。抽出された自己発話音声を通信装置の音声入力端子に接続すれば環境雑音が大きな場所でも雑音の少ない音声信号のみをピックアップして相手に送ることができる。

【００４３】なお、上記例では、参照信号としてＡ／Ｄ変換器３７の出力（ｐｏｉｎｔＡ信号）を演算処理装置３５に入力しているが、遅延装置３８の出力を入力するようにしてもよい。

【００４４】図７はこの発明を話速変換機能をもった補聴器に適用した実施形態を示す図である。この装置を制御する制御装置５０は伝達関数演算モードと通常動作モードで装置を制御する。指示装置５３は装用者がこの装置のオン／オフを制御装置５０に指示するためのスイッチを含んでいる。

【００４５】図７において、図１および図４（Ａ）のマイク５－遅延装置１０－レシーバ６の系統として、マイク５（この装置も外観は図１のものと同様である：以下同じ）、増幅器５６、Ａ／Ｄ変換器５７、話速変換・利得制御装置５８、Ｄ／Ａ変換器５９、増幅器６０およびレシーバ６が設けられている。また、図１および図４（Ａ）のマイク７の系統として、マイク７－増幅器６１、Ａ／Ｄ変換器６２が設けられている。また、図４（Ａ）の伝達関数補正装置１４、差分処理装置１３、比較演算装置１１の機能は、制御装置５０および演算処理装置５５がソフト的に実現する。また、この装置は、外耳道４の伝達関数を割り出す伝達関数演算モードにテスト音信号を発生するための音源５４を内蔵している。この音源５４が形成したオーディオ信号は、外部に向けて設けられたレシーバ９によって出力され、マイク５によって受信される。

【００４６】指示装置５３によってオンが指示されると、制御部５０は最初に伝達関数モードを設定し、制御装置５０は音源５４をオンするとともに、演算処理装置５５に伝達関数演算動作を指示する。音源５４が形成するテスト音信号は、ホワイトノイズやピンクノイズでよいが、他の信号、例えばインパルスや周波数の定まった低周波信号の波形を形成するようにしてもよい。音源５４から出力されたテスト音信号は、レシーバ９から出力され、マイク５で受信・電気信号に変換されたのち、Ａ／Ｄ変換器５７、話速変換・利得制御装置５８、Ｄ／Ａ変換器５９、増幅器６０、レシーバ６を経て外耳道４内に放出される。外耳道４内に放出されたテスト音は外耳道４に向けて設けられたマイク７で検出される。マイク７の出力は増幅器６１、Ａ／Ｄ変換器６２を経て演算処理装置５５に入力される。演算処理装置５５はＤＳＰなどで構成すればよい。一方、この演算処理装置５５には、話速変換・利得制御装置５８から音源５４が形成し、レシーバ９から出力されたテスト音信号を話速変換したものも供給される。演算処理装置３５は、伝達関数演算動作として、ｐｏｉｎｔＣ信号（話速変換・利得制御装置５８から入力される信号）およびｐｏｉｎｔＢ信号（Ａ／Ｄ変換器６２から入力される信号）を比較し、これらの信号の相違に応じた補正係数を算出する。補正係数を求める手法は色々あるが例えば下式のようにｐｏｉｎｔＡ信号をＡ（ｋ）、ｐｏｉｎｔＢ信号をＢ（ｋ）とし、Ａ（ｋ）とＢ（ｋ）間の誤差Ｅ（ｋ）がゼロになるような関数Ｇ（ｋ）を求める。

【００４７】Ｇ（ｋ）・Ａ（ｋ）－Ｂ（ｋ）＝Ｅ（ｋ）

一度求めた関数Ｇ（ｋ）は装置を外耳道から取り外したりしなければ、比較的安定した状態にあるので、動作の最初に計算するのみでよい。演算が終了すると演算処理装置５５は待機状態になる。制御装置５０は演算処理装置５５が待機状態になったことを検出すると、音源５４を停止するとともに、演算処理装置５５に信号処理動作を指示して通常動作モードに切り換える。

【００４８】通常動作モードにおいて、マイク５から外来音声が入力されると、増幅器５６、Ａ／Ｄ変換器５７、話速変換・利得制御装置５８、Ｄ／Ａ変換器５９、増幅器６０を経て、ｔｓｅｃ後にレシーバ６から外耳道４に出力される。外耳道４に出力された外来音声は外耳道４に向けて設けられたマイク７によって受信され、増幅器６１、Ａ／Ｄ変換器６２を経て演算処理装置５５に入力される。信号処理動作が設定されている演算処理装置５５は、話速変換・利得制御装置５８から入力されるｐｏｉｎｔＣ信号、すなわち直接の外来音声信号に上記で求めた伝達関数Ｇ（ｋ）を畳み込んだのち、ｐｏｉｎｔＢ信号、すなわち外耳道４を伝達した外来音声信号を含む信号と差演算処理を行う。この処理によってｐｏｉｎｔＢ信号から外来音声成分は消去され自己発話音声成分のみが抽出される。マイク５が外来音声のみを受信しているときは、この自己発話音声成分は０であるが、自己発話しているときには、そのレベルの成分が抽出される。制御装置５０は演算処理装置５５が自己発話音声成分を抽出したとき、話速変換・利得制御装置５８に対して話速変換を禁止し、利得を小さめに制御する。これにより、自己発話に対する話速変換が禁止され、自己発話が滑らかに行われるようになる。

【００４９】なお、上記例では、参照信号としてＡ／Ｄ変換器５７の出力（ｐｏｉｎｔＡ信号）を演算処理装置５５にも入力するようにすればよい。

【００５０】これにより、話速変換は外来音声に作用させたいが、自分の発声音声には作用させたくない。自分が発話するとき、人は自分の声を聞きながら発声を制御しているため、発声音声が遅れて聞こえるとうまく話せないという問題が起こる。そのため、自己発話に対しては、発話区間を検出して話速変換処理を適応しないようにする必要がある。また外来音声に比べ、自己発話音声は大きく聞こえるので適当な大きさになるよう利得制御が必要となる。

【００５１】このように、簡略な構成で自己発話と外来音とを分離することができるため、話速変換補聴器にこれを用いた場合に、装用者の外耳道の形状にかかわらず同じセルでこれを使用することができる。

【００５２】

【発明の効果】この発明によれば、外耳道に向けて設けられた第２のマイクの入力信号から外来音声信号をキャンセルして自己発話音のみを抽出できるようにしたことにより、構造が複雑で装用者毎のセッティングが微妙な骨伝導マイクを用いなくても環境ノイズなどの外来音声を除去した自己発話音声のみを抽出処理することができる。

【００５３】また、これを補聴器にもちいれば自己発話音声の区間のみ話速変換を禁止し、発話が容易な補聴器を構成することができる。

【図面の簡単な説明】

【図１】この発明の実施形態である自己発話検出装置の外観図

【図２】請求項１の発明の実施形態である自己発話検出装置のブロック図および信号波形を示す図

【図３】請求項２の発明の実施形態である自己発話検出装置のブロック図および信号波形を示す図

【図４】請求項３の発明の実施形態である自己発話検出装置のブロック図および信号波形を示す図

【図５】道自己発話検出装置の他の実施形態を示す図

【図６】請求項４の発明の実施形態である音声入力装置のブロック図号波形を示す図

【図７】この発明の自己発話検出装置を補聴器に応用した例を示す図

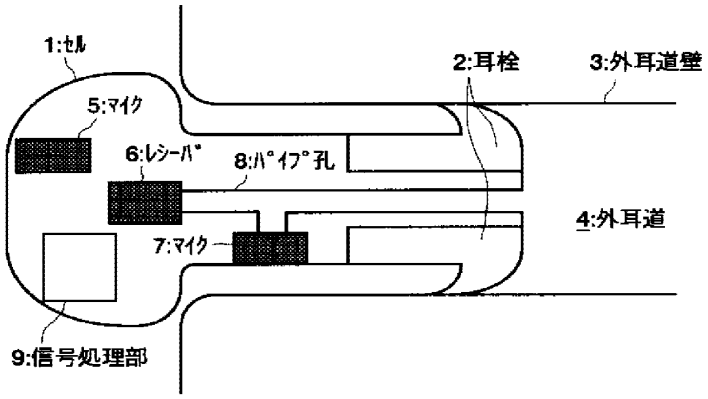
【図８】自己発話検出装置の各部における信号波形の例を示す図

【図９】自己発話検出装置の各部における信号波形の例を示す図

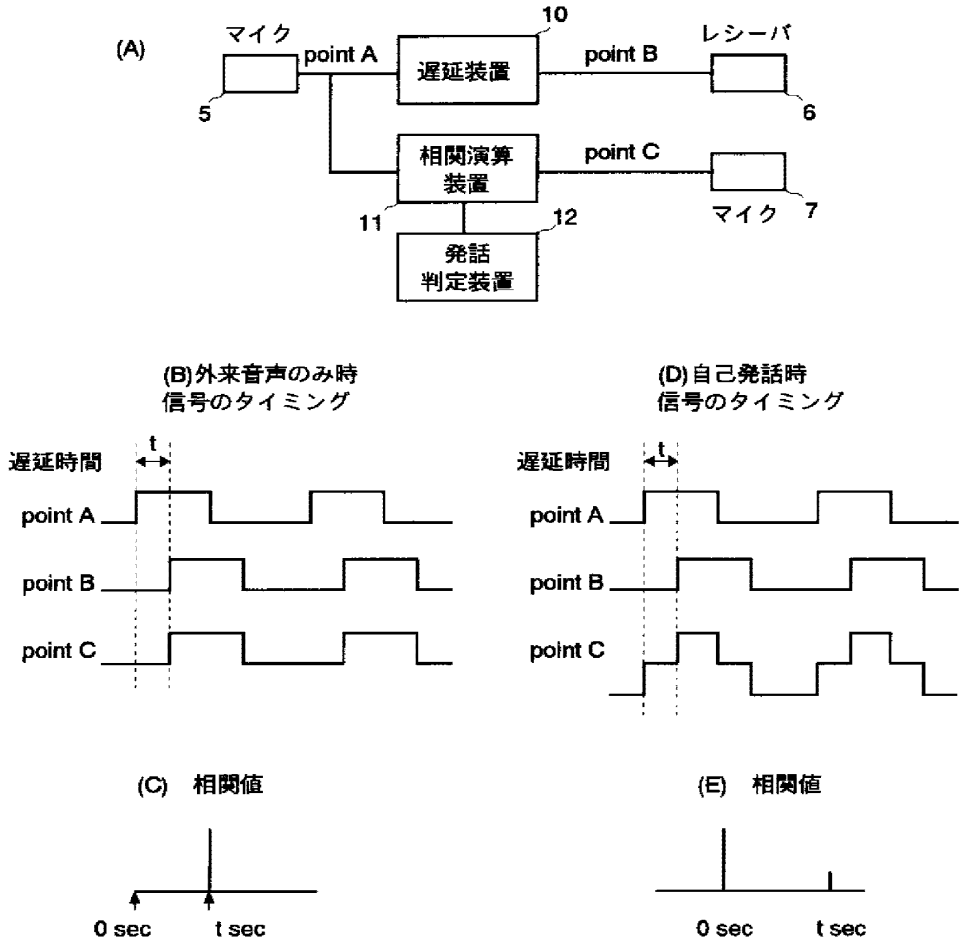
【符号の説明】

１…セル、２…耳栓、３…外耳道壁、４…外耳道、５…（外界に向けられた）マイク、６…レシーバ、７…（外耳道に向けられた）マイク、８…パイプ孔、９…信号処理部、１０…遅延装置、１１…相関演算装置、１２…発話判定装置、１３…差分処理装置、１４…伝達関数補正装置、

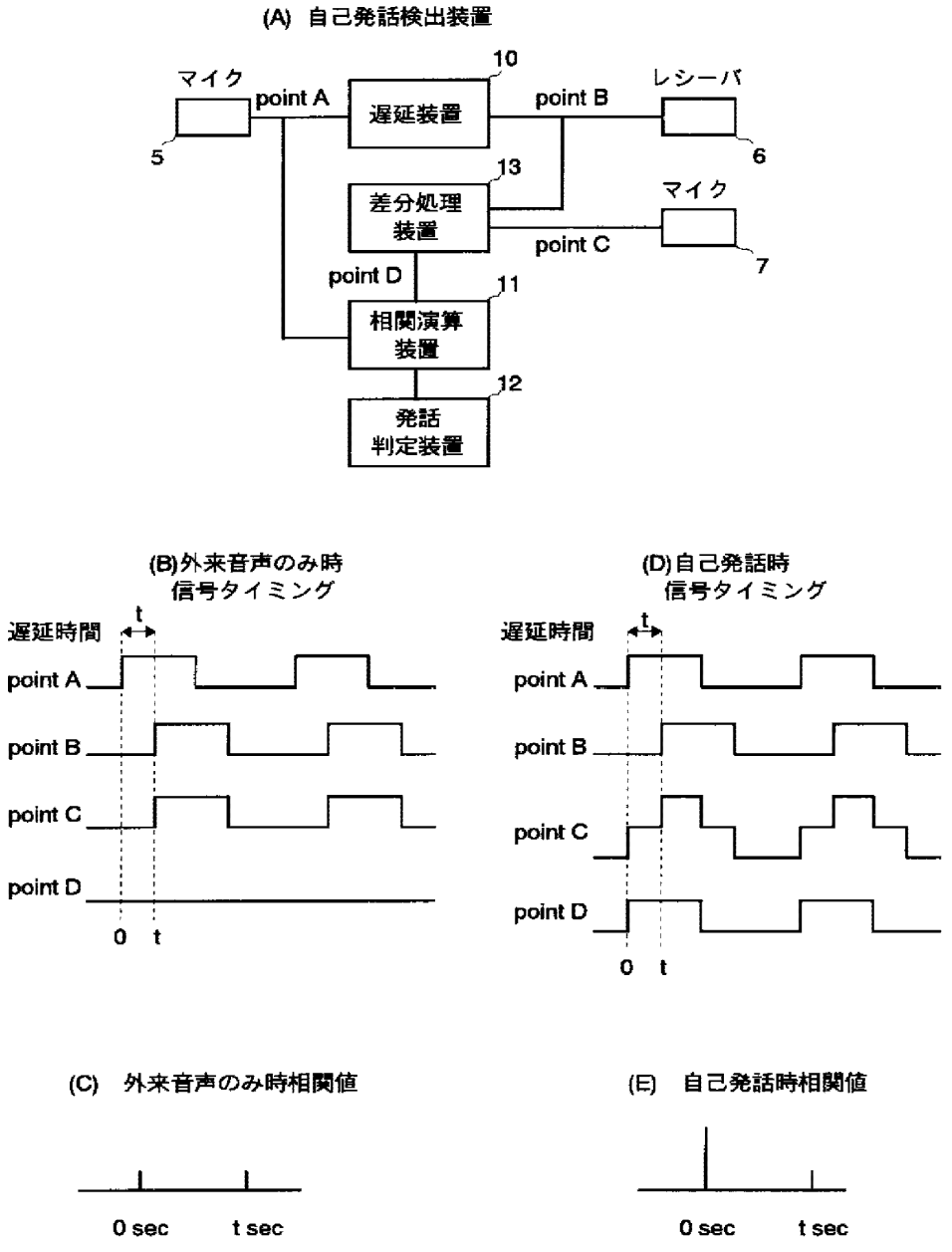
【図１】



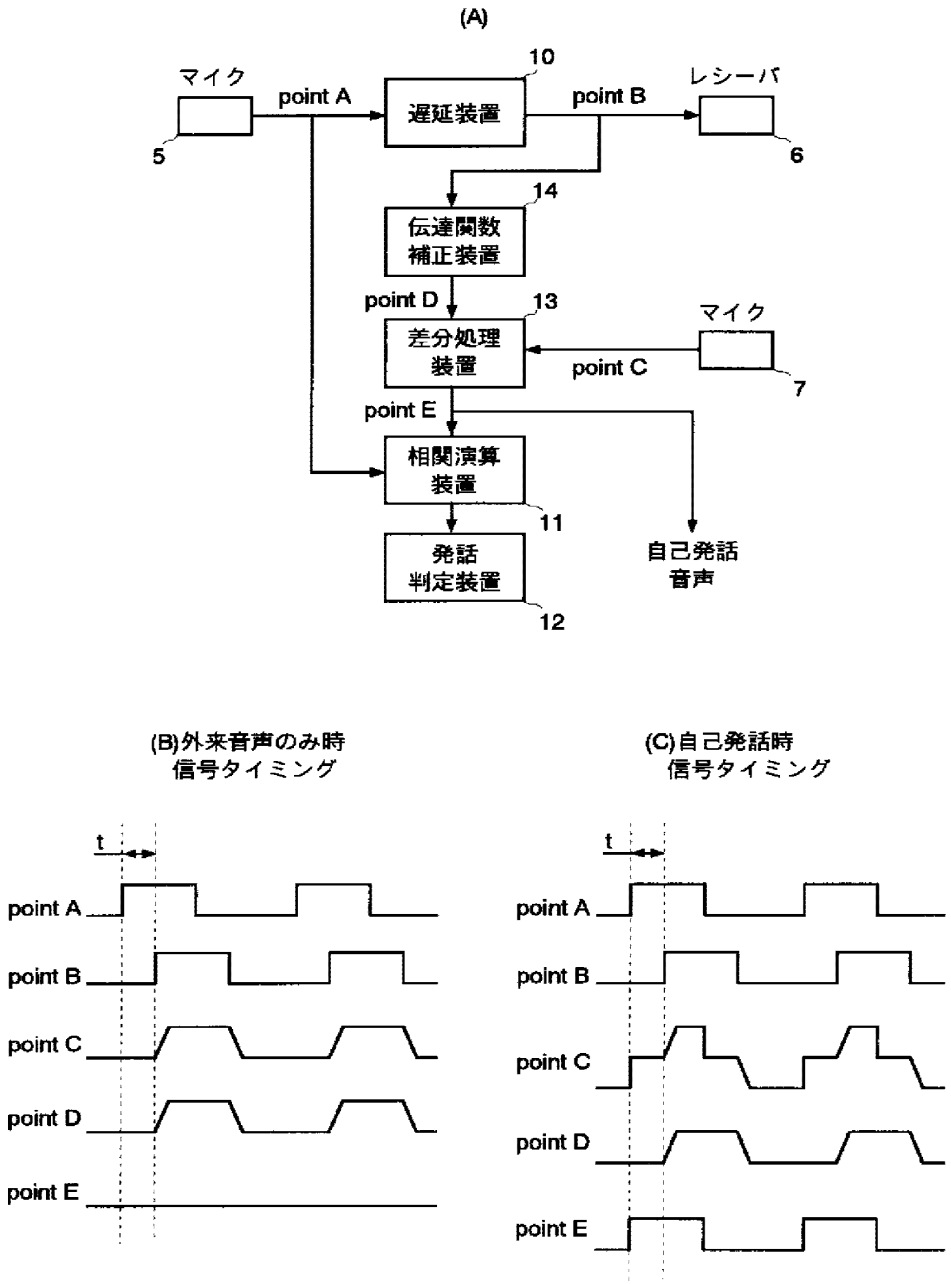
【図２】


(A)
マイク
point A
遅延装置
10
point B
レシーバ
5
6
相関演算
装置
point C
11
12
マイク
7
発話
判定装置
(B)外来音声のみ時
信号のタイミング
遅延時間
t
point A
point B
point C
(D)自己発話時
信号のタイミング
遅延時間
t
point A
point B
point C
(C)　相関値
0 sec
t sec
(E)　相関値
0 sec
t sec

【図３】

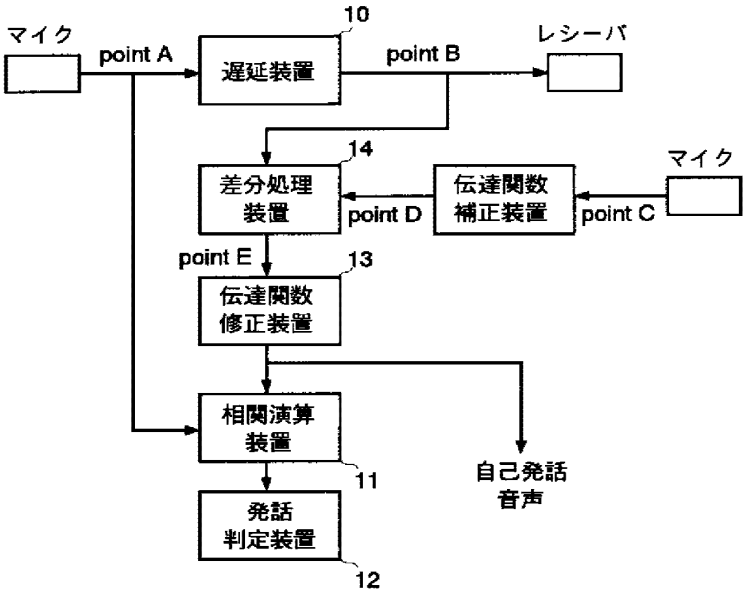

(A) 自己発話検出装置
マイク
point A
遅延装置
10
point B
レシーバ
5
6
差分処理装置
13
マイク
point C
7
point D
11
相関演算装置
12
発話判定装置
(B)外来音声のみ時 信号タイミング
遅延時間
t
point A
point B
point C
point D
0
t
(D)自己発話時 信号タイミング
遅延時間
t
point A
point B
point C
point D
0
t
(C) 外来音声のみ時相関値
0 sec
t sec
(E) 自己発話時相関値
0 sec
t sec

【図４】

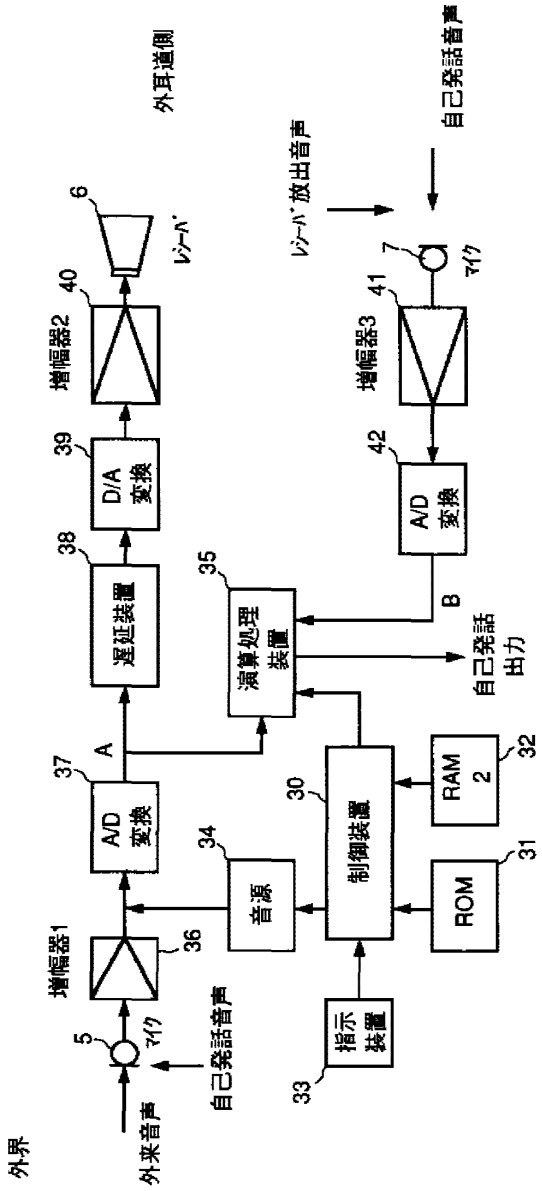

(A)
マイク
5
point A
10
遅延装置
point B
レシーバ
6
14
伝達関数
補正装置
point D
13
差分処理
装置
マイク
point C
7
point E
相関演算
装置
11
発話
判定装置
12
自己発話
音声
(B)外来音声のみ時
信号タイミング
t
point A
point B
point C
point D
point E
(C)自己発話時
信号タイミング
t
point A
point B
point C
point D
point E

【図５】

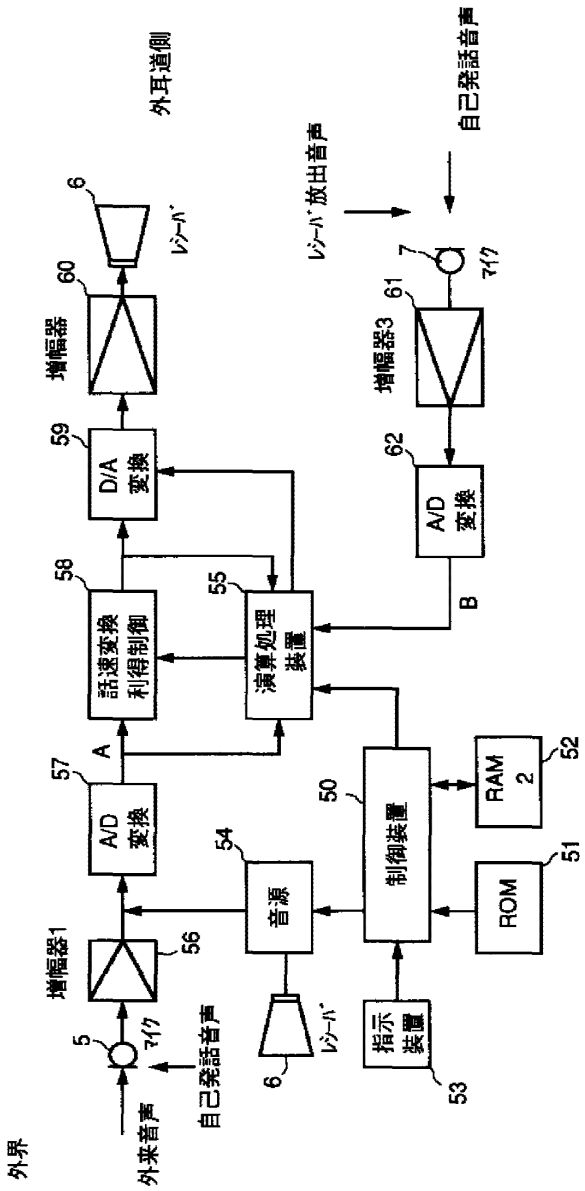
】


マイク
point A
10
遅延装置
point B
レシーバ
14
差分処理
装置
point D
伝達関数
補正装置
point C
マイク
point E
13
伝達関数
修正装置
相関演算
装置
11
自己発話
音声
発話
判定装置
12

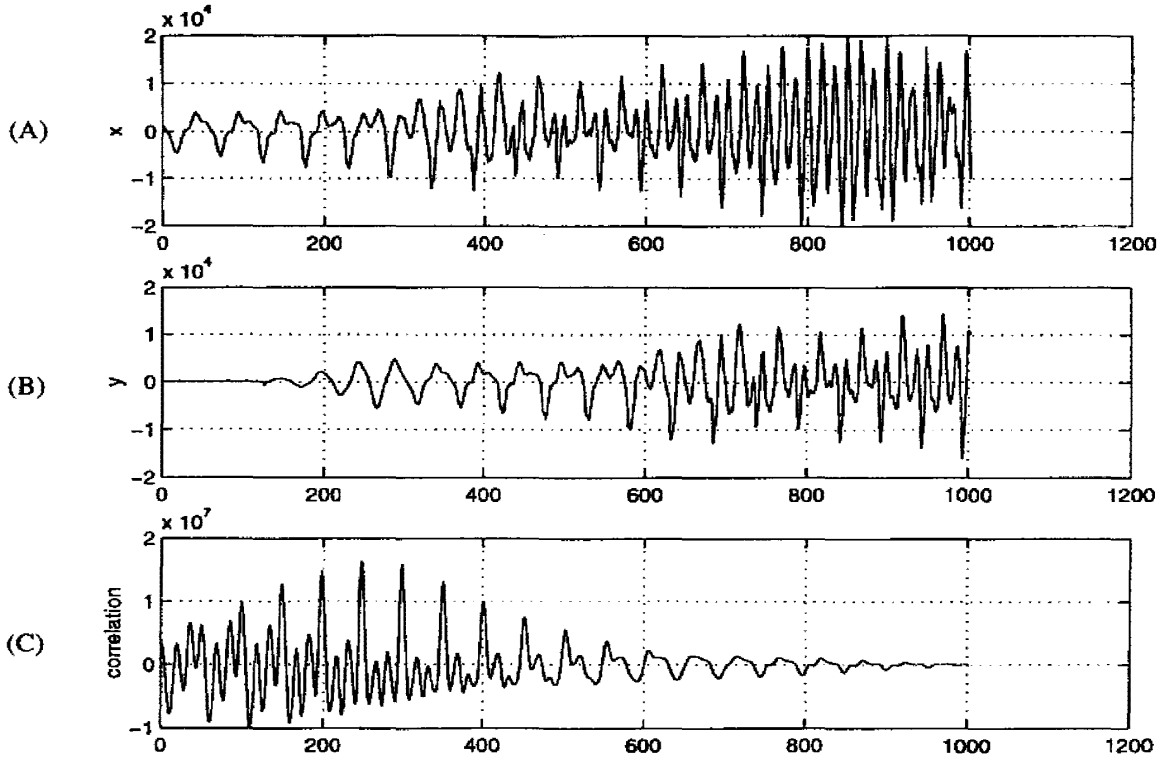
【図６】


外界
外来音声
マイク
5
自己発話音声
増幅器1
36
A/D 変換
37
A
遅延装置
38
D/A 変換
39
増幅器2
40
レシーバ
6
外耳道側
音源
34
制御装置
30
RAM 2
32
ROM
31
指示装置
33
演算処理装置
35
B
A/D 変換
42
増幅器3
41
マイク
7
レシーバ放出音
自己発話音声
自己発話出力

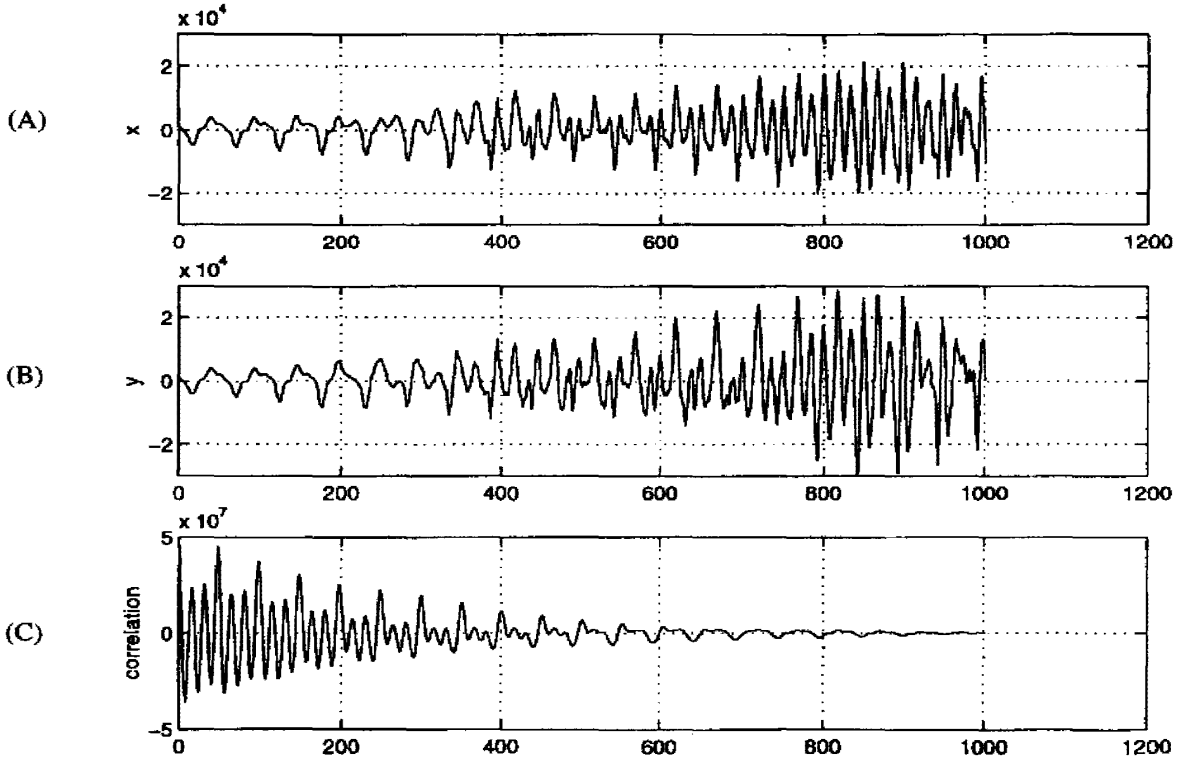
【図７】


外界
外来音声
マイク
5
自己発話音声
増幅器1
56
A/D 変換
57
A
D/A 変換
59
増幅器
60
レシーバ
6
外耳道側
音源
54
制御装置
50
RAM 2
52
ROM
51
指示装置
53
話速変換利得制御
58
演算処理装置
55
B
A/D 変換
62
増幅器3
61
マイク
7
レシーバ放出音
自己発話音声

【図８】


(A)
x 10^4
x
2
1
0
−1
−2
0
200
400
600
800
1000
1200
(B)
x 10^4
y
2
1
0
−1
−2
0
200
400
600
800
1000
1200
(C)
x 10^7
correlation
2
1
0
−1
0
200
400
600
800
1000
1200

【図９】



フロントページの続き

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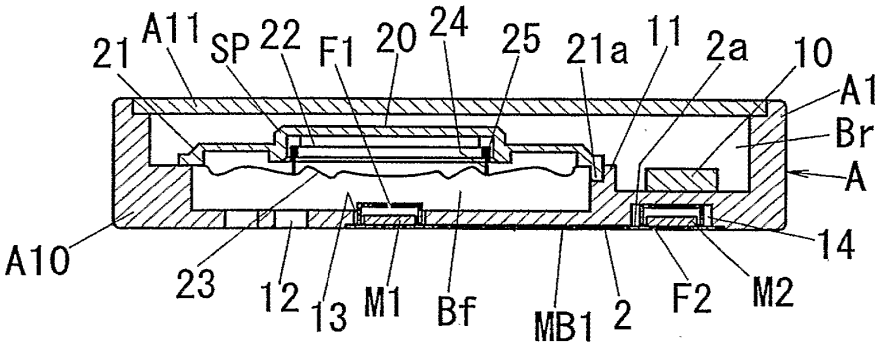
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(54) **Title:** INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME



(57) **Abstract:** A compact intercom device is provided, which efficiently prevents howling phenomenon. This intercom device includes a housing; a speaker accommodated in the housing such that a front air chamber is defined at a front side of the speaker and a rear air chamber is defined as a rear side of the speaker, a first microphone disposed such that its sound receiving surface faces the front air chamber, a second microphone disposed such that its sound receiving surface faces outside of the housing; and a signal processing unit. When an audio output of the speaker is picked up by the second microphone, the signal processing unit removes a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

WO 2008/050583 A1

DESCRIPTION

INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME

TECHNICAL FIELD

The present invention relates to an intercom device suitable for an interphone system or the like, and a wiring system using the same.

BACKGROUND ART

Conventionally, an interphone system has been widely used as short-distance conversation means between separated rooms or between an entrance and a room of a building or a house. A typical interphone system for household use is mainly formed with an indoor intercom device installed on a wall surface of the room, and an outdoor intercom device installed on a wall surface of the entrance, and connected to the indoor intercom device through a transmission line. For example, the outdoor intercom device has a microphone for receiving a visitor's voice, and a speaker for outputting a dweller's voice. By operating the outdoor intercom device, the visitor can have a conversation with the dweller using the indoor intercom device.

In this kind of intercom device, since the speaker is located near the microphone, there is an inconvenience that howling phenomenon easily occurs when the audio output of the speaker is picked up by the microphone. To solve this inconvenience, various measures or methods for preventing the howling phenomenon have been proposed.

For example, Japanese Patent Publication No. 2607257 discloses a loudspeaker communication device **300**, which is suitable for a handsfree telephone. As shown in FIG. 35A, this communication device **300** has a speaker **SP**, and a pair of microphones (**M1**, **M2**). In addition, to prevent the howling phenomenon caused when an audio output of the speaker **SP** is picked up by the microphone (**M1**, **M2**), the communication device **300** also includes a pair of amplifying circuits (**310**, **320**), a level-control amplifying circuit **330**, a

delay circuit **340**, and a differential amplifying circuit **350**, as shown in FIG. 35B. The delay circuit **340** delays an output of the microphone **M1** located closer to the speaker **SP** by a delay time, which is determined according to sound velocity and a difference between a distance **d1** between the microphone **M1** and the speaker **SP** and a distance **d2** between the microphone **M2** and the speaker **SP**. The level-control amplifying circuit **330** controls the output level of the microphone **M2** such that an amplification factor is **d2/d1**. In other words, the output level of the microphone **M2** is controlled so as to be substantially equal to the output level of the microphone **M1** with respect to the signal component corresponding to the audio output of the speaker **SP**. The differential amplifying circuit **350** receives an output of the delay circuit **340** and an output of the level-control amplifying circuit **330**, and provides an output signal corresponding to a difference between these outputs.

According to this communication device, the signal component corresponding to the audio output of the speaker **SP** is cancelled from the output of the microphone (**M1**, **M2**) by the differential amplifying circuit **350**. Therefore, it is possible to achieve a comfortable communication, while preventing the howling phenomenon.

However, as shown in FIG. 35A, since each of the microphones (M1, M2) is spaced from the speaker SP by a relatively large distance, there is another problem that the device itself becomes large in size. In addition, since the intercom device is usually installed on the wall surface or the like, it is strongly desired that the intercom device has a thin thickness. On the other hand, when the speaker is accommodated in an excessively thin housing, a volume of a rear air chamber, which is defined as a space of a rear side of the speaker, is reduced, so that a radiating sound pressure of the speaker lowers and a minimum resonance frequency of the speaker shifts to a higher frequency level. As a result, there is a fear that deteriorations in sound quality and electro-acoustic conversion efficiency of the speaker occur.

Thus, the conventional intercom device still has plenty of room for

improvement in downsizing of the device itself, sound quality, electro-acoustic conversion efficiency as well as the prevention of howling phenomenon.

SUMMARY OF THE INVENTION

Therefore, a primary concern of the present invention is to provide a compact intercom device, which has the capability of preventing howling phenomenon, and achieving an improvement in sound quality and efficiency.

That is, the intercom device of the present invention is characterized by comprising:

a housing;

a speaker accommodated in the housing such that a front air chamber is defined as a space enclosed by a front surface of the speaker and an inner surface of the housing, and a rear air chamber is defined as a space enclosed by a rear surface of the speaker and an inner surface of the housing;

a first microphone disposed such that its sound receiving surface faces the front air chamber;

a second microphone disposed such that its sound receiving surface faces outside of the housing; and

a signal processing unit configured to, when an audio output of the speaker is picked up by the second microphone, remove a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

According to the invention, since the first microphone easily and efficiently collects the audio information output from the speaker, howling phenomenon can be efficiently prevented by the signal processing unit. In addition, a distance between the first microphone and the speaker becomes short, thereby allowing downsizing of the intercom device. Furthermore, since the sound receiving surface of the second microphone faces the outside of the housing, an acoustic coupling between the speaker and the second microphone reduces, and the second microphone becomes hard to pick up the audio output of the

speaker. As a result, the second microphone can be located closer to the speaker. Thus, it is possible to provide a compact intercom device capable of preventing the howling phenomenon.

From the viewpoint of more efficiently preventing the howling phenomenon, it is preferred that the rear air chamber is the space hermetically-sealed by the rear surface of the speaker and the inner surface of the housing. In addition, it is preferred that the first microphone is arranged such that the sound receiving surface of the first microphone faces a diaphragm of the speaker. Moreover, when the first microphone and the second microphone are mounted on a same circuit board, it is preferred that the circuit board is mounted on an outside surface of the housing such that the sound receiving surface of the first microphone faces the front air chamber and the sound receiving surface of the second microphone faces the outside of the housing.

On the other hand, from the viewpoint of further downsizing the intercom device, the second microphone is preferably located adjacent to the speaker. In addition, it is preferred that the first and second microphones are mounted on a single board.

From the viewpoint of improving sound quality and electro-acoustic conversion efficiency of the speaker, it is preferred that the intercom device has an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end. Alternatively, the intercom device may have a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end. In addition, it is preferred that the acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of the speaker is needed.

Furthermore, it is preferred that the acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant frequency of the speaker that is supposed to be mounted on an infinite baffle and a lowest

resonant frequency of the speaker mounted on the rear air chamber with the absence of the acoustic tube. The acoustic tube can be formed along an inner surface of the rear air chamber. If necessary, a sound absorbing material may be disposed at the opening of the acoustic tube.

A further concern of the present invention is to provide a wiring system using the intercom device described above. That is, the wiring system comprises at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of the building structure, and connected to the transmission line. The intercom device has a first connector detachably connectable to a second connector formed in each of the base units. This wiring system is characterized in that when the first connector is connected to the second connector of one of the base units, the intercom device functions such that the information signals provided through the transmission line are output as audio information from the speaker unit, and an output of the signal processing unit is sent out through the transmission line.

By the way, in the intercom device installed on the wall surface of the house, one provided with a liquid crystal display or the like for displaying not only the audio information but also visual information such as images comes into practical use recently. Once such an intercom device is installed on the wall surface, not only electric wiring work but also mounting/repair work to the wall surface are required in a case where a layout of the intercom device has to be changed. However, it is not easy for a general user himself/herself to do the work. In addition, a function of the conventional intercom device is completed by itself, and a new-type intercom device is required in a case where it is desired to add another function. In such a case, not only a purchase of the new-type intercom device but also the above-described mounting/repair work becomes necessary, so that this is a great burden in cost for the user. The wiring system of the present invention is effect. According to the wiring system of the present

invention, since the intercom device is detachably connectable to the base unit, there advantages that the wiring system allows great flexibility for the change of layout of the intercom device, and makes it easy to replace the intercom device without cumbersome repairing work. Thus, the present invention provides the wiring system capable of solving the problems described above.

As a preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units, the function unit having a third connector detachably connectable to one of the second connector of the base unit, and an extended connector formed in the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, or when the first connector of the intercom device is connected to the second connector of the base unit, and the third connector of the function unit is connected to the extended connector of the intercom device, the function unit provides at least one of functions for supplying electric power provided through the transmission line, outputting information signals provided through the transmission line, and sending out information signals through the transmission line. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

As another preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units. The function unit has a third connector detachably connectable to the second connector of the base unit and an extended connector detachably connectable to the first connector of the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, and the extended connector of the function unit is connected to the first connector of the

intercom device, the function unit provides at least one of functions for supplying electric power provided from the transmission line through the base unit, outputting information signals provided from the transmission line through the base unit, and sending out information signals to the transmission line through the base unit, and the intercom device operates such that the information signals provided from the transmission line through the base unit and the function unit are output as audio information from the speaker unit, and the output of the signal processing unit is sent out to the transmission line through the base unit and the function unit. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

In the wiring system described above, it is preferred that the first and second connectors provide an electric power transmission between the base unit and the intercom device by means of electromagnetic coupling, and/or the first and second connectors provide a signal transmission between the base unit and the intercom device by means of optical coupling.

Further characteristics of the present invention and advantages brought thereby will be clearly understood from the best mode for carrying out the invention described below.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a perspective view of an intercom device according to a first embodiment of the present invention;

FIG. 2 is a cross-sectional view of the intercom device;

FIG. 3 is a circuit diagram of a voice processing unit of the intercom device;

FIG. 4 is a cross-sectional view of a microphone board of the intercom device;

FIG. 5 is a cross-sectional view of a bare chip employed in the microphone board;

FIG. 6A is a simplified plan view of the microphone board, and FIG. 6B is a simplified circuit diagram of the microphone board;

FIG. 7 is a circuit diagram of an impedance converting circuit of the microphone board;

FIG. 8 is a circuit diagram of a signal processing unit of the intercom device;

FIG. 9A and FIG. 9B are diagrams showing signal waveforms in the signal processing unit;

FIG. 10A and FIG. 10B are diagrams showing further signal waveforms in the signal processing unit;

FIG. 11A and FIG. 11B are diagrams showing another signal waveforms in the signal processing unit;

FIGS. 12A to 12C are diagrams showing still another signal waveforms in the signal processing unit;

FIG. 13 is a cross-sectional view of an intercom device according to a second embodiment of the present invention;

FIGS. 14A to 14C are cross-sectional and plan views of another housing of the intercom device according to the second embodiment;

FIG. 15 is a graph showing frequency characteristics of the cancelled amount of voice components of the speaker;

FIG. 16 is a graph showing frequency characteristics of the radiating sound pressure of the speaker;

FIG. 17 is a cross-sectional view of an intercom device using an ideal baffle plate;

FIG. 18 is a graph showing sound pressure characteristics of the intercom device having an acoustic tube and the intercom device having no acoustic tube;

FIG. 19 is a simplified cross-sectional view showing a modification of the housing having the acoustic tube;

FIG. 20 is a simplified cross-sectional view of a housing having a plurality of acoustic tubes;

FIG. 21 is a plan view of a housing having a sound absorbing material;

FIG. 22 is a schematic diagram of a wiring system using an intercom device according to a third embodiment of the present invention;

FIG. 23 is a schematic circuit diagram of a base unit of the wiring system;

FIG. 24 is a rear perspective view of the base unit;

FIG. 25A is a perspective view showing a gate housing and a main housing of the base unit, and FIG. 25B is a plan view of a module port of the gate housing;

FIG. 26 is a plan view of an attachment plate for mounting the base unit on a switch box;

FIG. 27 is a schematic circuit diagram of the function unit;

FIG. 28 is a perspective view of the intercom device for the wiring system;

FIG. 29 is a schematic circuit diagram of the intercom device for the wiring system;

FIG. 30 is a perspective view of the intercom device connected to the function unit or the base unit;

FIG. 31 is an exploded perspective view of the base unit;

FIG. 32A is a front view of the function unit, FIG. 31B is a side view of the function unit, FIG. 32C is an exploded side view of the function unit, and FIG. 32D is a perspective view of a joining member;

FIG. 33 is a perspective view illustrating how to mechanically couple the function unit to the base unit;

FIG. 34 is a schematic circuit diagram of an intercom device for a power line carrier type wiring system; and

FIG. 35A is a schematic perspective view of a conventional loudspeaker communication device, and FIG. 35B is a partial block diagram of the loudspeaker communication device.

BEST MODE FOR CARRYING OUT THE INVENTION

Referring to the attached drawings, an intercom device of the present invention and a wiring system using the intercom device are explained in detail

according to preferred embodiments.

(First Embodiment)

As shown in FIGS. 1 and 2, an intercom device **A** according to a first embodiment of the present invention includes a housing **A1**, a speaker **SP**, a microphone board **MB1**, and a voice processing unit **10**. The housing **A1** consists of a body **A10** having an opening at its back face, and a cover **A11** for closing the opening of the body **A10**, and accommodates the speaker **SP** and the voice processing unit **10** therein. The housing **A10** has a plurality of sound holes **12** formed in a front face of the body **A10**. The speaker **SP** accommodated in the housing **A1** provides a voice output through the sound holes **12** to outside of the housing **A1**. The microphone board **MB1** has a sound hole **F2** that picks up a voice emitted from a user of the intercom device **A**. In operation, when a conversation switch **SW1** is operated to the ON position, the intercom device **A** sends the picked-up voice to another intercom device **A** and receives signals from another intercom device **A** via an information line **Ls**.

As shown in FIG. 2, the speaker **SP** includes a cylindrical yoke **20** that is open at one end, and a round support **21** that extends radially outward from the open end of the yoke **20**. The yoke **20** is formed of an iron-containing material, such as a cold rolling steel plate (SPCC, SPCEN) or an electromagnetic soft iron (SUY), and has a thickness of about 0.8 mm. A columnar permanent magnet **22** (having a residual flux density of, for example, 1.39 T to 1.43 T) formed of, for example, neodymium is placed in the yoke **20**. An edge surface of the round support **21** is fixed on an edge portion on an outer peripheral side of a dome-shaped diaphragm **23**.

The diaphragm **23** is formed of a thermoplastic material, such as PET (PolyEthylene Terephthalate) or PEI (Polyetherimide), and has a thickness of, for example, 12 μm to 50 μm. A cylindrical bobbin **24** is fixed to the back of the diaphragm **23**, and a voice coil **25** is provided at the rear end of the bobbin **24**. The voice coil **25** is located at the open end of the yoke **20**, and the bobbin **24**

and the voice coil **25** are allowed to freely move back and forth (in the vertical direction in FIG. 2) in the vicinity of the open end of the yoke **20**. The voice coil **25** is formed by winding a polyurethane copper wire (having a diameter of 0.05 mm, for example) around a paper sleeve of craft paper.

When the polyurethane copper wire of the voice coil **25** receives a voice signal, an electromagnetic power is generated in the voice coil **25** by electric current of the voice signal and a magnetic field of the permanent magnet **22**, thereby to vibrate the bobbin **24**, along with the diaphragm **23**, forward and backward. As a result, a sound corresponding to the voice signal is emitted from the diaphragm **23**. That is to say, the speaker **SP** of this embodiment is an electrodynamic speaker. The speaker has, for example, a diameter of 20 mm to 25 mm, and a thickness of about 4.5 mm.

A rib **11** is formed on an inner surface of the front wall of the housing **A1**, which is opposed to the diaphragm **23**, and is in contact with an end face of a convex portion **21a** that protrudes frontward from an outer peripheral portion of the round support **21**. Thus, the speaker **SP** is fixed to the housing **A1** such that the diaphragm **23** faces the inner surface of the front wall of the housing **A1**.

With the speaker **SP** thus fixed within the housing **A1**, a front air chamber **Bf** is formed as a space enclosed by the inner surface of the front wall of the housing **A1** and a front surface (on which the diaphragm **23** is located) of the speaker **SP**, and a rear air chamber **Br** is formed as a space enclosed by an inner surface of the rear and side walls of the housing **A1** and the rear surface (on which the yoke **20** is located) of the speaker **SP**. The front air chamber **Bf** communicates with outside of the housing **A1** through the sound holes **12** formed in the front wall of the housing **A1**. The rear air chamber **Br** is insulated from (or does not communicate with) the front air chamber **Bf** since the end portion of the support **21** and the rib **11** are attached firmly to each other. Furthermore, the cover **A11** is tightly fitted in the opening formed in the rear wall of the body **A10**, so that the rear air chamber **Br** provides a

hermetically-sealed space that is insulated also from the outside of the housing **A1**. A gasket may be provided between the end portion (or the convex portion **21a**) of the support **21** and the rib **11** so as to increase the degree of adhesion between the support **21** and the rib **11**.

As shown in FIG. 3, the voice processing unit **10** is formed by an integral circuit (IC), which includes a communication unit **10a**, echo canceling units (**10b**, **10c**), an amplification unit **10d** and a signal processing unit **10e**. In operation, the communication unit **10a** receives a voice signal transmitted from an intercom device **A** installed, for example, in another room through the information line **Ls**. The received voice signal is transmitted through the echo canceling unit **10b** to the amplification unit **10d**, in which the signal is amplified, and then output from the speaker **SP**. When the conversation switch **SW1** is operated to the ON position, the voice communication function of the intercom device **A** becomes available. In this condition, voice signals respectively received from a microphone **M1** (first microphone) and a microphone **M2** (second microphone) on the microphone board **MB1** are processed by the signal processing unit **10e**. In the signal processing unit **10e**, each received signal is processed in the manner as described later, and the resulting signal passes through the echo canceling unit **10c**. Then, upon reception of the resulting signal from the echo canceling unit **10c**, the communication unit **10a** transmits the resulting signal through the information line **Ls** to the intercom device **A** installed in, for example, another room. In this manner, the intercom device **A** serves as an interphone that permits two-way voice communications between the rooms. An electric power for the intercom device **A** may be supplied from a wall socket disposed in the vicinity of the location where the intercom device **A** is installed. Alternatively, the electric power may be supplied through the information line **Ls**, or may be supplied from an internal battery accommodated in the intercom device **A**.

As shown in FIG. 4, the microphone board **MB1** has a module board **2** and the microphones **M1** and **M2**, which are mounted on a (top) surface **2a** of the

module board **2**. The microphone **M1** is mainly formed with a bare chip **BC1**, an IC chip **Ka1**, and a shield case **SC1**. Each of the bare chip **BC1**, the IC chip **Ka1**, and a wiring pattern (not shown) on the module board **2** are connected to each other with wires **W** (wire bonding). The shield case **SC1** covers the bare chip **BC1** and the IC chip **Ka1**. The microphone **M2** has the same construction as that of the microphone **M1**. That is to say, the microphone **M2** is formed with a bare chip **BC2**, an IC chip **Ka2**, and a shield case **SC2** mounted on the module board **2** that covers the bare chip **BC2** and the IC chip **Ka2**. The microphones (**M1**, **M2**) are preferred to be formed on a silicon substrate by a micro structure manufacturing process. That is, it is preferred that the microphones (**M1**, **M2**) are IC chips of Micro Electro Mechanical System (MEMS).

Each of the bare chips **BC1** and **BC2**, which will be generally referred to as "bare chip **BC**", provides a condenser-type silicon microphone, as shown in FIG. 5. More specifically, the bare chip BC includes a silicon thin film **1d** that is formed on a (top) surface of a silicon substrate **1b** so as to cover a cavity **1c** formed through the silicon substrate **1b**. An electrode **1f** is formed above the silicon thin film **1d** with an air gap **1e** interposed therebetween, and two pads **1g** are provided on the surface of the silicon substrate **1b** for producing an output in the form of a voice signal. When the silicon thin film **1d** is vibrated by a sound wave from outside, such as a human voice, a capacitance between the silicon thin film **1d** and the electrode **1f** changes, namely, the amount of electric charge changes. With the change in the electric charge amount, electric current proportional to the received sound wave flows from the two pads **1g**. Each of these bare chips **BC1** and **BC2** is fixed by die-bonding the silicon substrate **1b** on the module board **2**, as shown in FIG. 4. The silicon thin film **1d** of the bare chip **BC2** faces the sound hole **F2** formed in the module board **2**.

As shown in FIG. 4, a sound receiving surface of the microphone **M1** is positioned on a surface having the sound hole **F1** of the shield case **SC1**. On the other hand, a sound receiving surface of the microphone **M2** is positioned

on a surface having the sound hole **F2** of the module board **2**. Thus, the sound receiving surface of the microphone **M1** is to receive sound from an upper side of FIG. 4, and the sound receiving surface of the microphone **M2** is to receive sound from an opposite side, i.e., the lower side of FIG. 4. In brief, as shown in FIG. 2, the module board **2** is mounted to the housing **A1** such that the sound receiving surface of the microphone **M1** faces the diaphragm **23** of the speaker **SP**, and the sound receiving surface of the microphone **M2** faces outside of the housing **A1**. Since both of the microphones (**M1**, **M2**) are mounted on the same surface **2a** of the module board **2**, the thickness of the microphone board **MB1** can be advantageously reduced.

FIG. 6A is a simplified plan view of the module board **2**. The module board **2** has a substantially T-shape, which is composed of a rectangular portion **2f**, on which the microphone **M1** is mounted, a rectangular portion **2g**, on which the microphone **M2** is mounted, and a connecting portion **2h** that connects the rectangular portions **2f** and **2g**. The rectangular portion **2g** is formed to be larger than the rectangular portion **2f**. The rectangular portion **2g** has a negative power supply pad **P1**, a positive power supply pad **P2**, and output pads (**P3**, **P4**) on an edge portion of the rectangular portion **2g** such that each of the pads (**P1**, **P2**, **P3**, **P4**) forms a line along an end face of the edge portion of the rectangular portion **2g**.

As shown in FIG. 6B, the negative power supply pad **P1** is connected to a negative terminal of an external power supply, and the positive power supply pad **P2** is connected to a positive terminal of the external power supply. The electric power received from the pads (**P1**, **P2**) is supplied to the microphones (**M1**, **M2**) through the wiring patterns on the module board **2**. A voice signal corresponding to the voice picked up by the microphone **M1** is output from the output pad **P3** via the wiring pattern on the module board **2**. On the other hand, a voice signal corresponding to the voice picked up by the microphone **M2** is output from the output pad **P4** via the wiring pattern on the module board **2**. The negative power supply pad **P1** also serves as a ground pad that establishes

a ground for the voice signals delivered from the output pads (**P3**, **P4**).

As described above, electric powers needed to activate the microphones (**M1**, **M2**) are supplied from the power supply via the common negative and positive power supply pads (**P1**, **P2**). The negative power supply pad **P1** also serves as the ground pad for the outputs of microphones (**M1**, **M2**). Therefore, the number of pads used in the microphone board **MB1** can be reduced, and the construction of the microphone board **MB1** can be simplified.

Next, an operation of the microphone board MB1 will be explained. First, electric currents flowing from the bare chips **BC1** and **BC2** corresponding to the received sound signal are subjected to impedance-conversion, and converted to voltage signals by the IC chips **Ka1** and **Ka2**. Then, the voltage signals are output as voice signals from the output pads (**P3**, **P4**), respectively.

For example, each of the IC chip **Ka1** and the IC chip **Ka2**, which will be generally referred to as "IC chip **Ka**", has a circuit configuration, as shown in FIG. 7. The IC chip **Ka** has a constant-voltage circuit **Kb** formed of a chip IC that converts a power supply voltage +V (e.g., 5V) supplied from the power supply pads (**P1**, **P2**) to a constant voltage Vr (e.g., 12V). The constant-voltage circuit **Kb** applies the constant voltage Vr to a series circuit of a resistance **R11** and the bare chip **BC**. A connection midpoint of the resistance **R11** and the bare chip **BC** is connected to a gate terminal of a junction-type J-FET **S11** through a capacitor **C11**. A drain terminal of the J-FET **S11** is connected to an operating voltage +V, and a source terminal is connected through a resistance **R12** to the negative terminal of the power supply. Herein, the J-FET **S11** is intended for converting electrical impedance, and the voltage of the source terminal of the J-FET **S11** produces an output as the voice signal. It is to be understood that the impedance converting circuit of the IC chip **Ka** is not limited to the above-described configuration, and may be replaced by a circuit having a function of a source follower circuit by an operation amplifier. For example, an amplifying circuit for the voice signal may be provided in the IC chip **Ka**, if necessary.

In the microphone board **MB1**, on which signal transmission and supply of electric power are performed via the wiring patterns on the module board **2** as described above, the signal lines and the power-supply lines can be arranged with high efficiency, and attached to the outer surface of the housing **A1**, as shown in FIG. 1. In this embodiment, as shown in FIG. 1, the surface **2a** of the module board **2** is attached along an outer surface of the front wall of the housing **A1**, and the microphone **M1** is inserted in an opening **13** formed in the front wall of the housing **A1** such that the sound receiving surface of the microphone **M1** faces the front air chamber **Bf**. The diaphragm **23** faces the microphone **M1** through the sound hole **F1** formed in the shield case **SC1**. With this arrangement, the microphone **M1** has a high directivity to the voice emitted from the speaker **SP** via the sound hole **F1**, so that the voice emitted from the speaker **SP** can be efficiently picked up by the microphone **M1**. The microphone **M2** fits in a recess **14** formed in the front wall of the housing **A1**, and faces the outside (or the front) of the housing **A1** through the sound hole **F2** formed in the module board **2**. In this regard, a direction where the second microphone **M2** receives a voice from the outside of the housing **A1** is substantially equal to the direction of emitting a voice from the speaker **SP**. Thus, the microphone **M2** has a high directivity to the voice of a talker who is located in front of the intercom device **A**. When **X1**, **X2** represent distances from the center of the speaker **SP** to the centers of microphones (**M1**, **M2**), respectively, the relationship between the distances (**X1**, **X2**) is expressed as **X1** $<$**X2**.

Since the rear air chamber **Br** facing the rear surface of the speaker **SP** is hermetically-sealed in the housing **A1** as shown in FIG. 2, the voice emitted from the rear surface of the speaker **SP** (rear surface of the diaphragm **23**) is less likely to leak from the rear air chamber **Br**, so that an acoustic coupling between the speaker **SP** and the microphone **M2** is lowered. BY the way, the voice emitted from the rear surface of the speaker **SP** is reversed in phase with respect to the voice emitted from the front surface of the speaker **SP** (front surface of the

diaphragm **23**). Therefore, if the voice emitted from the rear surface of the speaker **SP** goes around to the front of the speaker **SP**, these voices emitted from the front and rear surfaces of the speaker **SP** are cancelled with each other. As a result, the radiating sound pressure of the speaker **SP** is lowered, and it becomes difficult for the talker located in front of the intercom device **A** to hear the voice (sound) emitted from the speaker **SP**. However, since the intercom device of this embodiment is formed to reduce the leaking of the voice emitted from the rear surface of the speaker **SP** to the outside of the housing **A1**, as described above, it is possible to prevent such a reduction in the radiating sound pressure, which is caused by the voice emitted from the rear surface of the speaker **SP** going around to the front of the speaker **SP**.

As shown in FIG. 2, since the recess **14**, in which the microphone **M2** is inserted, is an insulated space not communicated with the rear air chamber **Br**, the microphone **M2** is even less likely to pick up the voice (or sound) emitted from the speaker **SP**. Therefore, the acoustic coupling between the speaker **SP** and the microphone **M2** can be further reduced. With the above arrangement, the microphones (**M1**, **M2**) are adapted to separately pick up the voice emitted from the speaker **SP** and the voice of (generated by) the talker, respectively.

Furthermore, if the microphone board **MB1** is placed in the housing **A1**, it is difficult to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**. In this embodiment, however, the microphone board **MB1** is attached to the outer surface of the housing **A1**. Therefore, it is possible to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**.

As described below, the intercom device of this embodiment has a configuration to prevent a howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**.

As shown in FIG. 8, the signal processing unit **10e** of the voice processing unit **10** includes an amplifying circuit **30** for amplifying the output of the microphone **M1**, a band-pass filter **31** for removing noises of frequencies other

than a voice-band (300 to 4000 Hz) from the output of the amplifying circuit **30**, and a delaying circuit **32** for delaying the output of the band-pass filter **31**. The signal processing unit **10e** further includes an amplifying circuit **33** for amplifying the output of the microphone **M2** while reversing the phase of the output of the microphone **M2** by 180 degrees, a band-pass filter **34** for removing noises of frequencies other than the voice-band (300 to 4000 Hz) from the output of the amplifying circuit **33**, and an adder circuit **35** for adding the outputs of the delaying circuit **32** and the band-pass filter **34**.

Referring to FIG. 8 to FIG. 12C, operations of the signal processing unit **10e** will be explained below. First, an operation of the signal processing unit **10e** when the voice emitted from the speaker **SP** is picked up by the microphones (**M1**, **M2**) will be explained. FIGS. 9A and 9B, 10A and 10B, 11A and 11B, and 12A to 12C show voice signal waveforms that appear at respective units of the signal processing unit **10e** when the microphones (**M1**, **M2**) respectively pick up the voice emitted from the speaker **SP**. As mentioned above, the distance **X1** from the center of the speaker **SP** to the center of the microphone **M1** is smaller than the distance **X2** from the center of the speaker **SP** to the center of the microphone **M2**. Accordingly, as shown in FIGS. 9A and 9B, when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**, the amplitude of the output **Y21** of the microphone **M2** is smaller than that of the output **Y11** of the microphone **M1** due to the difference between the distances (**X1**, **X2**). In addition, the phase of the output **Y21** of the microphone **M2** is delayed with respect to that of the output **Y11** of the microphone **M1** by a delay time **Td** (=(**X2**-**X1**)/**Vs**, where **Vs** is sonic velocity), which is proportional to the difference between the distances (**X2**-**X1**).

The amplifying circuit **30** amplifies the output **Y11** to produce an output **Y12**, and the amplifying circuit **33** amplifies the output **Y21** while reversing the phase thereof by 180 degrees, to produce an output **Y22**. At this time, each of the amplifying circuits (**30**, **33**) performs level adjustment based on the difference between the distances (**X2**-**X1**), so that the output levels of the

microphones (**M1**, **M2**) relative to the voice emitted from the speaker **SP** become equal to each other, as shown in FIGS. 10A and 10B. In this embodiment, since an amplitude rate of the amplifying circuit **30** is set to substantially 1, the amplifying circuit **30** may be eliminated from the signal processing unit **10e**.

As shown in FIGS. 11A and 11B, the band-pass filters (**31**, **34**) remove frequencies other than the voice band from the outputs **Y12** and **Y22**, to produce outputs **Y13** and **Y23**, respectively.

The delaying circuit **32** can be formed by, for example, a time delay device or a CR phase delaying circuit. The delaying circuit **32** delays the output of the microphone **M1**, which is located closer to the speaker **SP**, by the above-described delay time **Td**, thereby producing an output **Y14** whose phase coincides with the phase of the output **Y23** of the band-pass filter **34**. Thus, it is possible to reduce noises contained in a voice signal to be transmitted.

Through the amplifying and delaying processes described above, a voice component emitted from the speaker **SP**, which is contained in the output **Y14**, and the voice component emitted from the speaker **SP**, which is contained in the output **Y23**, have the same amplitude and the reversed phase relative to each other, as shown in FIG. 12A and FIG. 12B. The adder circuit **35** adds the output **Y14** and the output **Y23** to cancel the voice component of the speaker **SP** from the output of the microphone **M2**. Thus, it is possible to produce an output **Ya** shown in FIG. 12C, in which the voice component from the speaker **SP** is removed or reduced. As described above, the signal processing unit **10e** removes the voice component of the speaker **SP** from the output signal **Y21** of the microphone **M2** by using the output signal **Y11** of the microphone **M1**.

On the other hand, an operation of the signal processing unit **10e** when the voice emitted from the talker **H** in front of the intercom device **A** is picked up by the microphones (**M1**, **M2**) will be described below. Since the microphone **M2** has a high directivity to the voice from the talker **H** located in front of the intercom device **A**, the amplitude of the output **Y21** of the microphone **M2** is larger than that of the output **Y11** of the microphone **M1**. Furthermore, since

the signal from the microphone **M1** is not amplified while the signal from the microphone **M2** is amplified in the amplifying circuit **33**, the voice component from the talker **H** contained in the output **Y23** is sufficiently larger than the voice component from the talker **H** contained in the output **Y14**. Thus, even after the above-described adding process is performed in the adder circuit **35**, the signal corresponding to the voice generated by the talker **H** remains in the output **Ya** with a sufficiently large amplitude.

In the manner described above, the voice component from the speaker **SP** is reduced in the output **Ya** of the adder circuit **35**, whereas the voice component generated by the talker **H** in front of the intercom device **A** remains in the output **Ya**. Thus, a relative difference in the output **Ya** between the voice component from the talker **H**, which is desired to be saved, and the voice component from the speaker **SP**, which is desired to be reduced, becomes larger. That is to say, even when the voice from the talker **H** and the voice from the speaker **SP** are simultaneously generated, only the voice component from the speaker **SP** is reduced while the voice component from the talker **H** remains sufficiently large. Accordingly, it is possible to prevent the howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice output of the speaker **SP**.

Since the microphone **M1** is disposed in the front air chamber **Bf** so as to face the diaphragm **23**, as shown in FIG. 2, the microphone **M1** efficiently picks up the voice emitted from the speaker **SP**. Therefore, the signal processing unit **10e** can efficiently achieve the howling phenomenon prevention function. Furthermore, since the sound receiving surface of the microphone **M2** faces outside of the housing **A1**, and the output direction of the speaker **SP** and the directivity of the microphone **M2** substantially conform to each other, the acoustic coupling between the speaker **SP** and the microphone **M2** is reduced. Thus, the microphone **M2** is less likely to pick up the voice emitted from the speaker **SP**, and can be placed close to the speaker **SP**, that is to say, close to the front air chamber **Bf**, which makes it possible to downsize the intercom device **A**.

On the other hand, the signal processing unit **10e** transmits the voice signal to the echo canceling unit **10c** through a digital-analog converter (not shown), and the echo canceling unit (**10b**, **10c**) shown in FIG. 3 perform the process as described below to further efficiently prevent the howling phenomenon.

First, the echo canceling unit **10c** retrieves the output of the echo canceling unit **10b** as a reference signal, and performs a suitable computation on the output of the signal processing unit **10e**, thereby canceling a voice signal, which goes around from the speaker **SP** to the microphones (**M1**, **M2**). On the other hand, the echo canceling unit **10b** also retrieves the output of the echo canceling unit **10c** as the reference signal, and performs a suitable computation on the output of the communication unit **10a**, thereby canceling a voice signal, which goes around from a speaker of an intercom device installed in another room to its microphones.

More specifically, the echo canceling units (**10b**, **10c**) control the amount of loss at variable loss means (not shown) provided in a loop circuit consisting of the speaker **SP**, the microphones (**M1**, **M2**), the signal processing unit **10e**, the echo canceling unit **10c**, the communication unit **10a**, the echo canceling unit **10b**, the amplifying unit **10d** and the speaker **SP**, so as to prevent the howling phenomenon by making the loop gain less than 1. Herein, one of the transmitted voice signal and the received voice signal, which has a smaller signal level, is regarded as not important, and a transmission loss of a variable loss circuit, which is inserted in a transmission path of the signal having the smaller level, is increased. In this embodiment, the intercom devices transmit signals via the information line **Ls**. However, the transmission of signals is not limited to a wired line. A non-wired transmission manner such as radio wave, optical communication system, and infrared radiation may be alternatively employed.

(Second Embodiment)

As shown in FIG. 13, an intercom device **A'** according to a second

embodiment of the present invention includes an acoustic tube **40**, which is provided in the rear air chamber **Br** having a small volume, so as to improve the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. The intercom device **A'** according to the second embodiment will be explained below.

The acoustic tube **40** has an opening at its one end and a closed end at the other end, and is provided in the rear air chamber **Br** to be along an inner wall surface(s) of the rear air chamber **Br**. By setting the resonance frequency of the acoustic tube **40** to an appropriate level, the lowest resonance frequency **fo** of the speaker **SP**, which is provided in the intercom device **A'**, is shifted to a lower frequency level, and the sound pressure level of the speaker **SP** increases, resulting in improvements of the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. It is noted that except for the sound tube **40**, the construction of the intercom device **A'** shown in FIG. 13 is the same as that of the intercom device **A** shown in FIG. 2. Accordingly, details of the intercom device **A'** will be omitted herein.

The lowest resonance frequency **fo** of the speaker **SP** is generally determined by an equivalent mass (diaphragm, coil, additional mass of air) **Mo** of a vibration system of the speaker, stiffness **So** of an edge or the like supporting the vibration system, and stiffness **Sc** of air in the rear air chamber **Br**. That is, it is represented by an equation:

$$fo = \{1/(2\pi)\} \cdot \sqrt{\{(So+Sc)/Mo\}}$$

FIG. 14A through FIG. 14C show another housing **A2** of the intercom device **A'** according to a modification of the present embodiment. The housing **A2** accommodates the acoustic tube **40** therein. Effects of the sound tube **40** are explained below referring to the housing **A2** as an example. The housing **A2** essentially consists of a body **A20** having opening at its rear surface, and a cover **A21** for closing the opening of the body **A20**. In FIG. 14B, only the body **A20** is illustrated. As in the intercom device **A** of the first embodiment, the housing **A2** is capable of accommodating a speaker **SP**, a microphone board **MB1**, a

conversation switch **SW1**, and a voice processing unit **10**. The acoustic tube **40** is provided in the body **A20**.

The housing **A2** has two spaces **A2a** and **A2b** therein, which are insulated from each other. The space **A2a** has a lateral dimension of 30 mm, a vertical dimension of 40 mm, and a thickness of 8 mm, to accommodate the speaker **SP** therein. In the space **A2a**, columnar bosses **51** are formed at regularly spaced four locations on the inner surface of the front wall of the body **A20**, and screw holes **52** are respectively formed in the bosses **51**. The speaker **SP** is fixed to the inner surface of the front wall of the body **A20** by using mounting screws, which are thread engageable with the screw holes **52**. With the construction described above, a front air chamber **Bf** and a rear air chamber **Br** similar to those of the first embodiment (FIG. 2) are formed. The front air chamber **Br** is provided by a space enclosed by the inner surface of the front wall of the housing **A2** and the front surface side (diaphragm **23** side) of the speaker **SP**. The rear air chamber **Br** is provided by a space enclosed by the inner surface of the rear and side walls of the housing **A2** and the rear surface side (yoke **20** side) of the speaker **SP**. A plurality of sound holes **12** are formed in the front wall of the housing **A2**.

As shown in FIGS. 14A through 14C, the acoustic tube **40**, which has a rectangular shape in cross section, is formed along the inner walls of the rear air chamber **Br** at the back of the speaker **SP** over substantially three-quarter of the circumference of the rear air chamber **Br**. The acoustic tube **40** has an opening at one end **40a** and a closed end at the other end **40b**. The overall length **L** of the acoustic tube **40** is set to an odd multiple of one quarter of a wavelength, which is determined by the sonic velocity and a frequency **f** of which sound pressure level is desired to be increased. In this embodiment, the acoustic tube **40** has a length of one quarter of the wavelength determined by the frequency where an increase in sound pressure level of the audio output of the speaker **SP** is needed and the sonic velocity. In detail, since an open-end correction is performed on the acoustic tube **40**, where sonic velocity is set to C

and an open-end correction value is set to σ (=0.8**d**, wherein **d** is an open-end diameter of the acoustic tube **40**), the overall length is determined from an equation described below:

$$f = [C / \{4 (L+\sigma)\}]$$

In this embodiment, the overall length L of the acoustic tube **40** is set 95.2 mm and a cross-sectional area of the acoustic tube **40** is set to 4.2 mm^2 (equivalent to $\phi = 2.3$). By forming the acoustic tube **40** along the inner walls of the rear air chamber **Br**, it is possible to minimize a volume reduction of the rear air chamber **Br** caused by the formation of the acoustic tube **40**. In addition, with the acoustic tube **40** thus provided, the sound pressure level of the speaker **SP** can be increased in a desired frequency band (at around 700 Hz in this embodiment). As a result, it is possible to improve the sound quality and the electro-acoustic conversion efficiency of the speaker **SP**.

By utilizing the fact that input impedance becomes extremely small at the resonance frequency (i.e., a frequency at which the overall length of the tube is equal to an odd-number multiple of an approximately quarter (1/4) wavelength) of a closed tube, the acoustic tube **40** can reduce the sound leakage from the rear air chamber **Br** to the outside of the housing. That is, the sound waves radiated from the rear surface of the speaker **SP** is reflected toward the rear surface side of the speaker **SP**.

Referring to FIG. 15 through FIG. 18, effects of the acoustic tube **40** provided in the housing **A2** will be explained below in comparison with the intercom device **A** of the first embodiment, which has no acoustic tube in the housing **A1**, and an intercom device having an infinite baffle, which will be explained in detail below.

FIG. 15 shows frequency characteristics of the cancelled amount of the voice component from the speaker **SP** in the case of using the signal processing unit **10e** of intercom device **A** of the first embodiment, and FIG. 16 shows frequency characteristics of the radiating sound pressure measured in front of the speaker **SP**. FIG. 15 and FIG. 16 also show results obtained in the case

where the housing has no acoustic tube as in the first embodiment, and the case where an ideal baffle plate is used as the housing.

FIG. 17 shows a baffle plate **C** having an infinite size as an example of the above-mentioned ideal baffle plate, but the baffle plate **C** is not realistic for mounting a real speaker thereon. Thus, in this embodiment, a standard baffle as specified in JIS C 5532 is used to measure a frequency equivalent to a lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C**. The lowest resonant frequency of the speaker **SP** mounted on the standard baffle is equivalent to that of the speaker **SP** that is supposed to be mounted on the baffle plate **C** (an ideal baffle plate). The cancelled amount (**Y1a** in FIG. 15) in the case where the speaker **SP** and the microphone board **MB1** are mounted on the standard baffle to form an intercom device is kept at 10 dB or more in a frequency band of 100 Hz to 10000 Hz, and the radiating sound pressure characteristics (**Y2a** in FIG. 16) in the same case indicate that the lowest resonance frequency **fo1** of the speaker **SP** is 600 Hz, which is an equally ideal characteristic as provided by the speaker **SP** alone. A standard closed box as specified in JIS C 5532 may be used to measure a frequency equivalent to the lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C** having an infinite size (an ideal baffle plate) as shown in Fig. 17.

In the case where a rear air chamber having no acoustic tube **40** is used, as in the rear air chamber **Br** of the first embodiment, the cancelled amount (**Y1b** in FIG. 15) of the voice component from the speaker **SP** is kept at 10 dB or more in the frequency band of 100 Hz to 10000 Hz. Thus, substantially the same cancelled amount as that of the case where the standard baffle is used can be obtained. However, in the case where the rear air chamber **Br** is formed as an enclosed space, radiating sound pressure characteristics comparable to those of the case where the standard baffle is used can be obtained when the volume of the rear air chamber **Br** is sufficiently larger. On the other hand, when the volume of the rear air chamber **Br** is small, the mechanical equivalent stiffness

of the rear air chamber **Br** increases, and the lowest resonance frequency of the speaker **SP** becomes higher. As higher the lowest resonance frequency, the radiating sound pressure reduces, so that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** deteriorate. For example, the radiating sound pressure characteristics (**Y2b** in FIG. 16) of the case where the rear air chamber **Br** of the first embodiment is used indicate that the lowest resonance frequency **fo2** of the speaker **SP** is 1200 Hz, which is noticeably higher than the lowest resonance frequency obtained when the standard baffle is used. Furthermore, the sound pressure level in a frequency band below 800 Hz is lower by about 5 to 20 dB than that of the case where the standard baffle is used. It means that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are deteriorated in this frequency band.

The radiating sound pressure characteristics (**Y2c** in FIG. 16) of the case where the volume of the rear air chamber **Br** is made three times larger than that of the rear air chamber **Br** of the first embodiment indicate that the lowest resonance frequency **fo3** of the speaker **SP** is 800 Hz, and the sound quality of the speaker **SP** is improved as compared with the case of using the housing **A1** of the first embodiment. However, the overall size of the housing increases with the increase in volume of the rear air chamber **Br**, which makes it difficult to downsize the intercom device. Thus, in this embodiment, by providing the acoustic tube **40** in the rear air chamber **Br** having a small volume, the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are improved, while making the intercom device smaller.

The overall length **L** of the acoustic tube **40** is set so that the resonance frequency of the acoustic tube **40** is located between the lowest resonance frequency **fo2** (=1200 Hz) of the case where the rear air chamber **Br** having no acoustic tube **40** is used, and the lowest resonance frequency **fo1** (=600 Hz) of the case where the standard baffle is used. By setting the overall length **L** of the acoustic tube **40**, as described above, the lowest resonance frequency **fo5** of

the speaker **SP** in the case where the housing **A2** having the acoustic tube **40** is used is set to about 700 Hz between **fo1** and **fo2**.

FIG. 18 shows frequency characteristics of the radiating sound pressure of the intercom device **A'**, which is provided with the housing **A2** and the sound tube **40** of the present embodiment, and the radiating sound pressure of the intercom device **A** of the first embodiment, which has no acoustic tube. The characteristics **Y3a** of the intercom device **A'** in FIG. 18 indicate that the sound pressure level in a frequency band not larger than 800 Hz is increased by about 5 to 10 dB, as compared with the characteristics **Y3b** of the intercom device **A**, and the lowest resonance frequency is reduced from **fo4** (=1200 Hz) to **fo5** (= about 700 Hz). That is, improvements in the speaker efficiency and the sound quality can be achieved. For example, the sound pressure level at around 700 Hz is increased by about 10 dB, and the improvement of the speaker efficiency brought by the increase of the sound pressure level corresponds to a 70% reduction in input current and a 90% reduction in input power of the speaker **SP**. Thus, the sound pressure level of the speaker **SP** increases around the lowest resonance frequency of the speaker **SP**. Therefore, by setting the overall length of the acoustic tube **40** to an appropriate length, and shifting the lowest resonance frequency of the speaker **SP** to a lower frequency side, it is possible to achieve improvements in sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As another preferred acoustic tube of the present invention, an acoustic tube **41** that communicates with the rear air chamber **Br** may be provided at the outside of a housing **A3**. For example, the acoustic tube **41** can be obtained by attaching a tubular member to into a hole (not shown) formed in the housing **A3**, as schematically shown in FIG. 19. When tubular member is made of urethane or the like, the overall length **L** of the acoustic tube **41** can be easily changed, if desired.

As a further modification of this embodiment, two or more acoustic tubes may be formed in the rear air chamber **Br**. As an example, three acoustic tubes

42, **43**, **44** can be formed in the rear air chamber **Br** of a housing **A4**, as schematically shown in FIG. 20. In this example, the acoustic tubes (**42**, **43**, **44**) have different lengths from each other. This means that they have mutually different resonance frequencies. Accordingly, the sound pressure level of the speaker **SP** is increased at a plurality of frequencies. By setting the overall lengths of the respective acoustic tubes (**42**, **43**, **44**) to appropriate values, it is possible to provide desired sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As shown by a dotted line in FIG. 21, a sound absorbing material **45** such as a nonwoven fabric may be provided at or in the vicinity of the open end **40a** of the acoustic tube **40**. By use of the sound absorbing material **45**, it is possible to provide an enhanced sound absorbing effect, and perform a fine adjustment of the resonance frequency of the acoustic tube **40**.

(Third Embodiment)

The present embodiment provides a wiring system using an intercom device **101** of the present invention. That is, this wiring system enables power and signal transmissions between electric devices spaced away from each other in a building structure.

As shown in FIG. 22, the wiring system comprises a power line **Lp** and an information line **Ls** installed in the building structure, which are connected to a commercial power source **AC** and the Internet network **NT** through a distribution board, switch boxes **102** embedded in wall surfaces of the building structure, base units **103** mounted on the switch boxes, and connected to the power line **Lp** and the information line **Ls**, and function units **104**, each of which is detachably connected to a desired one of the base units **103**, and has at least one of functions for supplying electric power from the power line **Lp**, outputting information from the information line **Ls**, and inputting information into the information line **Ls** in a connected state with the desired base unit. An intercom device **101** of the present invention, which is preferably provided by

the intercom device (**A**, **A'**) of the first or second embodiment described above, can be regarded as one of the function units **104**. In the present description, the wall surfaces of the building structure are not limited to surfaces of sidewalls standing between adjacent rooms. That is, the meaning of the wall surfaces includes outdoor and indoor wall surfaces, and the indoor wall surfaces comprise ceiling and floor surfaces as well as the sidewall surfaces. In the drawings, **MB** designates a main circuit breaker, **BB** designates a branch circuit breaker, and **GW** designates a gateway (e.g., a router or a built-in hub).

As shown in FIGS. 23 and 24, each of the base units **103** is formed with a gate housing **130** having terminals (**131a**, **132a**, **131b**, **132b**) connected to the power supply line **Lp** and the information line **Ls**, and a main housing **135** detachably connected to the function unit **104**. Theses housings can be made of a synthetic resin having electrical insulating characteristics (e.g., a non-crystalline plastic such as ABS resin). The terminals (**131b**, **132b**) are used to extend the wirings. The gate housing **130** and the main housing **135** have a pair of a module port **134** and a module connector **142**, which are detachably connected to each other to simultaneously establish both of supplying the electric power from the gate housing **130** to the main housing **135**, and making a signal transmission therebetween. In addition, when the function unit **104** has the module connector **142**, it can be connected to the gate housing **130** in place of the main housing **135**. If necessary, the gate housing **130** and main housing **135** may be integrally formed.

Circuit components of the base unit **103** are designed to transmit the electric power and the information signal to the function unit **104**. For example, the base unit **103** is provided with an AC/AC converter **160**, DC power section **161**, transceiver section **162**, E/O converter **163**, O/E converter **165**, and a function portion **167**.

The AC/AC converter **160** converts commercial AC voltage to a lower AC voltage having an increased frequency, and applies the lower AC voltage to a coil

172. The DC power section **161** generates an operating voltage of the internal circuit components from a stable DC voltage obtained by rectifying and smoothing the lower AC voltage. The transceiver section **162** transmits and receives the information signal for enabling the mutual communication through the information line **Ls**. The E/O converter **163** converts the information signal received from the information line **Ls** to an optical signal, and outputs the optical signal though a light emitting device (LED) **164**. On the other hand, the O/E converter **165** receives the optical signal provided from the outside, e.g., the function unit **104** by a light receiving device (PD) **166**, and converting the received optical signal into the information signal to transmit it to the transceiver section **162**. The base unit **103** shown in FIG. 23 has an outlet tap as the function portion **167**. Alternatively, a sensor device or a controller may be formed as the function portion **167**.

The module port **134** formed at a front surface of the gate housing **130** is composed of an electric power port **134a** for supplying the electric power and an information signal port **134b** for accessing the information line **Ls**, as shown in FIG. 25B. Arrangement and shapes of the electric power port **134a** and the information signal port **134b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25B, each of the electric power port **134a** and the information signal port **134b** is configured in a substantially rectangular shape such that they are arranged in parallel to each other.

On the other hand, the module connector **142** formed at a rear surface of the main housing **135** is composed of an electric power connector **142a** and an information signal connector **142b**, as shown in FIG. 25A. Arrangement and shapes of the electric power connector **142a** and the information signal connector **142b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25A, each of the electric power connector **142a** and the information signal connector **142b** is configured

in a substantially rectangular shape such that they are arranged in parallel to each other.

In this embodiment, the module port **134** has a guide portion **133** such as a ring-like wall or a ring-like groove extending around the electric power port **134a** and the information signal port **134b**. This guide portion **133** is formed to be engageable to an engaging portion **145** such as a ring-like wall of the main housing **135**, which is formed around the electric power connector **142a** and the information signal connector **142b**. By simply engaging the engaging portion **145** to the guide portion **133**, the electric power connector **142a** and the information signal connector **142b** can be simultaneously connected to the electric power port **134a** and the information signal port **134b**. Therefore, there advantages that the main housing **135** can be easily attached to or removed from the gate housing **130** with improved connection reliability. A module connector having the same structure described above may be formed in the function unit **104**. The module port **134** and the module connector **142** may be formed by female and male connectors.

In the present embodiment, the gate housing **130** is directly fixed to the switch box **102**. If necessary, the gate housing **130** may be fixed to the switch box **102** through an attachment member **175**, as shown in FIG. 26. For example, this attachment member **175** is formed by a rectangular frame **176** with a window hole **177**. In this case, the gate housing **130** is mounted on the attachment member **175** by fitting a front portion of the gate housing into the window hole **177**, and then engaging hooks (not shown) formed at both sides of the gate housing to the rectangular frame **176**. The attachment member **175** mounting the gate housing **130** thereon is fixed to the switch box **102** by use of fixing screws. Alternatively, the gate housing **130** may be directly fixed to the wall surface by use of exclusive clamps without using the switch box **102**.

The function unit **104** is designed to provide a desired function by using the electric power supplied to the function unit **104** through the base unit **103** or

using the mutual communication of the information signal between the function unit **104** and the information line **Ls** through the base unit **103**. For example, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a high position near the ceiling, it preferably has a receptacle function of receiving a plug with hook of a lighting apparatus, security function such as a motion sensor, temperature sensor, and monitoring camera, or a sound function such as speaker. In addition, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a middle height, at which the function unit **104** can be easily operated by the user, it preferably has a switch function of turning on/off the lighting apparatus, control function for electric appliances such as air-conditioning equipments, a display function such as liquid crystal display or a timer function. The intercom device of the present invention is preferably connected as one of the function units **104** to the base unit **103** at the middle height. Moreover, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a low position near the floor, it preferably has a receptacle function for receiving a plug of an electric appliance such as electric vacuum cleaner, the sound function such as speaker, or a footlight function.

Specifically, as shown in FIG. 27, when a function section **181** of the function unit **104** is formed by a switch, operation data obtained by operating the switch is transmitted to a processing section **188** such as CPU through an I/O interface **189**. Then the processed data is sent to, for example, an infrared remote controller (not shown) through a transceiver section **187**, so that an electric appliance to be controlled is turned on/off by receiving a remote control signal emitted from the infrared remote controller. In addition, when the function section **181** is formed by a sensor, data detected by the sensor is transmitted as the information signal to the information line **Ls**, and then informed to the user by a required communicator.

The coil **172** wound around a core **170** of the base unit **103** shown in FIG.

23 is used as a power transmission means for supplying electric power to the function unit **104** in a noncontact manner. That is, the coil **172** of the base unit **103** provides an electromagnetic coupling portion that works as a first side of a transformer. On the other hand, as shown in FIG. 27, the function unit **104** has an electromagnetic coupling portion comprised of a coil **182** wound around a core **180**, which works as a second side of the transformer. Therefore, by forming electromagnetic coupling between the base unit **103** and the function unit **104**, a low AC voltage can be induced in the coil **182** of the function unit **104** to supply of electric power from the base unit **103** to the function unit **104**. Similarly, an electromagnetic coupling portion **X2** is formed at the opposite side of the electromagnetic coupling portion **X1**, and used to make a connection between the function units. In this embodiment, since the low AC voltage having the higher frequency than the commercial AC voltage is obtained by the AC/AC converter **160**, there is an advantage that the electromagnetic coupling portions used as the transformer can be downsized.

In addition, the light emitting device (LED) **164** of the base unit **103** shown in FIG. 23 is used to transmit an optical signal as the information signal to the function unit **104** in a noncontact manner. On the other hand, as shown in FIG. 27, a light receiving device (PD) **186** is disposed in the function unit **104** such that the light emitting device **164** is in a face-to-face relation with the light receiving device **186** when the function unit **104** is connected to the base unit **103**. Similarly, to transmit the optical signal as the information signal from the function unit **104** to the base unit **103**, the function unit **104** has a light emitting device (LED) **184**, which is disposed in the face-to-face relation with the light receiving element (PD) **166** when the function unit **104** is connected to the base unit **103**. Thus, each of the base unit **103** and the function unit **104** has the pair of the E/O converter (**163**, **183**) and the O/E converter (**165**, **185**) as an optical coupling portion to enable the mutual communication of the information signal therebetween in a noncontact manner.

As shown in FIG. 23, the electromagnetic coupling portion **X** used to supply the electric power and the optical coupling portion **Y** used for the mutual communication of the information signal are disposed at a side surface of the base unit **103** to be spaced from each other by a constant distance. Arrangements and shapes of the electromagnetic coupling portion **X** and the optical coupling portion **Y** are standardized (or stylized) to make the base unit **103** shareable among the function units **104**. In addition, the pair of the electromagnetic coupling portion **X** and the optical coupling portion **Y** may be formed at each of both sides of the function unit **104**, as shown in FIG. 27. That is, the optical coupling portion **Y1** formed at one side (e.g., left side) of the function unit **104** is composed of the light receiving device **186** located at the upper side and the light emitting device **184** located at the lower side, and the optical coupling portion **Y2** formed at the opposite side (e.g., right side) of the function unit **104** is composed of a light emitting device **194** located at the upper side and a light receiving device **196** located at the lower side. In this case, since the base unit **103** is connected to the one side of the function unit **104**, and another function unit or the intercom device of the present invention can be connected to the other side of the function unit **104**, there is an advantage of further improving function expandability of the wiring system.

The intercom device **101** of this embodiment can be connected to the base unit **103**, as shown in FIG. 28. In addition, internal components of the intercom device **101** of this embodiment are shown in FIG. 29. As understood from the comparison with the function unit **104** of FIG. 27, the intercom device **101** of FIG. 29 is substantially equivalent to a function unit having the components of the intercom device shown in FIG. 2 as the function section **181**.

Since this intercom device **101** has the stylized connectors as well as the function unit **104** and base unit **103** described above, it can be used in a desired combination with the base unit **103** and the function unit **104**. For example, as shown by the arrow ① in FIG. 30, the intercom device **101** can be

connected to the function unit **104** having the timer function, which is connected to the base unit **103**. Alternatively, as shown by the arrow ② in FIG 30, the intercom device **101** may be connected at its one side to the base unit **103**. At this time, the function unit **104** having the timer function can be connected to the other side of the intercom device **101**. Thus, there is a remarkable advantage that a layout of the function units **104** having various functions and the intercom device **101** can be changed with a high degree of freedom according to the user's needs.

In addition, when the intercom device **101** is operated in conjunction with at least one functions unit **104**, the wiring system of this embodiment can provide a higher order function in addition to the communication function. For example, when the intercom device **101** is connected to one of the base units **103**, and the function unit **104** having a sensor function (e.g., motion sensor or fire sensor) is connected to another one of the base units **103**, which is disposed at a proper location, the intercom device **101** can output an alarm sound by receiving alarm signal from the function unit. Therefore, the intercom device **101** of this case has an alarm generating function for disaster or crime prevention as well as the communication function. In addition, since it is not needed to individually prepare a speaker for generating the alarm sound, ther is a further advantage of achieving an improvement in cost performance of the wiring system as a whole.

Next, means for mechanically coupling between the base unit **103** and the function unit **104** (or the intercom device **101**) is explained.

For example, the base unit **103** shown in FIG. 31 has a cosmetic cover **112**, which is detachably attached to a front surface of the gate housing **135**, and horizontal guide rails **114** formed at upper and lower portions of the gate housing. The numeral **115** designates a stopper wall formed in a substantially center position in the longitudinal direction of the guide rail **114**. On the other hand, as shown in FIGS. 32A to 32C, the function unit **104** (or the intercom

device **101**) has concave portions **128** for accommodating joining members **150** therein, which are formed at both of its upper and lower sides, horizontal guide rails **124** extending in the concave portions **128**, and cover members **126** pivotally supported at upper and lower ends of the function unit **104**. In addition, as shown in FIG. 32D, the joining member **150** has a groove **152**, in which the horizontal guide rail (**114**, **124**) can be slidably fitted.

In this case, a cosmetic cover **112** is firstly removed from the gate housing **135** of the base unit **103**. Then, the electromagnetic coupling portion **X** and the optical coupling portion **Y** of the base unit **103** are connected to the electromagnetic coupling portion **X1** and the optical coupling portion **Y1** of the function unit **104** (or the intercom device **101**). Next, the cover member **126** is opened, and the joining member **150** is slid along the guide rail **124**, as shown by an arrow in FIG. 33. The slide movement of the joining member **150** is performed until the joining member **150** contacts the stopper wall **115**. As a result, the joining member **150** is engaged to the guide rail **114** of the base unit **103** over about half length of the joining member **150**. On the other hand, the remaining portion of the joining member **150** is still engaged to the guide rail **124** of the function unit **104** (or the intercom device **101**). Thus, by engaging the joining member **150** to both of the base unit **103** and the function unit **104** the base unit **103** can be mechanically coupled with the function unit **104** (or the intercom device **101**). Finally, the cosmetic cover **112** is attached to the base unit **103**, and the cover member **126** is closed.

According to this coupling means, since the joining member **150** is held between the cosmetic cover **112** and the gate housing **135**, it is possible to prevent accidental falling of the joining member **150**, and obtain the stable mechanical connection therebetween without spoiling the beauty of them. In addition, since the joining member **150** is always accommodated in the concave portion **128** of the function unit **104** (or the intercom device **101**), there is no worry about loss of the joining member **150**. This coupling means is also usual

in the case of mechanically coupling between the function units **104**, or between the function unit **104** and the intercom device **101**.

As an information-signal transmitting method available in the wiring system, a baseband transmission or a broadband transmission can be used. In addition, the protocol is not limited to a specific one. For example, sound and visual information signals may be transmitted and received according to JT-H232 packet to make the mutual communication between the intercom devices. In a control system, it is also preferred to use a routing protocol for a broadcast or a unicast that controlling can be performed at a control ratio of 1:1 or 1:N according to operation data. Alternatively, the protocol used between the base units may be different from the protocol used in the function unit or the intercom device connected to the base unit. In this case, a protocol conversion is preferably performed by the base unit.

In addition, the wiring system of the above embodiment is essential to use both of the information line and the power line installed in the building structure. As a modification of this embodiment, a power-line carrier communication may be used in the wiring system.

That is, the wiring system according to this modification is essential to install only the power line in the building structure. Therefore, the base unit is connected to only the power line. On the other hand, the base unit, the function unit and/or the intercom device need to have a transceiver unit for transmitting and receiving information signals in the power-line carrier communication manner. For example, when the transceiver unit is formed in the base unit, the electric power carrying the information signals thereon is separated to the electric power transmission and the information signal transmission by the base unit. Therefore, the same function unit and the intercom device used in the wiring system of the above embodiment are also available in this power-line carrier type wiring system.

Alternatively, the transceiver unit may be formed in the intercom device **101**, as shown in FIG. 34. In this case, the intercom device is detachably

connected to the base unit and/or the function unit by a power-line carrier type connector **Z**. In addition, the intercom device also comprises a PLC modem **210** for receiving the information signals carried by the power line, and transmitting information signals through the power line, a processing section **220** connected to the PLC modem, an I/O interface **230** between the processing section, and a function section **240** for providing the intercom function. Similar circuit components may be formed in the function unit and/or the base unit. As a modulation method for the power-line carrier communication, for example, it is possible to use a broad spectrum diffusion model, multi carrier model or an orthogonal Frequency division multiplexing (OFDM) model.

INDUSTRIAL APPLICABILITY

Thus, according to the present invention, the first microphone disposed in the front air chamber efficiently receives the voice output of the speaker, and the output of the first microphone is effectively used in the signal processing unit to remove the signal component corresponding to the voice output of the speaker from the output of the second microphone. Therefore, it is possible to improve the howling prevention effect. In addition, the intercom device can be downsized because a distance between the first microphone and the speaker becomes short. Furthermore, the intercom device is suitable for the wiring system with excellent function expandability, as described in the above embodiments.

Therefore, the present invention is expected to further increase the demand of the intercom device, and especially as one of the components of the wiring system of the next generation.

CLAIMS:

1. An intercom device comprising:
a housing;
a speaker accommodated in said housing such that a front air chamber is defined as a space enclosed by a front surface of said speaker and an inner surface of said housing, and a rear air chamber is defined as a space enclosed by a rear surface of said speaker and an inner surface of said housing;
a first microphone disposed such that its sound receiving surface faces the front air chamber;
a second microphone disposed such that its sound receiving surface faces outside of said housing; and
a signal processing unit configured to, when an audio output of said speaker is picked up by said second microphone, remove a signal component corresponding to the audio output of said speaker from an output signal of said second microphone by use of an output signal of said first microphone.

2. The intercom device according to claim 1, wherein the rear air chamber is the space hermetically-sealed by the rear surface of said speaker and the inner surface of said housing.

3. The intercom device according to claim 1, wherein said first microphone is arranged such that the sound receiving surface of said first microphone faces a diaphragm of said speaker.

4. The intercom device according to claim 1, wherein said second microphone is located adjacent to said speaker.

5. The intercom device according to claim 1, wherein said first microphone and said second microphone are mounted on a same circuit board.

6. The intercom device according to claim 5, wherein the circuit board is mounted on an outside surface of said housing such that the sound receiving surface of said first microphone faces the front air chamber and the sound receiving surface of said second microphone faces the outside of said housing.

7. The intercom device according to claim 1, further comprising an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

8. The intercom device according to claim 7, wherein said acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of said speaker is needed.

9. The intercom device according to claim 1, further comprising a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

10. The intercom device according to claim 7, wherein said acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant

frequency of said speaker that is supposed to be mounted on an infinite baffle and a lowest resonant frequency of said speaker mounted on the rear air chamber with the absence of said acoustic tube.

11. The intercom device according to claim 7, wherein said acoustic tube is formed along an inner surface of the rear air chamber.

12. The intercom device according to claim 7, further comprising a sound absorbing material disposed at the opening of said acoustic tube.

13. A wiring system using the intercom device as set forth in claim 1, comprising at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of said building structure, and connected to said transmission line;

wherein the intercom device has a first connector detachably connectable to a second connector formed in each of said base units,

when said first connector is connected to said second connector of one of said base units, the intercom device functions such that the information signals provided through said transmission line are output as audio information from said speaker unit, and an output of said signal processing unit is sent out through said transmission line.

14. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units, said function unit having a third connector detachably

connectable to one of said second connector of said base unit, and an extended connector formed in the intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, or when said first connector of the intercom device is connected to said second connector of said base unit, and said third connector of said function unit is connected to said extended connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided through said transmission line, outputting information signals provided through said transmission line, and sending out information signals through said transmission line.

15. The wiring system as set forth in claim 14, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

16. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units,

said function unit has a third connector detachably connectable to said second connector of said base unit and an extended connector detachably connectable to said first connector of said intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, and said extended connector of said function unit is connected to said first connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided from said transmission line through said base unit, outputting information signals provided from said transmission line through said base unit, and sending out

information signals to said transmission line through said base unit, and the intercom device operates such that the information signals provided from said transmission line through said base unit and said function unit are output as audio information from said speaker unit, and the output of said signal processing unit is sent out to said transmission line through said base unit and said function unit.

17. The wiring system as set forth in claim 16, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

18. The wiring system as set forth in claim 13, wherein said first and second connectors provide an electric power transmission between said base unit and the intercom device by means of electromagnetic coupling.

19. The wiring system as set forth in claim 13, wherein said first and second connectors provide a signal transmission between said base unit and the intercom device by means of optical coupling.

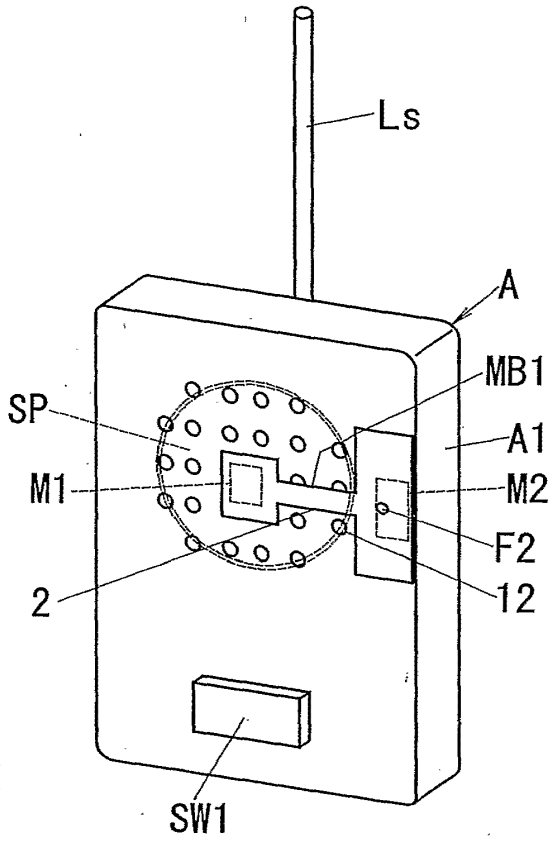


FIG. 1

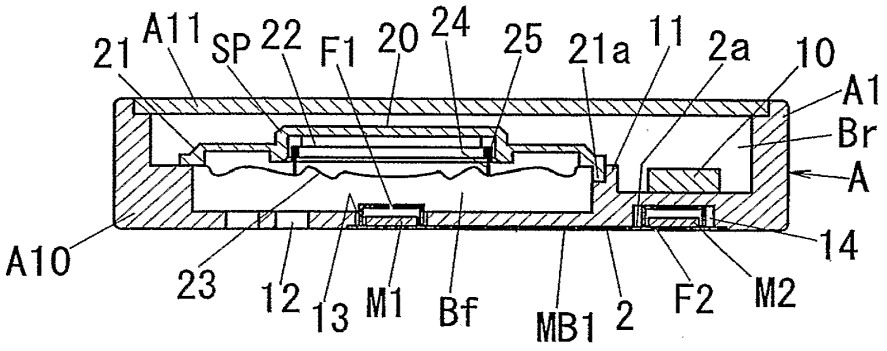

21
A11
SP
22
F1
20
24
25
21a
11
2a
10
A1
Br
A
14
A10
23
12
13
M1
Bf
MB1
2
F2
M2

FIG. 2

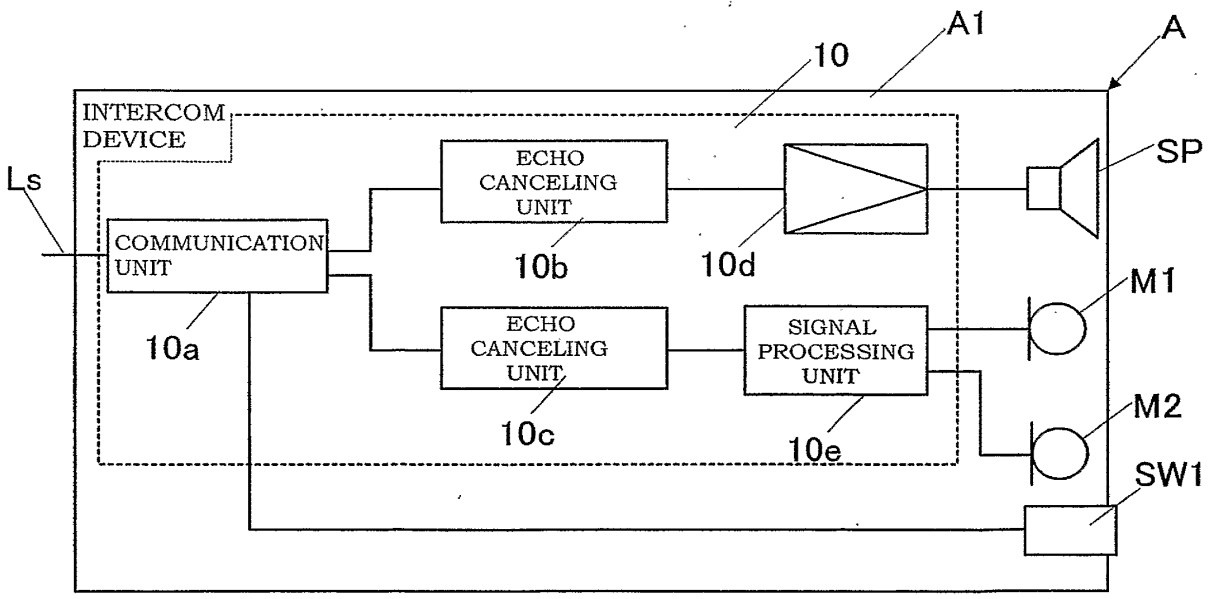

A1
A
10
INTERCOM DEVICE
Ls
COMMUNICATION UNIT
10a
ECHO CANCELING UNIT
10b
10d
SP
ECHO CANCELING UNIT
10c
SIGNAL PROCESSING UNIT
10e
M1
M2
SW1

FIG. 3

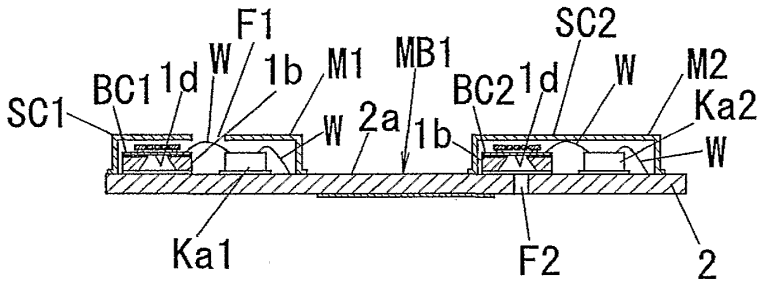

F1
W
1b
M1
MB1
SC2
SC1
BC1
1d
BC2
1d
W
M2
Ka2
W
2a
1b
W
Ka1
F2
2

FIG. 4

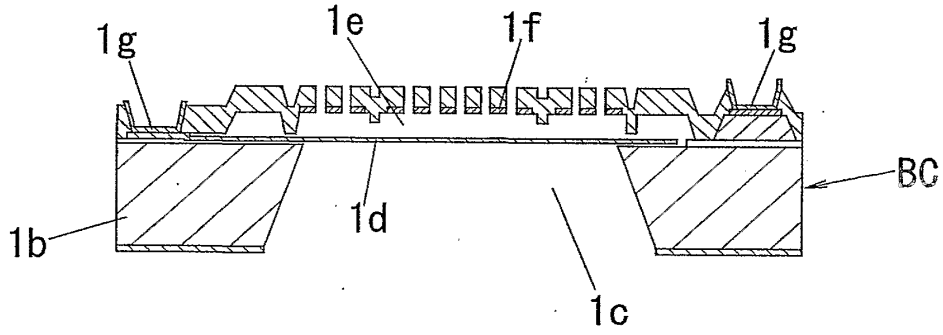

1g
1e
1f
1g
BC
1b
1d
1c

FIG. 5

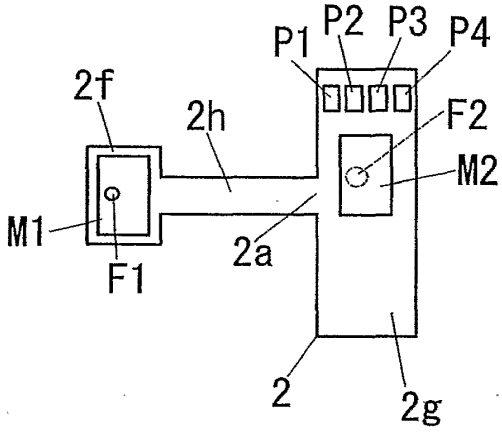


FIG. 6A

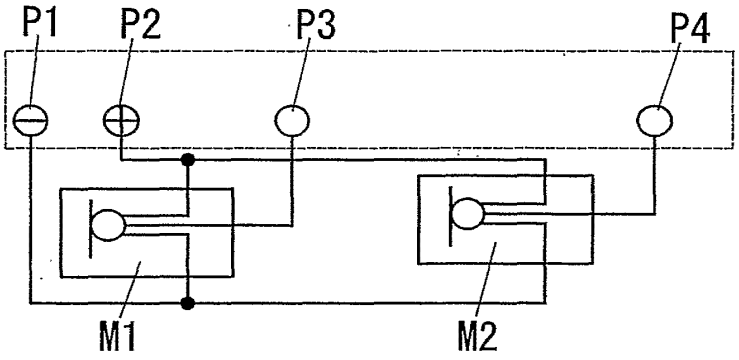


FIG. 6B

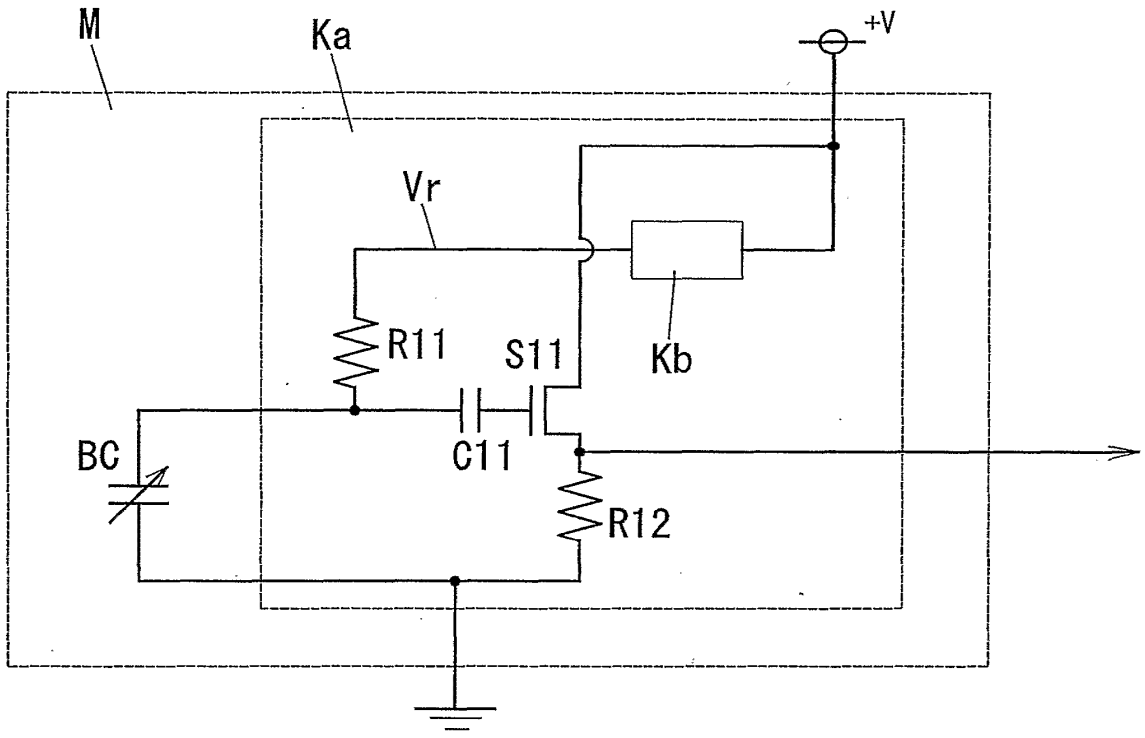


FIG. 7

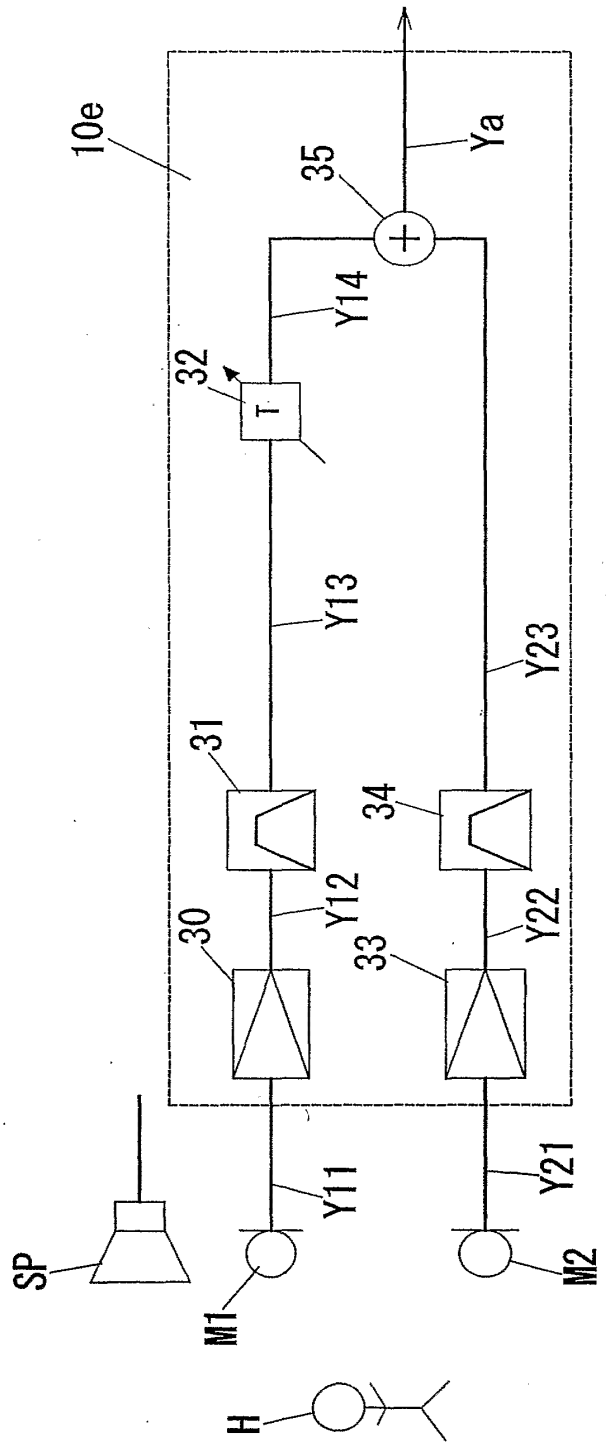


FIG. 8

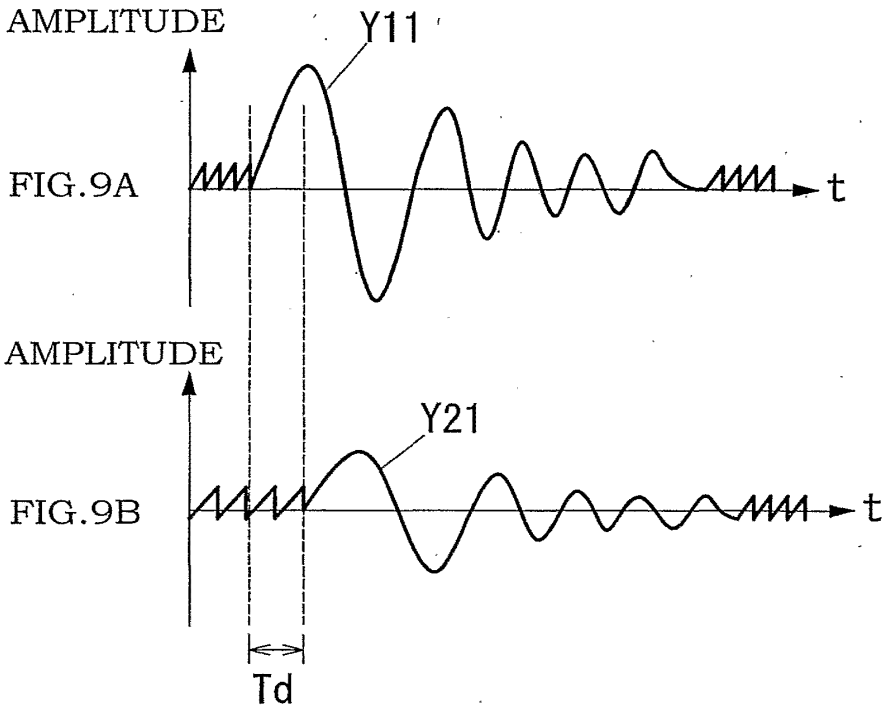


FIG.9A

FIG.9B

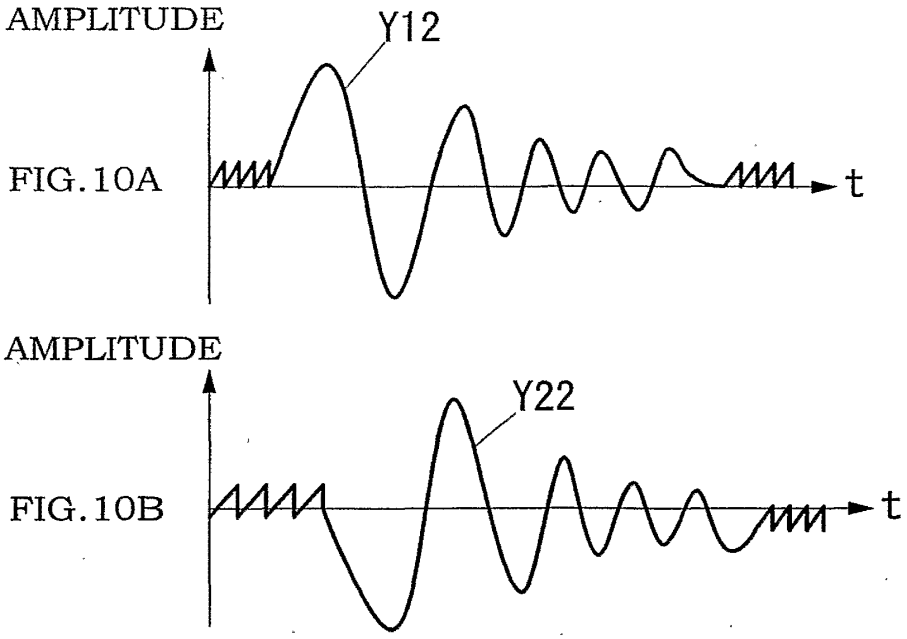


FIG.10A

FIG.10B

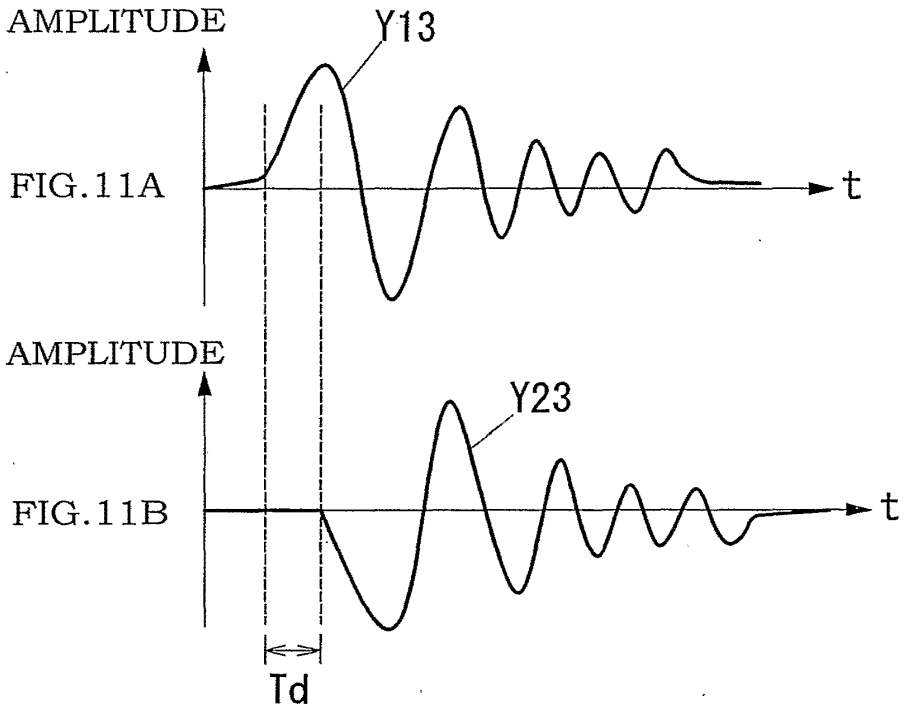


FIG.11A

FIG.11B

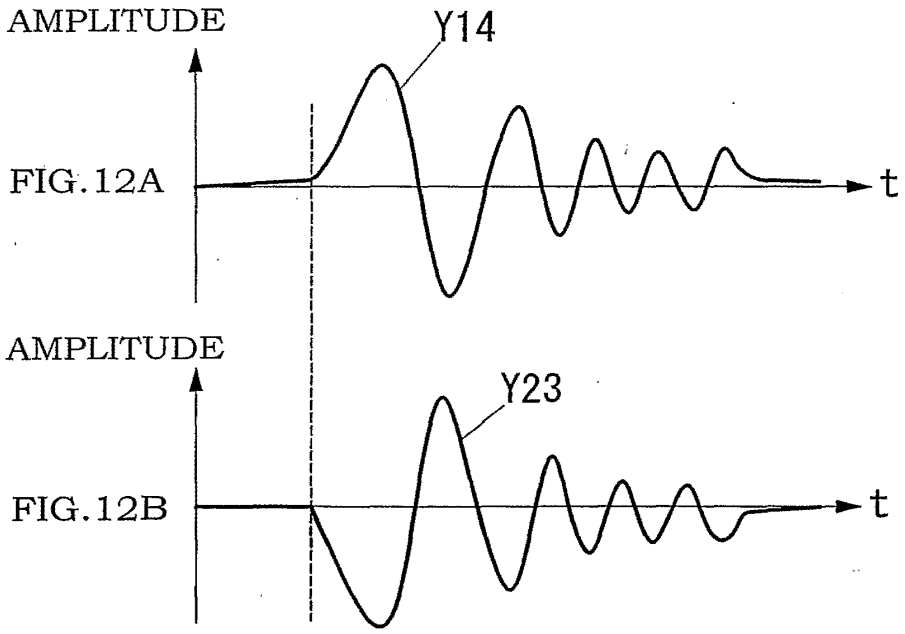


FIG.12A

FIG.12B

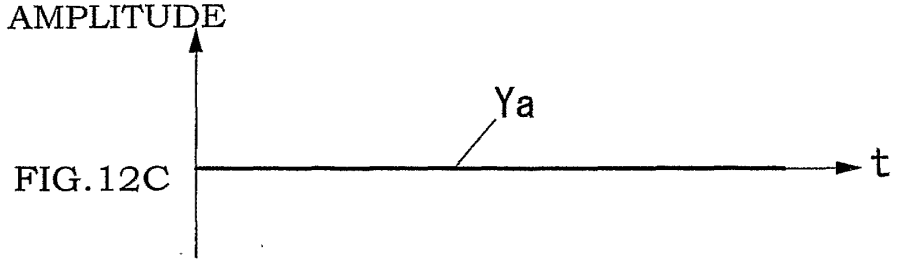
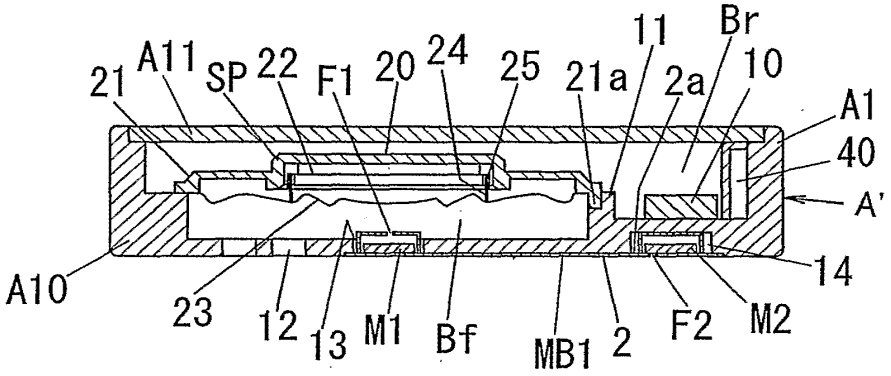


FIG.12C

FIG. 13



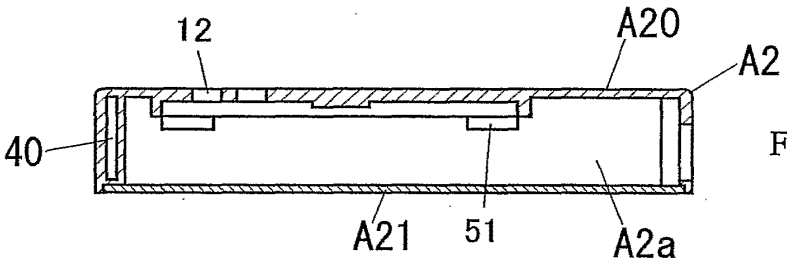
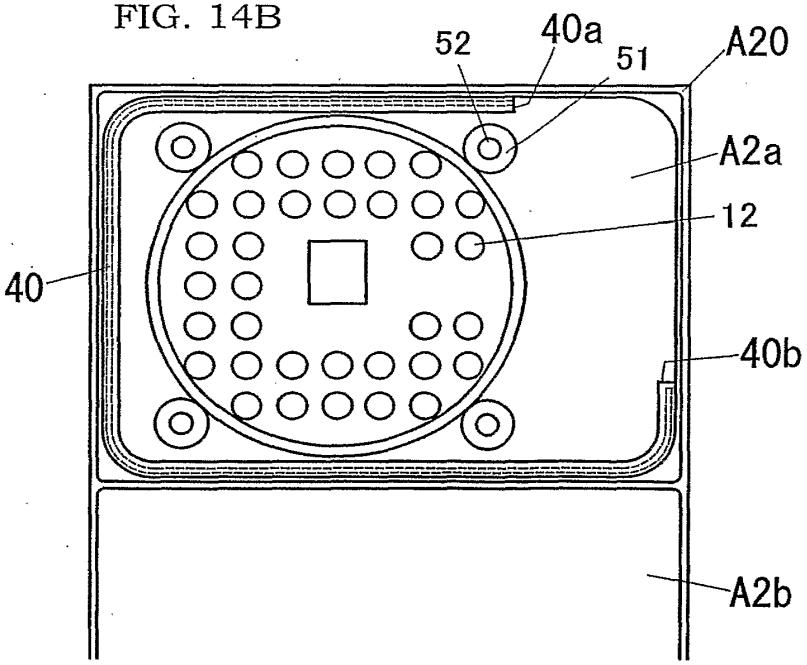


FIG. 14A

FIG. 14B



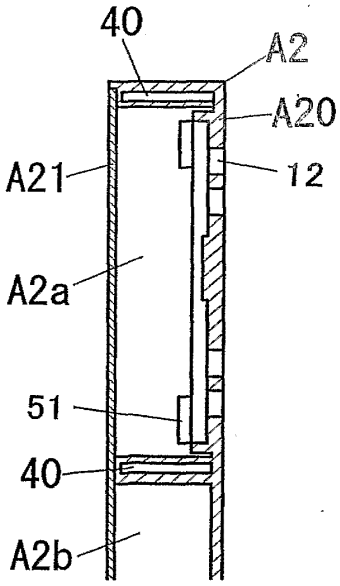


FIG. 14C

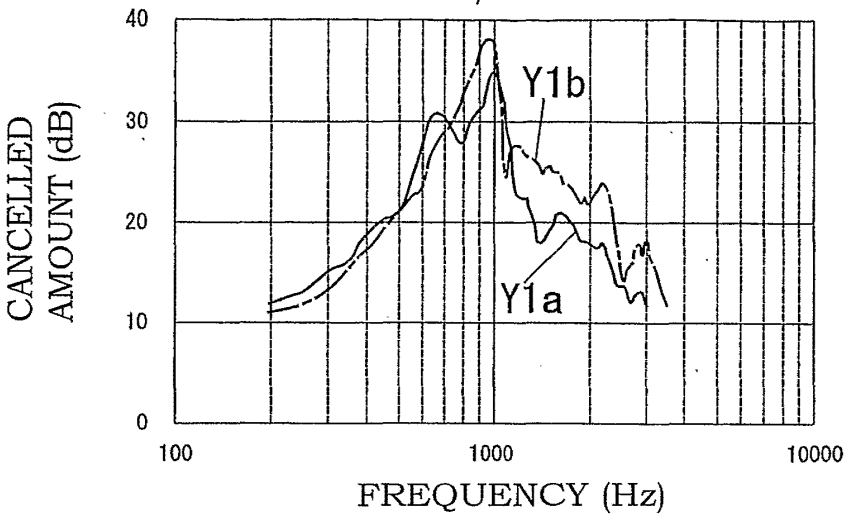


FIG. 15

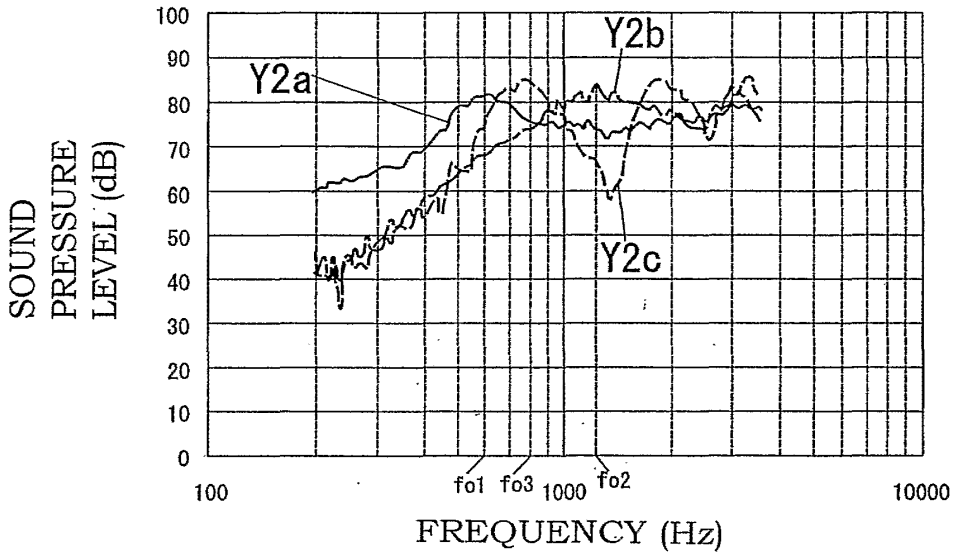


FIG. 16

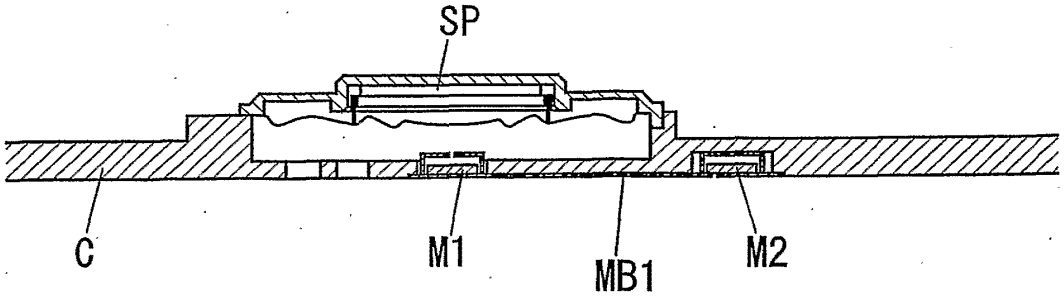


FIG. 17

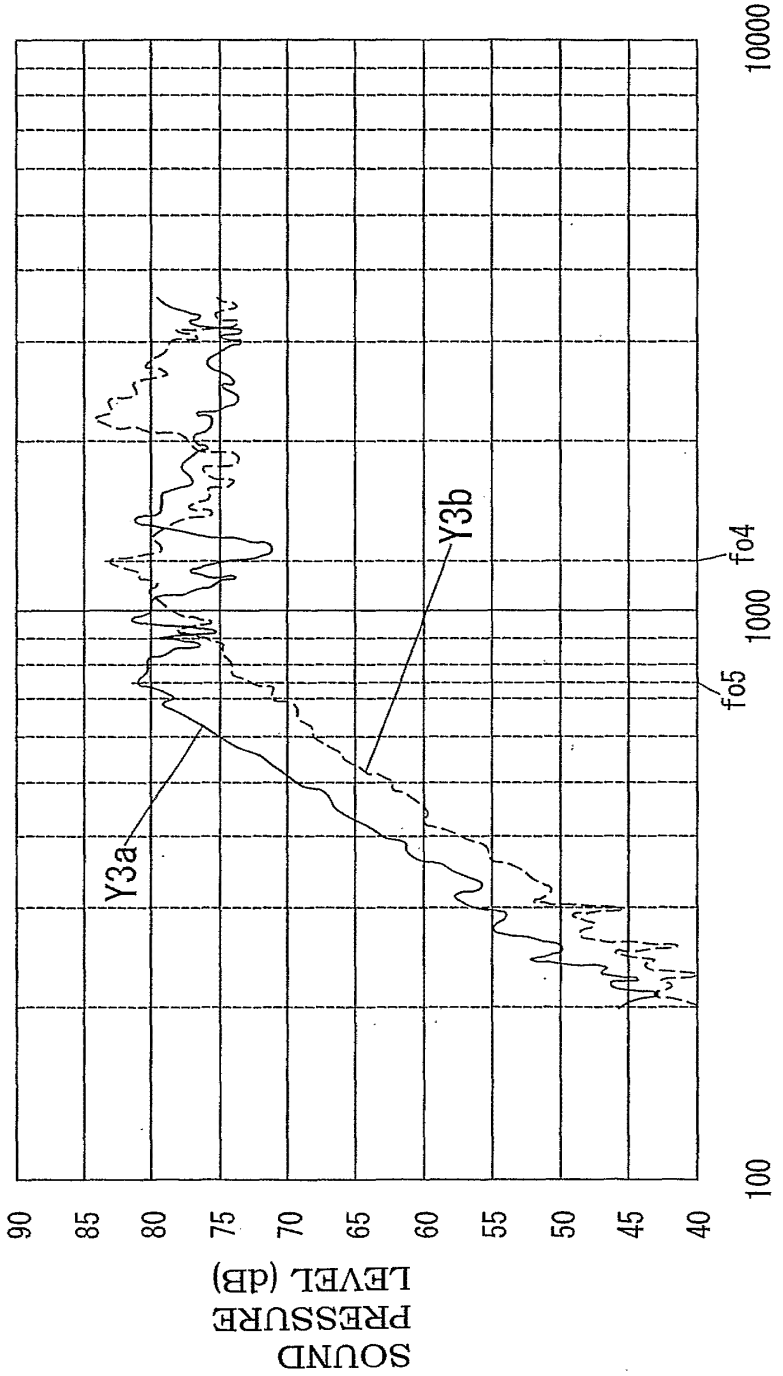

SOUND PRESSURE LEVEL (dB)
90
85
80
75
70
65
60
55
50
45
40
100
1000
10000
FREQUENCY (Hz)
fo4
fo5
Y3a
Y3b

FIG. 18

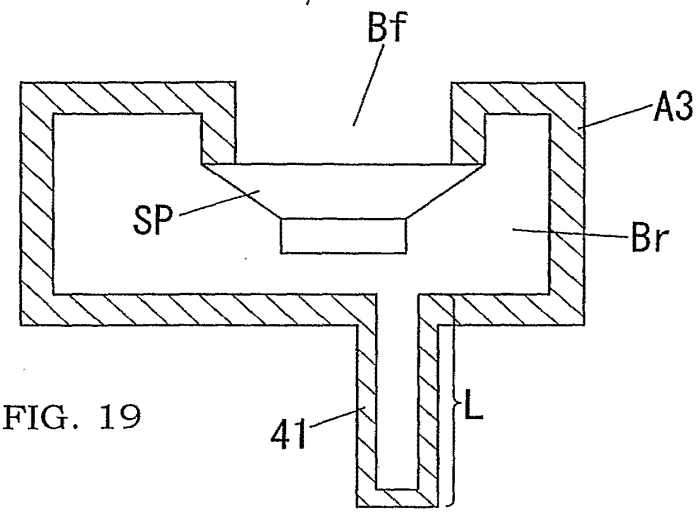


FIG. 19

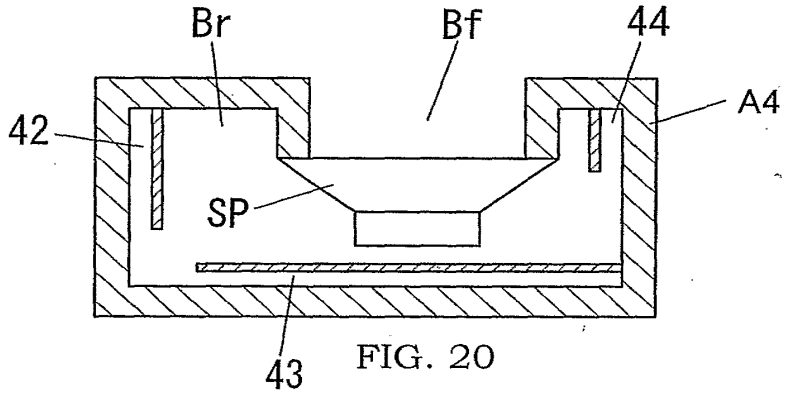


FIG. 20

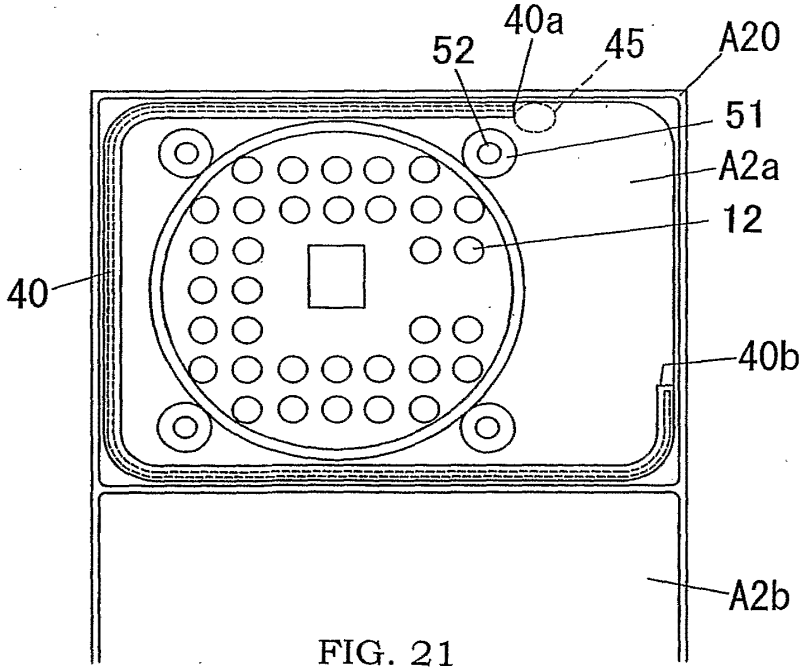


FIG. 21

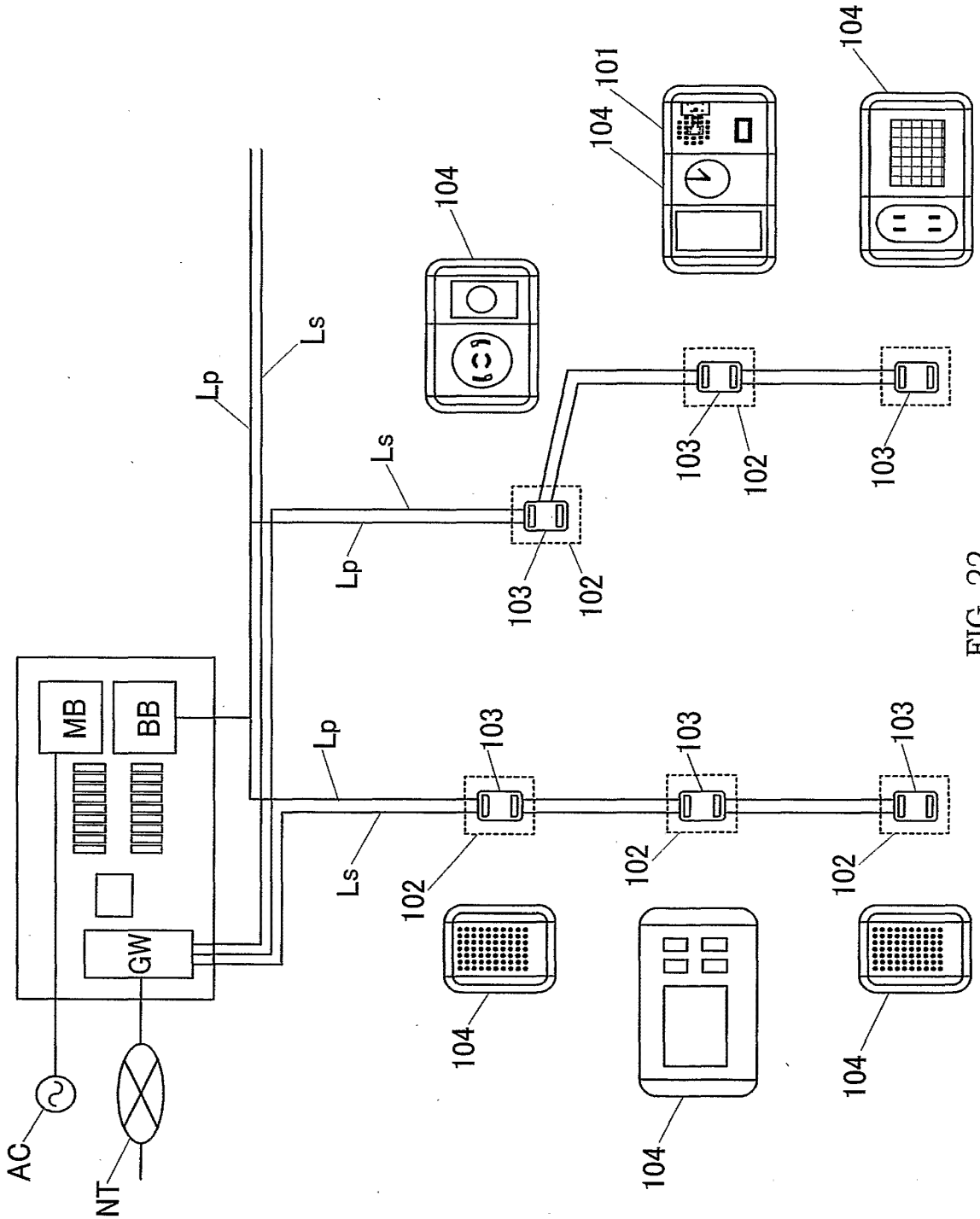


FIG. 22

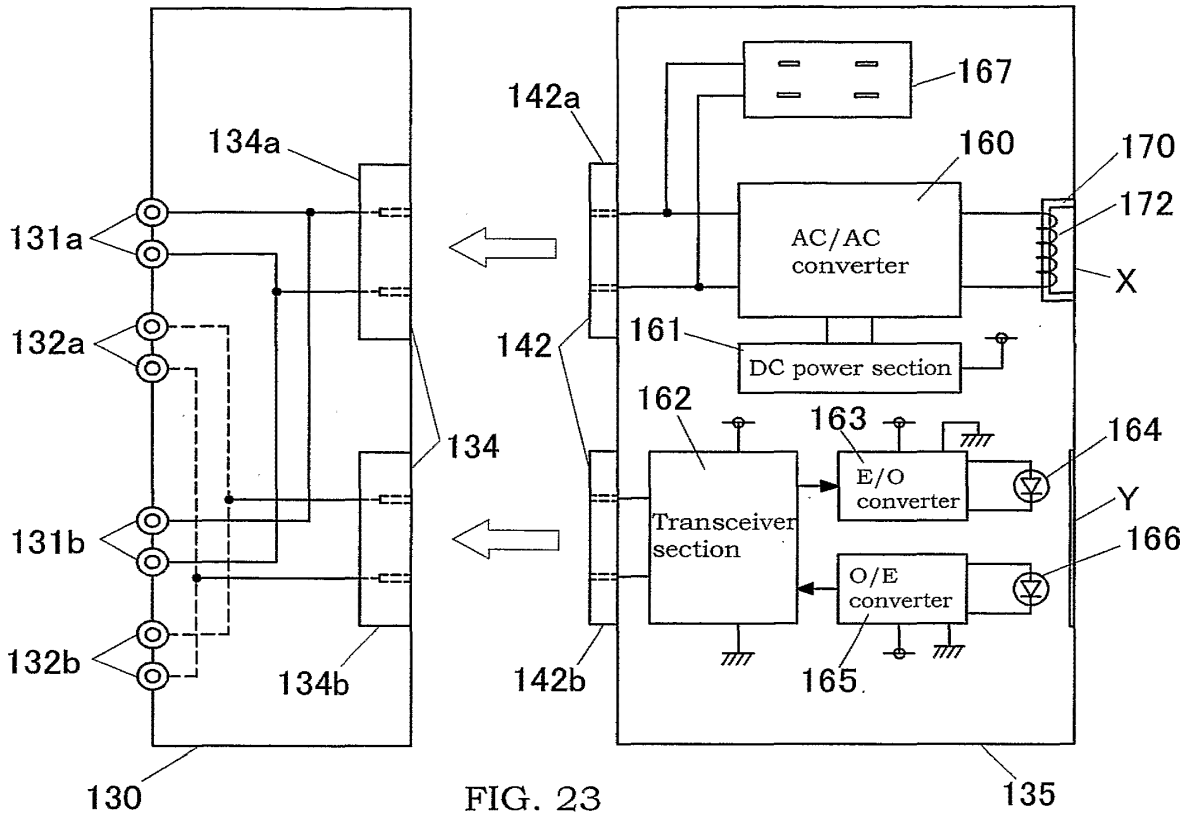

134a
131a
132a
131b
132b
134
134b
130
142a
167
160
170
172
X
AC/AC converter
142
161
DC power section
162
163
164
Y
166
E/O converter
Transceiver section
O/E converter
142b
165
135

FIG. 23

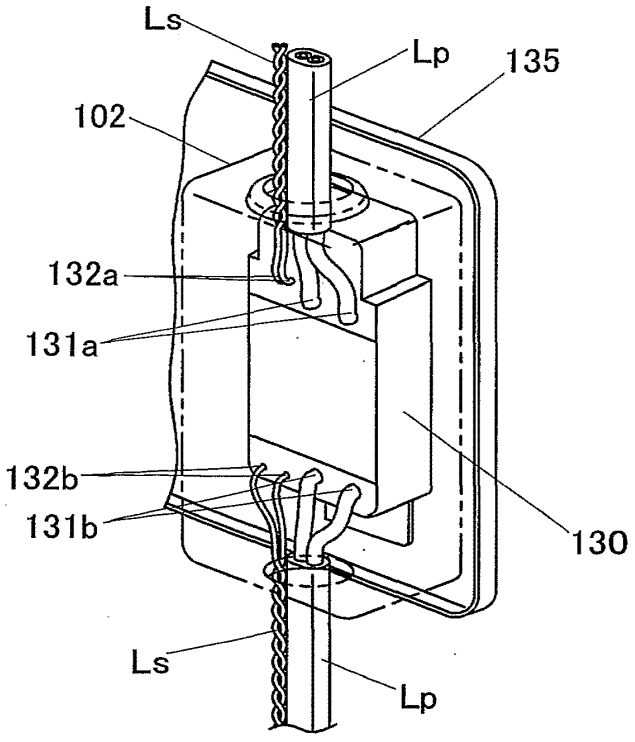

Ls
Lp
135
102
132a
131a
132b
131b
130
Ls
Lp

FIG. 24

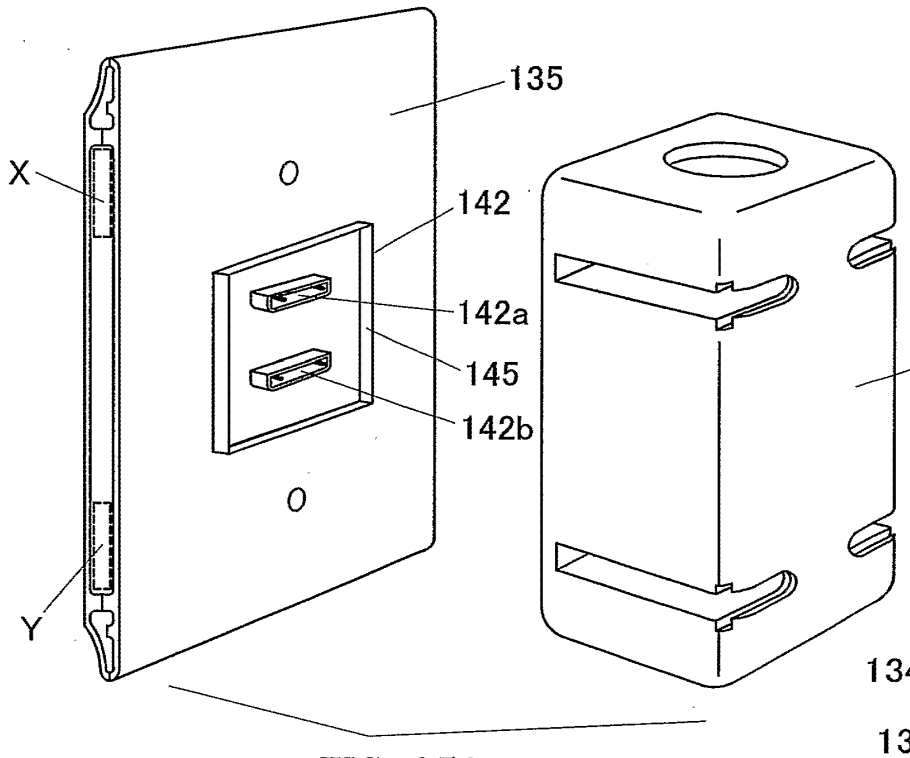


FIG. 25A

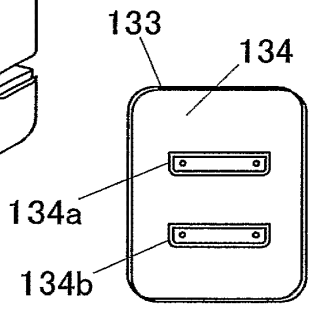


FIG. 25B

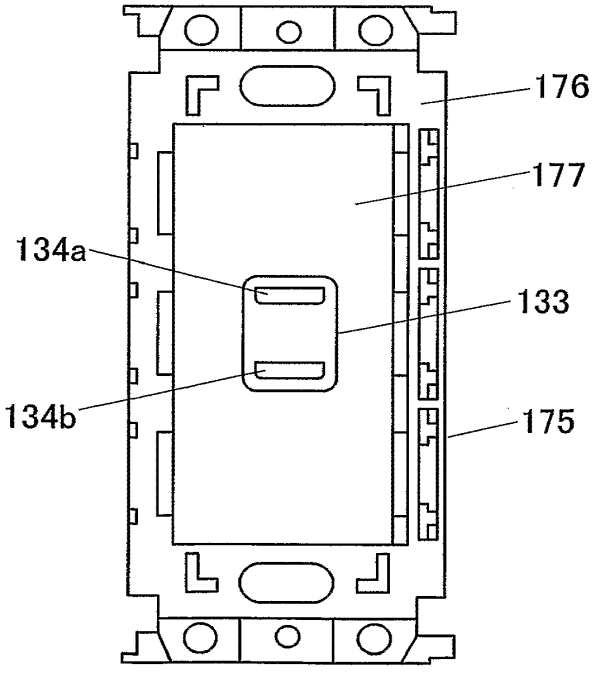


FIG. 26

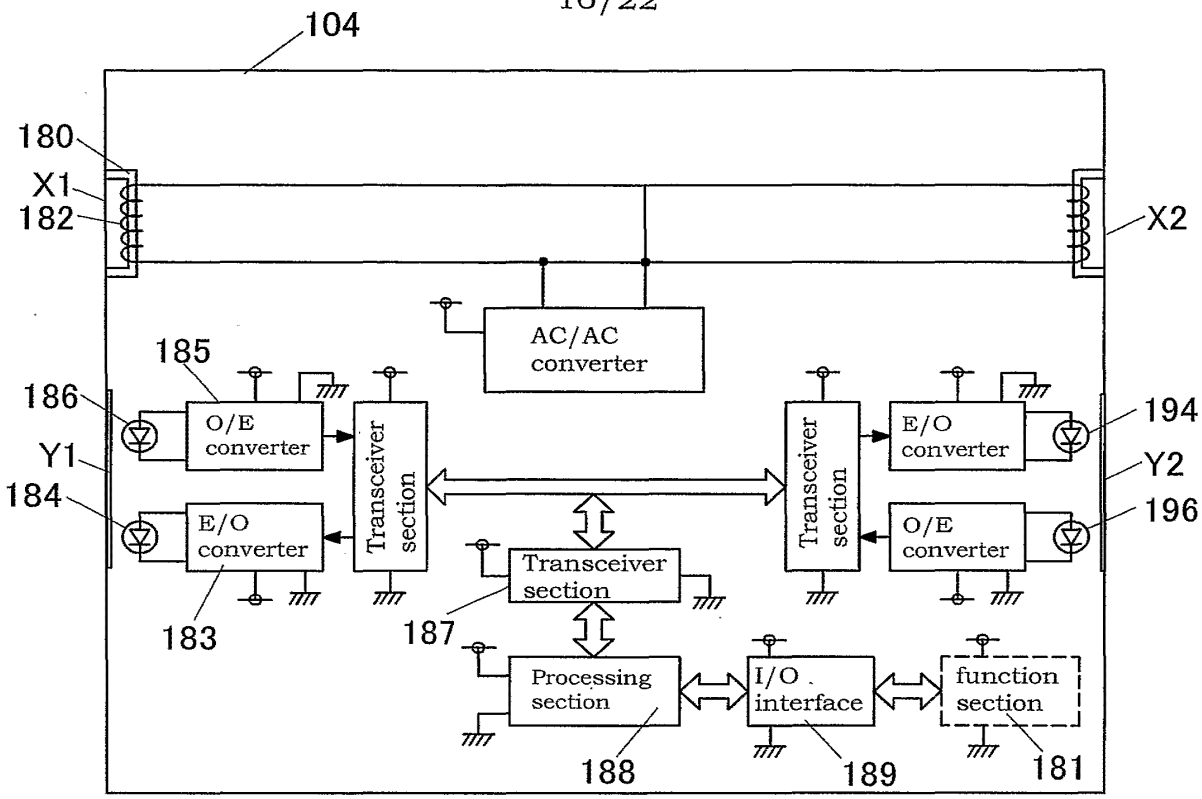

104
180
X1
182
X2
AC/AC converter
185
186
O/E converter
Transceiver section
Transceiver section
E/O converter
194
Y1
Y2
184
E/O converter
O/E converter
196
Transceiver section
183
187
Processing section
I/O interface
function section
188
189
181

FIG. 27

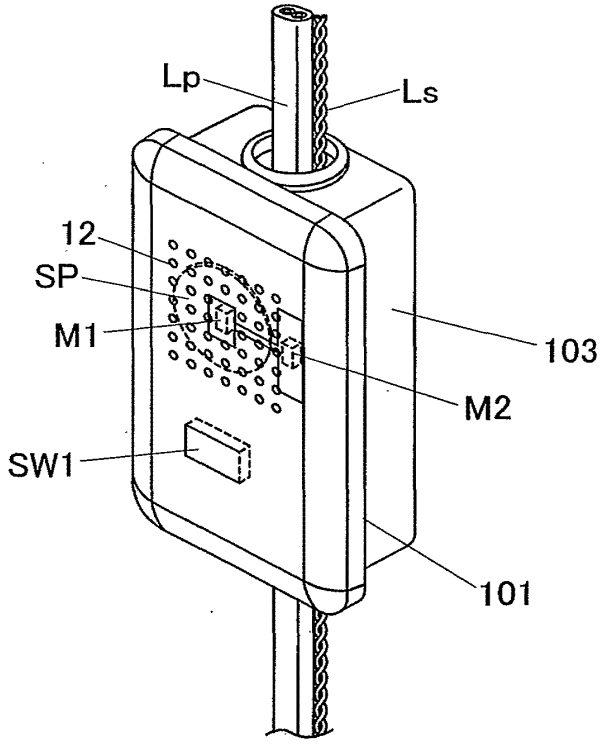

Lp
Ls
12
SP
M1
103
M2
SW1
101

FIG. 28

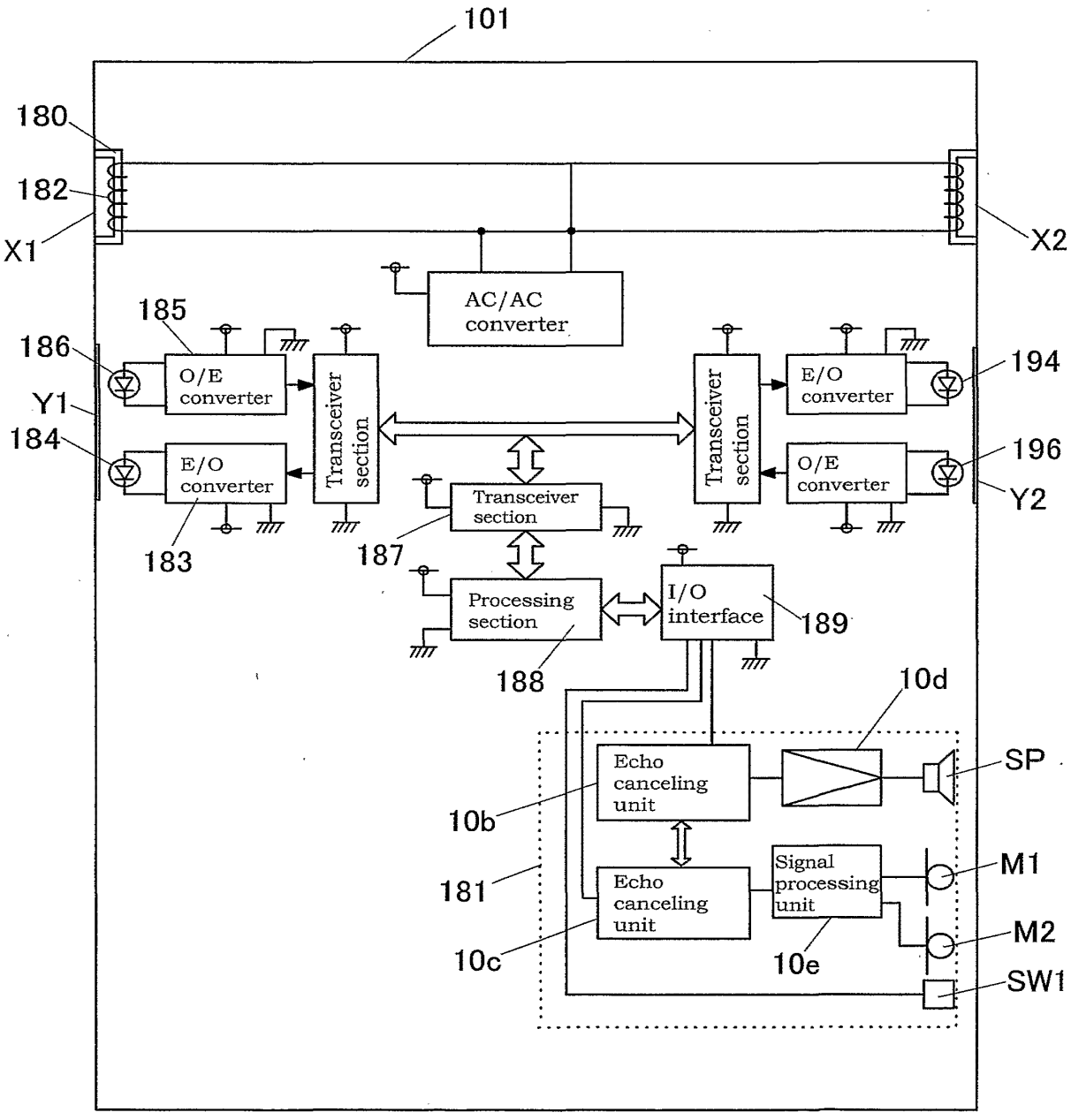

101
180
182
X1
X2
AC/AC converter
185
186
Y1
184
O/E converter
E/O converter
Transceiver section
183
187
Transceiver section
Processing section
I/O interface
189
188
194
196
Y2
10d
SP
10b
Echo canceling unit
181
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1

FIG. 29

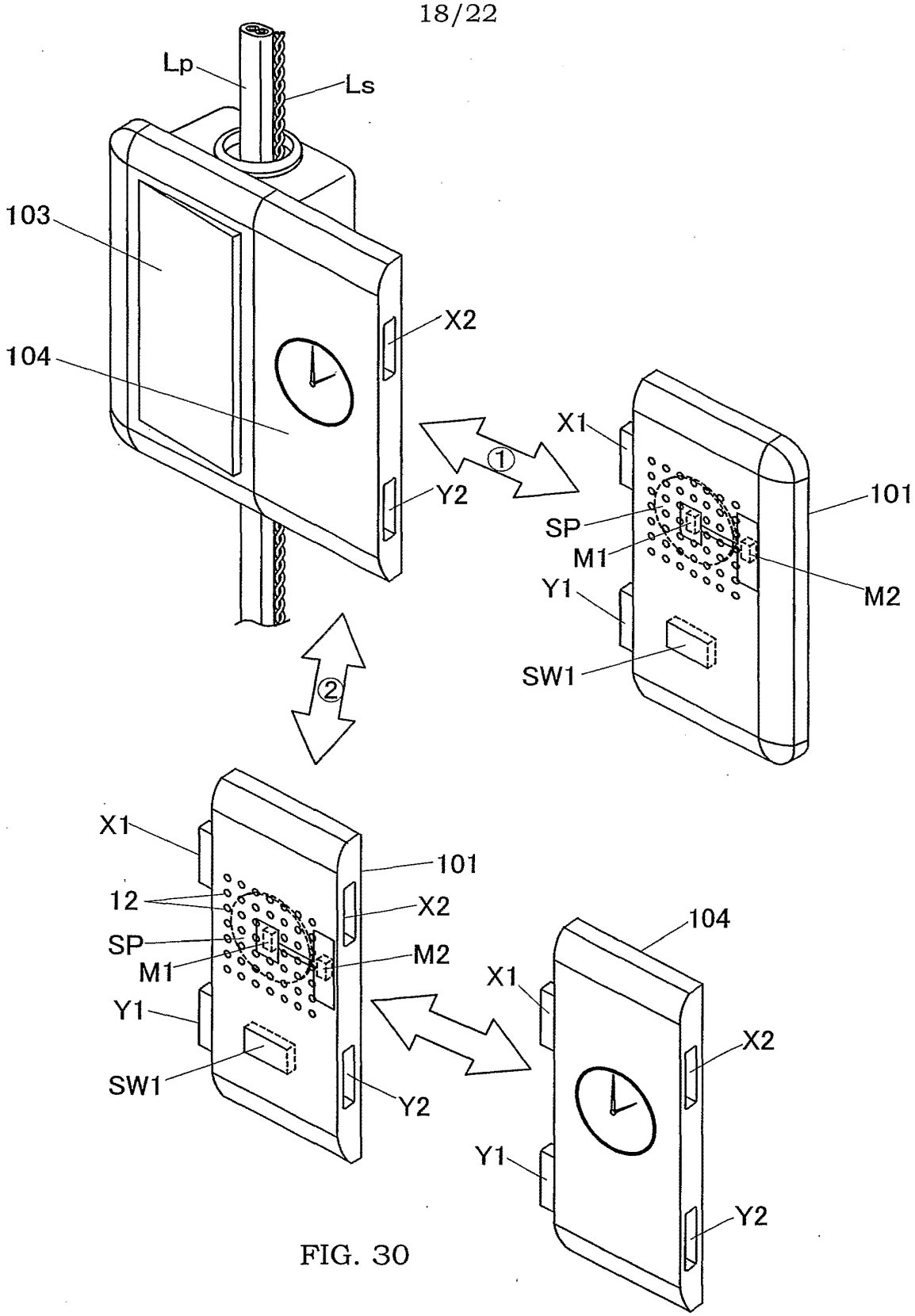

Lp
Ls
103
104
X2
Y2
①
X1
101
SP
M1
M2
Y1
SW1
②
X1
101
12
X2
SP
M1
M2
Y1
SW1
Y2
104
X1
X2
Y1
Y2

FIG. 30

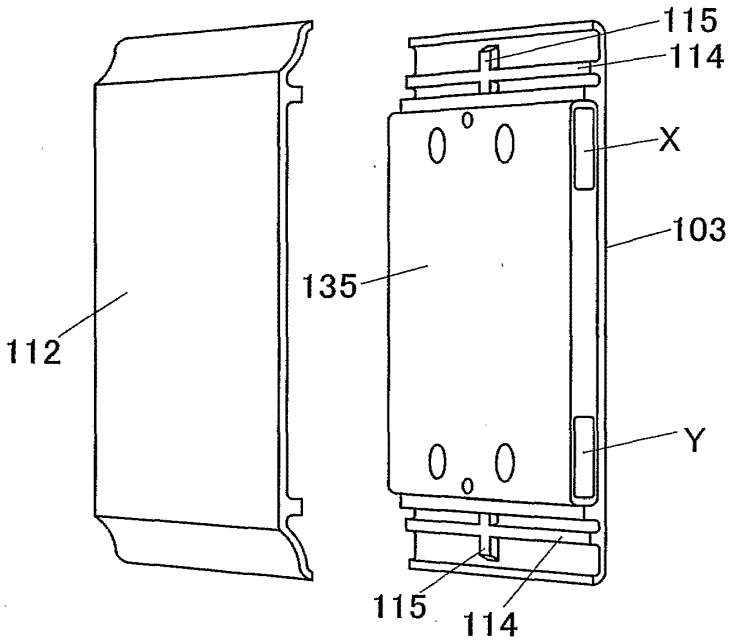


FIG. 31

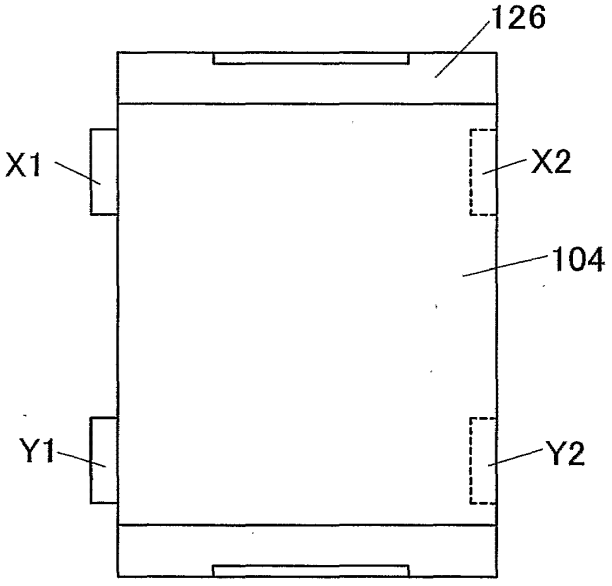


FIG. 32A

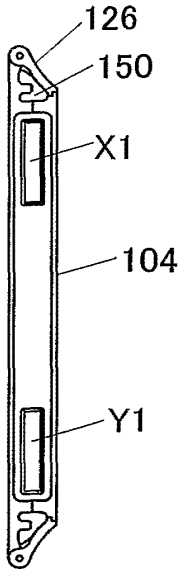


FIG. 32B

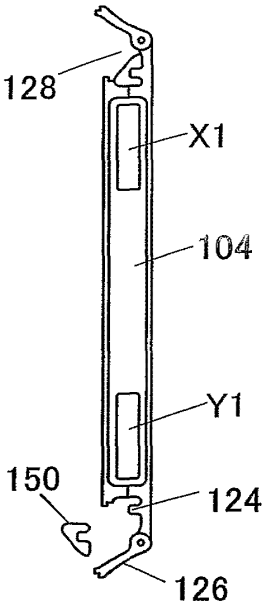


FIG. 32C

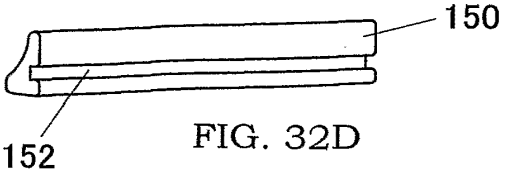


FIG. 32D

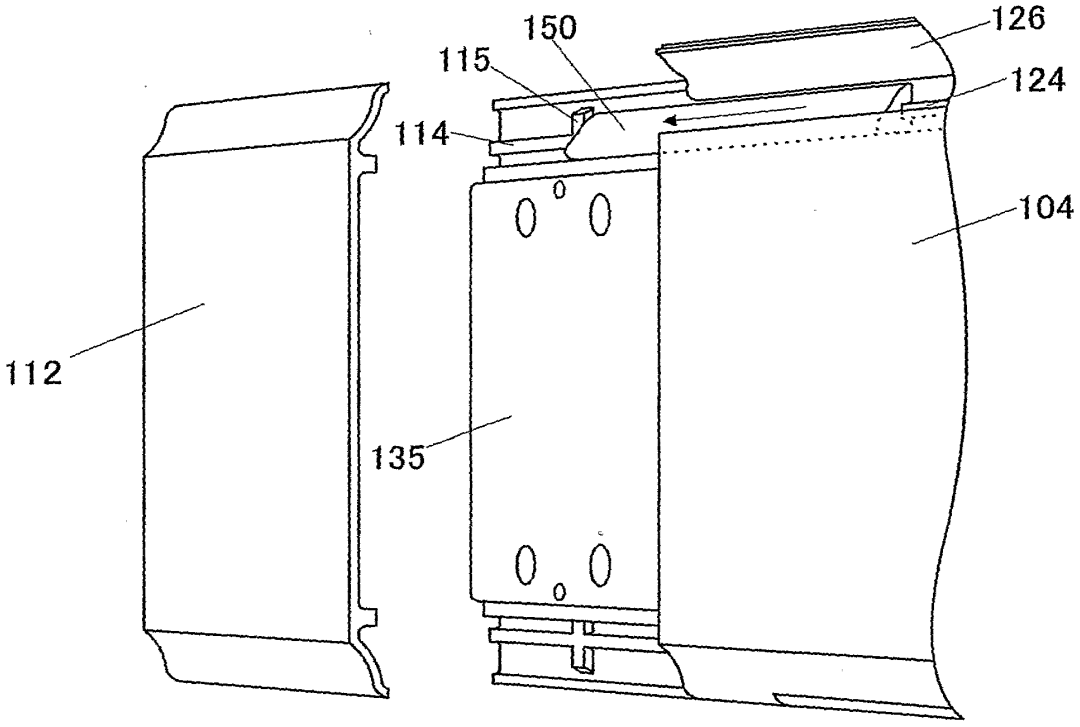


FIG. 33

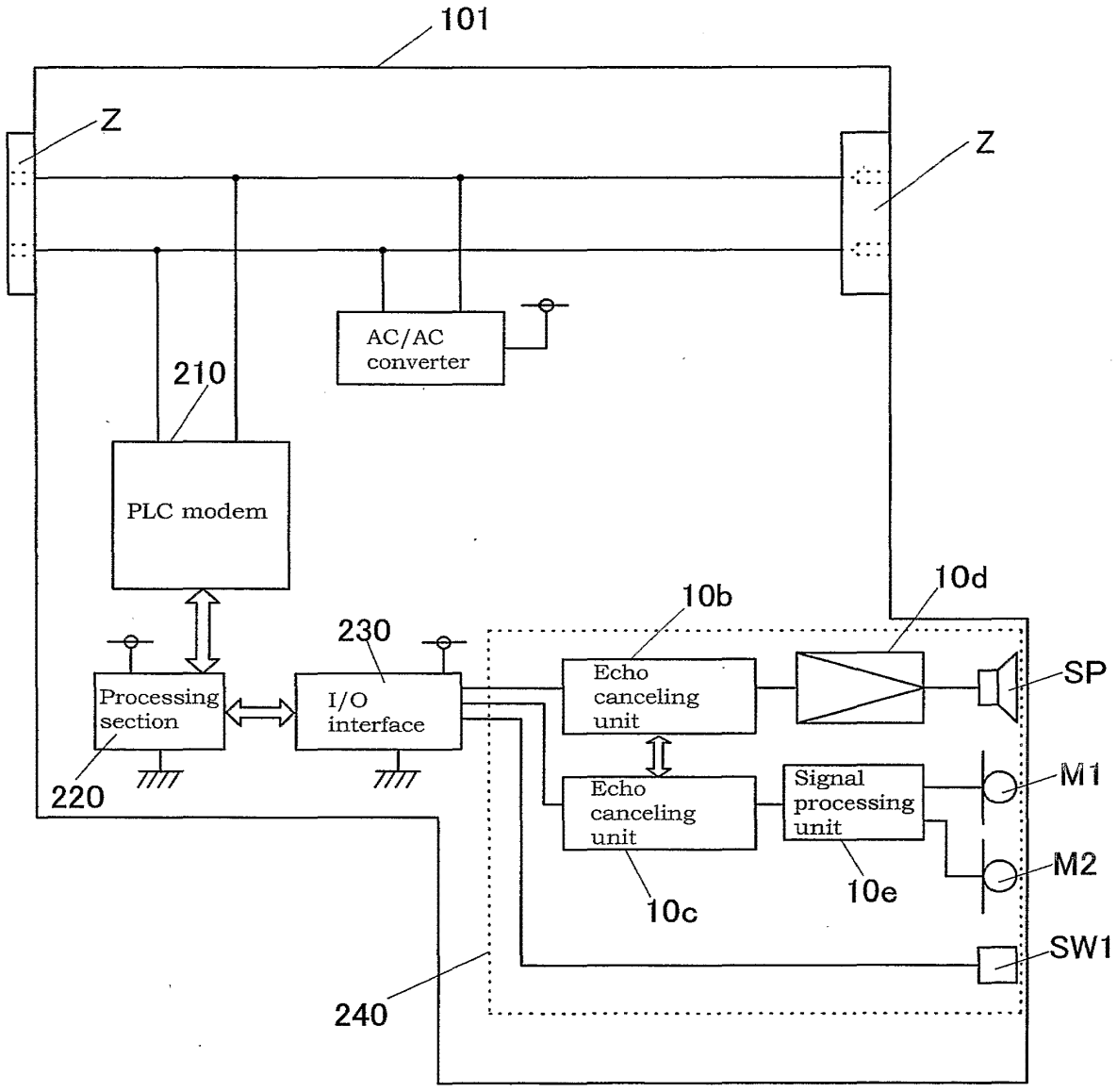

101
Z
Z
AC/AC converter
210
PLC modem
230
10b
10d
Processing section
I/O interface
Echo canceling unit
SP
220
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1
240

FIG. 34

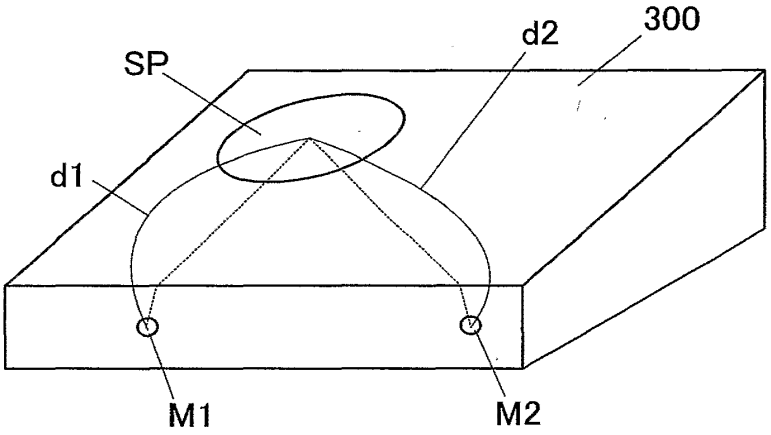


FIG. 35A

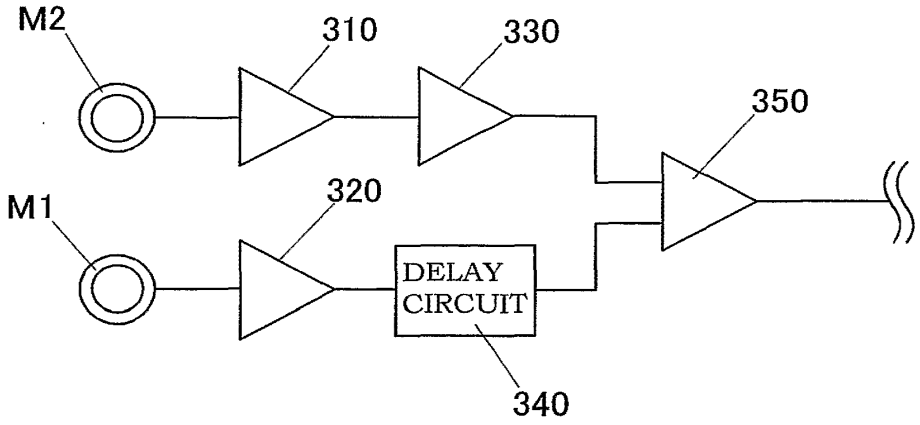


FIG. 35B

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP2007/069131

A. CLASSIFICATION OF SUBJECT MATTER

Int.Cl. H04R3/02(2006.01)i, H04M1/62(2006.01)i, H04R1/02(2006.01)i, H04R1/28(2006.01)i

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)

Int.Cl. H04R3/02, H04M1/62, H04R1/02, H04R1/28

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Published examined utility model applications of Japan 1922-1996
Published unexamined utility model applications of Japan 1971-2007
Registered utility model specifications of Japan 1996-2007
Published registered utility model applications of Japan 1994-2007

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.
A	JP 63-135100 A (BIBA KABUSHIKI KAISHA) 1988.06.07, Full Text; Fig.1 (No Family)	1-19
A	JP 2003-143677 A (MITSUBISHI ELECTRIC CORPORATION) 2003.05.16, Full Text; Fig.2 (No Family)	1-19
A	JP 01-253352 A (MATSUSHITA ELECTRIC WORKS,LTD.) 1989.10.09, Full Text; Fig.2 (No Family)	1-19

☐ 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
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"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
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"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
25.12.2007	08.01.2008

Name and mailing address of the ISA/JP
Japan Patent Office
3-4-3, Kasumigaseki, Chiyoda-ku, Tokyo 100-8915, Japan

Authorized officer
Hiroshi Ono
5Z 3654
Telephone No. +81-3-3581-1101 Ext. 3541

Form PCT/ISA/210 (second sheet) (April 2007)

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(19) World Intellectual Property Organization
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WO 2004/114722 A1

(51) International Patent Classification[7]: H04R 25/00

(21) International Application Number:
PCT/DK2004/000442

(22) International Filing Date: 23 June 2004 (23.06.2004)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
PA 2003 00944 24 June 2003 (24.06.2003) DK

(71) Applicant *(for all designated States except US)*: **GN RESOUND A/S** [DK/DK]; Maarkaervej 2A, DK-2630 Taastrup (DK).

(72) Inventor; and

(75) Inventor/Applicant *(for US only)*: **PEDERSEN, Brian, Dam** [DK/DK]; Kildebrinken 13, DK-4100 Ringsted (DK).

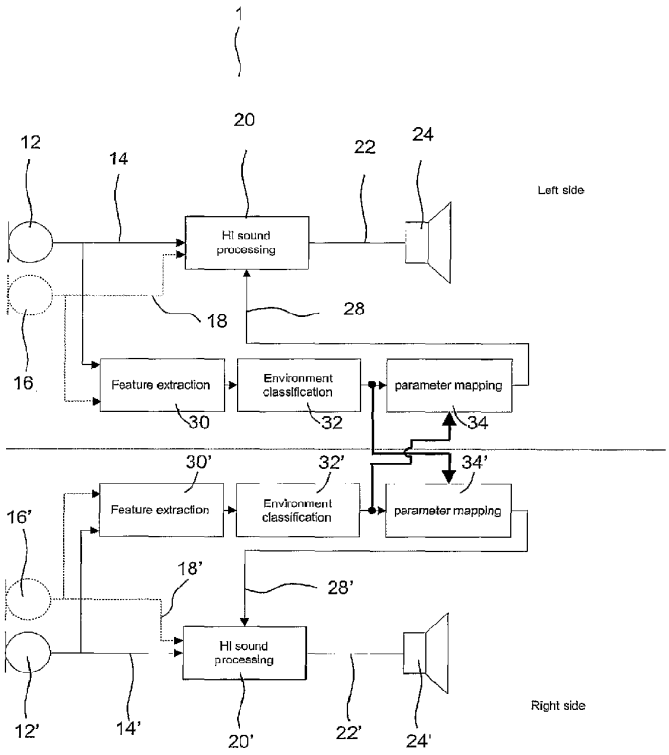
(74) Agent: **JAKOBSEN, Gert, Høy**; Albihns A/S, H.C. Andersens Boulevard 49, DK-1553 Copenhagen (DK).

(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NA, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK, SL, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, YU, ZA, ZM, ZW.

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PL, PT, RO, SE, SI,

[Continued on next page]

(54) Title: A BINAURAL HEARING AID SYSTEM WITH COORDINATED SOUND PROCESSING



(57) Abstract: The present invention relates to a binaural hearing aid system comprising a first hearing aid and a second hearing aid, each of which comprises a microphone and an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment, a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal, and a binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid for provision of outputs for each of the first and second hearing aids for selection of the signal processing algorithm of each of the respective hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

WO 2004/114722 A1

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— *before the expiration of the time limit for amending the claims and to be republished in the event of receipt of amendments*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

A BINAURAL HEARING AID SYSTEM WITH COORDINATED SOUND PROCESSING

FIELD OF THE INVENTION

The present invention relates to a binaural hearing aid system with a first hearing aid and a second hearing aid, each of which comprises a microphone, an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment, a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal.

BACKGROUND OF THE INVENTION

Today's conventional hearing aids typically comprise a Digital Signal Processor (DSP) for processing of sound received by the hearing aid for compensation of the user's hearing loss. As is well known in the art, the processing of the DSP is controlled by a signal processing algorithm having various parameters for adjustment of the actual signal processing performed. The gains in each of the frequency channels of a multi-channel hearing aid are examples of such parameters.

The flexibility of the DSP is often utilized to provide a plurality of different algorithms and/or a plurality of sets of parameters of a specific algorithm. For example, various algorithms may be provided for noise suppression, i.e. attenuation of undesired signals and amplification of desired signals. Desired signals are usually speech or music, and undesired signals can be background speech, restaurant clatter, music (when speech is the desired signal), traffic noise, etc.

The different algorithms or parameter sets are typically included to provide comfortable and intelligible reproduced sound quality in different sound environments, such as speech, babble speech, restaurant clatter, music, traffic noise, etc. Audio signals obtained from different sound environments may possess very different characteristics, e.g. average and maximum sound pressure levels (SPLs) and/or frequency content. Therefore, in a hearing aid with a DSP, each type of sound environment may be associated with a particular program wherein a particular setting of algorithm parameters of a signal processing algorithm provides processed sound of optimum signal quality in a specific sound environment. A set of such parameters may typically include parameters related to broadband gain, corner frequencies or slopes of frequency-selective filter algorithms and

parameters controlling e.g. knee-points and compression ratios of Automatic Gain Control (AGC) algorithms.

Consequently, today's DSP based hearing instruments are usually provided with a number of different programs, each program tailored to a particular sound environment category and/or particular user preferences. Signal processing characteristics of each of these programs is typically determined during an initial fitting session in a dispenser's office and programmed into the instrument by activating corresponding algorithms and algorithm parameters in a non-volatile memory area of the hearing aid and/or transmitting corresponding algorithms and algorithm parameters to the non-volatile memory area.

Some known hearing aids are capable of automatically classifying the user's sound environment into one of a number of relevant or typical everyday sound environment categories, such as speech, babble speech, restaurant clatter, music, traffic noise, etc.

Obtained classification results may be utilised in the hearing aid to automatically select signal processing characteristics of the hearing aid, e.g. to automatically switch to the most suitable algorithm for the environment in question. Such a hearing aid will be able to maintain optimum sound quality and/or speech intelligibility for the individual hearing aid user in various sound environments.

US 5,687,241 discloses a multi-channel DSP based hearing instrument that utilises continuous determination or calculation of one or several percentile values of input signal amplitude distributions to discriminate between speech and noise input signals. Gain values in each of a number of frequency channels are adjusted in response to detected levels of speech and noise.

However, it is often desirable to provide a more subtle characterization of a sound environment than only discriminating between speech and noise. As an example, it may be desirable to switch between an omni-directional and a directional microphone preset program in dependence of, not just the level of background noise, but also on further signal characteristics of this background noise. In situations where the user of the hearing aid communicates with another individual in the presence of the background noise, it would be beneficial to be able to identify and classify the type of background noise. Omni-directional operation could be selected in the event that the noise being traffic noise to allow the user to clearly hear approaching traffic independent of its direction of arrival. If, on the other hand, the background noise was classified as being babble-noise, the directional listening program could be selected to allow the user to hear a target speech signal with improved signal-to-noise ratio (SNR) during a conversation.

Applying Hidden Markov Models for analysis and classification of the microphone signal may obtain a detailed characterisation of e.g. a microphone signal. Hidden Markov Models are capable of modelling stochastic and non-stationary signals in terms of both short and long time temporal variations. Hidden Markov Models have been applied in speech recognition as a tool for modelling statistical properties of speech signals. The article "A Tutorial on Hidden Markov Models and Selected Applications in Speech Recognition", published in Proceedings of the IEEE, VOL 77, No. 2, February 1989 contains a comprehensive description of the application of Hidden Markov Models to problems in speech recognition.

WO 01/76321 discloses a hearing aid that provides automatic identification or classification of a sound environment by applying one or several predetermined Hidden Markov Models to process acoustic signals obtained from the listening environment. The hearing aid may utilise determined classification results to control parameter values of a signal processing algorithm or to control switching between different algorithms so as to optimally adapt the signal processing of the hearing aid to a given sound environment.

The different available signal processing algorithms may change the signal characteristics significantly. In binaural hearing aid systems, it is therefore important that the determination of sound environment does not differ for the two hearing aids. However, since sound characteristics may differ significantly at the two ears of a user, it will often occur that sound environment determination at the two ears of a user differs, and this leads to undesired different signal processing of sounds for each of the ears of the user.

SUMMARY OF THE INVENTION

Thus, there is a need for a binaural hearing aid system wherein sound environment determination does not differ for the two hearing aids so that signal processing in the two hearing aids may be coordinated and the user be provided with desired processed sound in both ears at the same time.

According to the present invention, this and other objects are solved by provision of a binaural hearing aid system of the above-mentioned type wherein the hearing aids are connected either by wire or by a wireless link to at least one binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid whereby the sound environment is determined and classified based on binaural signals. The one or more binaural sound environment detectors provide outputs for each of the first and second hearing aids for selection of the

signal processing algorithm of each of the hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

In this way both hearing aids may process sound in response to a common determination of sound environment. Sound environment determination may be performed by one common environment detector, for example situated in one of the hearing aids or in a remote control, or, by a plurality of environment detectors, such as an environment detector in each of the first and second hearing aids.

In the event that the user has substantially the same hearing loss in both ears and the sound environment is omni-directional, i.e. the sound environment does not change with direction, coordination of sound processing in the hearing aids leads to execution of identical signal processing algorithms in the respective signal processors of the hearing aids. In the event that the hearing aid user suffers from a binaural hearing loss, the signal processing algorithms may desirably differ for compensation of the different binaural hearing losses.

It is an important advantage of the present invention that binaural sound environment detection is more accurate than monaural detection since signals from both ears are taken into account.

It is a further advantage of the present invention that signal processing in the hearing aids of the binaural hearing aid system is coordinated since the sound environment detection is the same for both hearing aids.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the present invention reference will now be made, by way of example, to the accompanying drawings, in which:

Fig. 1 illustrates schematically a prior art monaural hearing aid with sound environment classification,

Fig. 2 illustrates schematically a first embodiment of the present invention,

Fig. 3 illustrates schematically a second embodiment of the present invention,

Fig. 4 illustrates schematically a third embodiment of the present invention, and

Fig. 5 illustrates schematically a fourth embodiment of the present invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Fig. 1 illustrates schematically a prior art monaural hearing aid 10 with sound environment classification.

The monaural hearing aid 10 comprises a first microphone 12 and a first A/D converter (not shown) for provision of a digital input signal 14 in response to sound signals received at the microphone 12 in a sound environment, and a second microphone 16 and a second A/D converter (not shown) for provision of a digital input signal 18 in response to sound signals received at the microphone 16, a processor 20 that is adapted to process the digital input signals 14, 18 in accordance with a predetermined signal processing algorithm to generate a processed output signal 22, and a D/A converter (not shown) and an output transducer 24 for conversion of the respective processed sound signal 22 to an acoustic output signal.

The hearing aid 10 further comprises a sound environment detector 26 for determination of the sound environment surrounding a user of the hearing aid 10. The determination is based on the output signals of the microphones 12, 16. Based on the determination, the sound environment detector 26 provides outputs 28 to the hearing aid processor 20 for selection of the signal processing algorithm appropriate in the determined sound environment. Thus, the hearing aid processor 20 is automatically switched to the most suitable algorithm for the determined environment whereby optimum sound quality and/or speech intelligibility is maintained in various sound environments.

The signal processing algorithms of the processor 20 may perform various forms of noise reduction and dynamic range compression as well as a range of other signal processing tasks.

The sound environment detector 26 comprises a feature extractor 30 for determination of characteristic parameters of the received sound signals. The feature extractor 30 maps the unprocessed sound inputs 14, 18 sound features, i.e. the characteristic parameters. These features can be signal power, spectral data and other well-known features.

The sound environment detector 26 further comprises an environment classifier 32 for categorizing the sound environment based on the determined characteristic parameters. The environment classifier categorizes the sounds into a number of environmental classes, such as speech, babble speech, restaurant clatter, music, traffic noise, etc. The classification process may consist of a simple nearest neighbour search, a neural network, a Hidden Markov Model system or another system capable of pattern recognition. The output of the environmental classification can be a "hard" classification containing one

single environmental class or a set of probabilities indicating the probabilities of the sound belonging to the respective classes. Other outputs may also be applicable.

The sound environment detector 26 further comprises a parameter map 34 for the provision of outputs 28 for selection of the signal processing algorithms.

The parameter map 34 maps the output of the environment classification 32 to a set of parameters for the hearing aid sound processor 20. Examples of such parameters are amount of noise reduction, amount of gain and amount of HF gain. Other parameters may be included.

Figs. 2-5 illustrate various preferred embodiments of the present invention. The illustrated binaural hearing aid system 1 comprises a first hearing aid 10 and a second hearing aid 10', each of which comprises a first microphone 12, 12 ' and an A/D converter (not shown) and a second microphone 16, 16' and A/D converter (not shown) for provision of a digital input signals 14, 14', 18, 18' in response to sound signals received at the respective microphones 12, 12', 16, 16' in a sound environment, a processor 20, 20' that is adapted to process the digital input signals 14, 18, 14', 18' in accordance with a predetermined signal processing algorithm to generate a processed output signal 22, 22', and a D/A converter (not shown) and an output transducer 24, 24' for conversion of the respective processed sound signals 22, 22' to an acoustic output signal

In the embodiments illustrated in Figs. 2-4, each of the hearing aids 10, 10' of the binaural hearing aid system 1 further comprises a binaural sound environment detector 26, 26' for determination of the sound environment surrounding a user of the binaural hearing aid system 1. The determination is based on the output signals of the microphones 12, 12', 16, 16'. Based on the determination, the binaural sound environment detector 26, 26' provides outputs 28, 28' to the hearing aid processors 20, 20' for selection of the signal processing algorithm appropriate in the determined sound environment. Thus, the binaural sound environment detectors 26, 26' determines the sound environment based on signals from both hearing aids, i.e. binaurally, whereby hearing aid processors 20, 20' is automatically switched in co-ordination to the most suitable algorithm for the determined environment whereby optimum sound quality and/or speech intelligibility is maintained in various sound environments by the binaural hearing aid system 1.

The binaural sound environment detectors 26, 26' illustrated in Figs. 2-4 are both similar to the binaural sound environment detector shown in Fig. 1 apart from the fact that the monaural environment detector only receives inputs from one hearing aid while each of the binaural sound environment detectors 26, 26' receives inputs from both hearing aids. Thus,

according to the present invention, signals are transmitted between the hearing aids 10, 10' so that the algorithms executed by the signal processors 20, 20' are selected in co-ordination, e.g. in case of an omni-directional sound environment, i.e. the sound environment does not change with direction, the algorithms are selected to be identical apart from possible differences in hearing loss compensation of the two ears.

In the embodiment of Fig. 2, the unprocessed signals 14, 14', 18, 18' from the microphones 12, 12', 16, 16' of one hearing aid 10, 10' are transmitted to the other hearing aid and inputted to the respective feature extractor 30, 30'. Thus, feature extraction in each of the hearing aids is based on the identical four input signals so that identical sound environment characteristic parameters will be determined binaurally in both hearing aids 10, 10'.

The signals may be transmitted in analogue form or in digital form, and the communication channel may be wired or wireless.

In the embodiment shown in Fig. 3, the output 36, 36' of the feature extractor 30, 30' of one hearing aid 10, 10' is transmitted to the respective other hearing aid 10', 10. The environment classifier 32, 32' then operates on two sets of features 36, 36' to determine the environment. Since both environment classifiers 32, 32' receive the same data, they will produce the same output.

In the embodiment shown in Fig. 4, the output 38, 38' of the environment classifier 32, 32' of one hearing aid 10, 10' is transmitted to the respective other hearing aid 10, 10'. The parameter map 34, 34' then operates on two inputs 38, 38' to produce the parameters for the processor algorithms, but since both parameter mapping units 34, 34' receive identical inputs, identical parameter values will be produced.

This embodiment has a number of advantages: Usually classification systems take both past and present data into account – they have memory. This makes them sensitive to missing data, since a classification requires a complete data set. Therefore it is required that the data link is safe, in the sense that data is guaranteed to be transmitted. The parameter mapping can be implemented without memory so that only present data is taken into account when generating parameters. This makes the system robust to packet loss and latency, since the parameter mapping may simply re-use old data in the event that data is missing. This will of course delay the correct action, but to the user the systems will appear to be synchronized.

The transmission data rate is low, since only a set of probabilities or logic values for the environment classes has to be transmitted.

Rather high latency can be accepted. By applying time constants to the variables that will change according to the output of the parameter mapping it is possible to smooth out any differences that is caused by latency. As described earlier it is important that signal processing in the two hearing instruments is coordinated. However if transition periods of a few seconds are allowed the system can operate with only 3-4 transmissions per second. Hereby, power consumption is kept low.

A binaural hearing aid system 1 with a remote control 40 is shown in Fig. 5. The environment detector 26 is positioned in the remote control 40. The required signals are transmitted to and from both hearing aids.

CLAIMS

1. A binaural hearing aid system comprising

a first hearing aid and a second hearing aid, each of which comprises

a microphone and an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment,

a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and

a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal, and

a binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid for provision of outputs for each of the first and second hearing aids for selection of the signal processing algorithm of each of the respective hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

2. A system according to claim 1, wherein the binaural sound environment detector having inputs for each of the digital input signals is provided in a remote control of the system.

3. A system according to claim 1, wherein the binaural sound environment detector is provided in one of the hearing aids and provides an output for the other hearing aid.

4. A system according to claim 1, wherein each of the hearing aids comprises a binaural sound environment detector.

5. A system according to any of the preceding claims, wherein the binaural sound environment detector comprises

a feature extractor for determination of characteristic parameters of the received sound signals,

an environment classifier for categorizing the sound environment based on the determined characteristic parameters, and

a parameter map for the provision of outputs for selection of the signal processing algorithms.

6. A system according to claim 4 and 5, wherein each of the feature extractors has inputs for each of the digital input signals.

7. A system according to claim 4 and 5, wherein each of the environment classifiers has inputs connected to each of the feature extractors.

8. A system according to claim 4 and 5, wherein each of the parameter maps has inputs for connected to each of the environment classifiers.

1/5

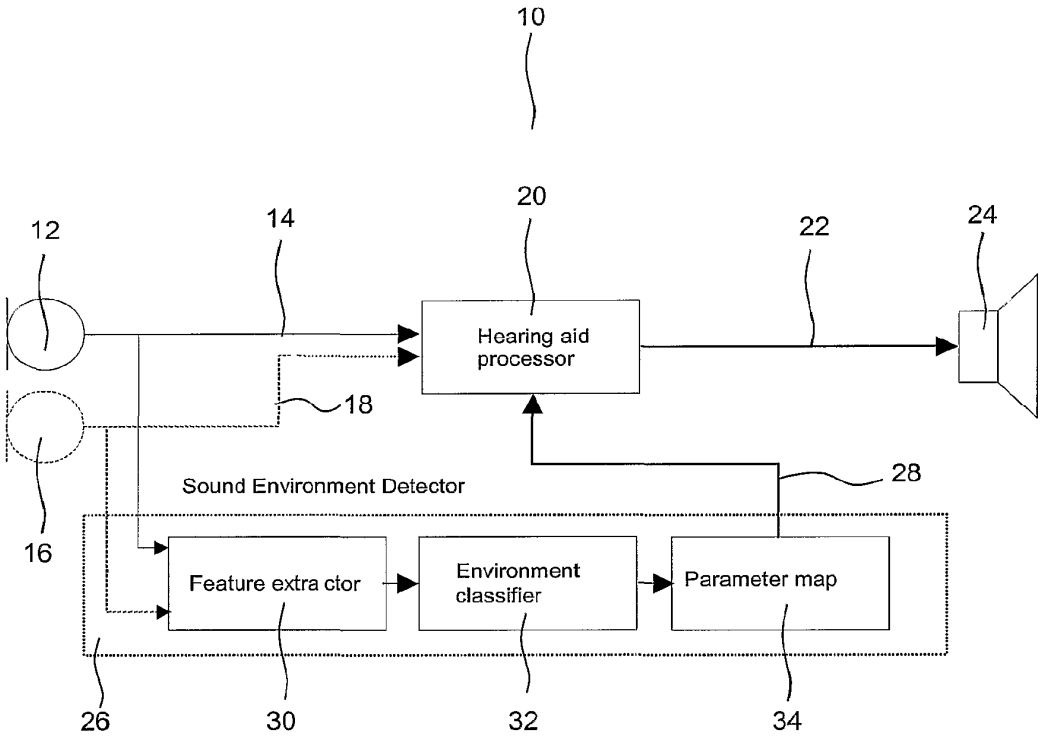

10
12
14
20
22
24
Hearing aid processor
18
16
Sound Environment Detector
28
Feature extra ctor
Environment classifier
Parameter map
26
30
32
34

Fig. 1

2/5

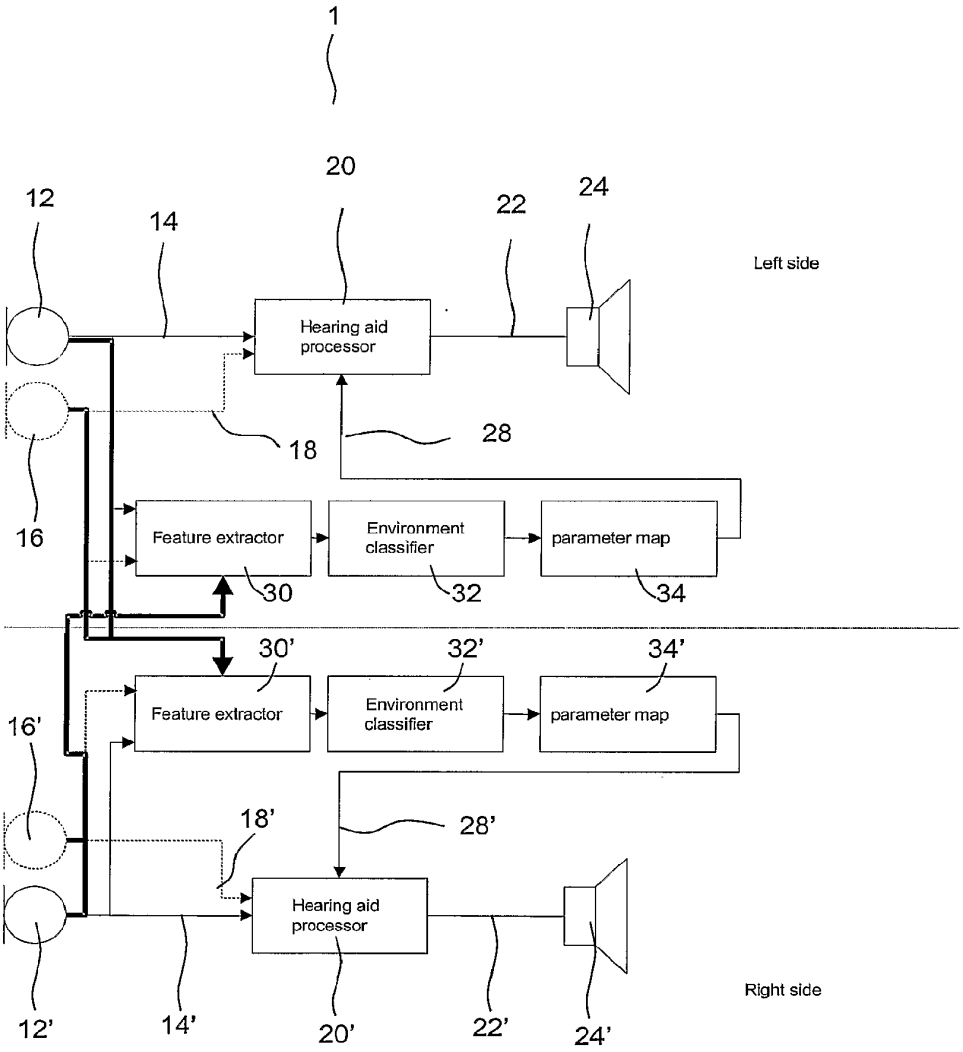

1
20
12
14
22
24
Left side
Hearing aid processor
18
28
16
Feature extractor
Environment classifier
parameter map
30
32
34
30'
32'
34'
Feature extractor
Environment classifier
parameter map
16'
18'
28'
Hearing aid processor
Right side
12'
14'
20'
22'
24'

Fig. 2

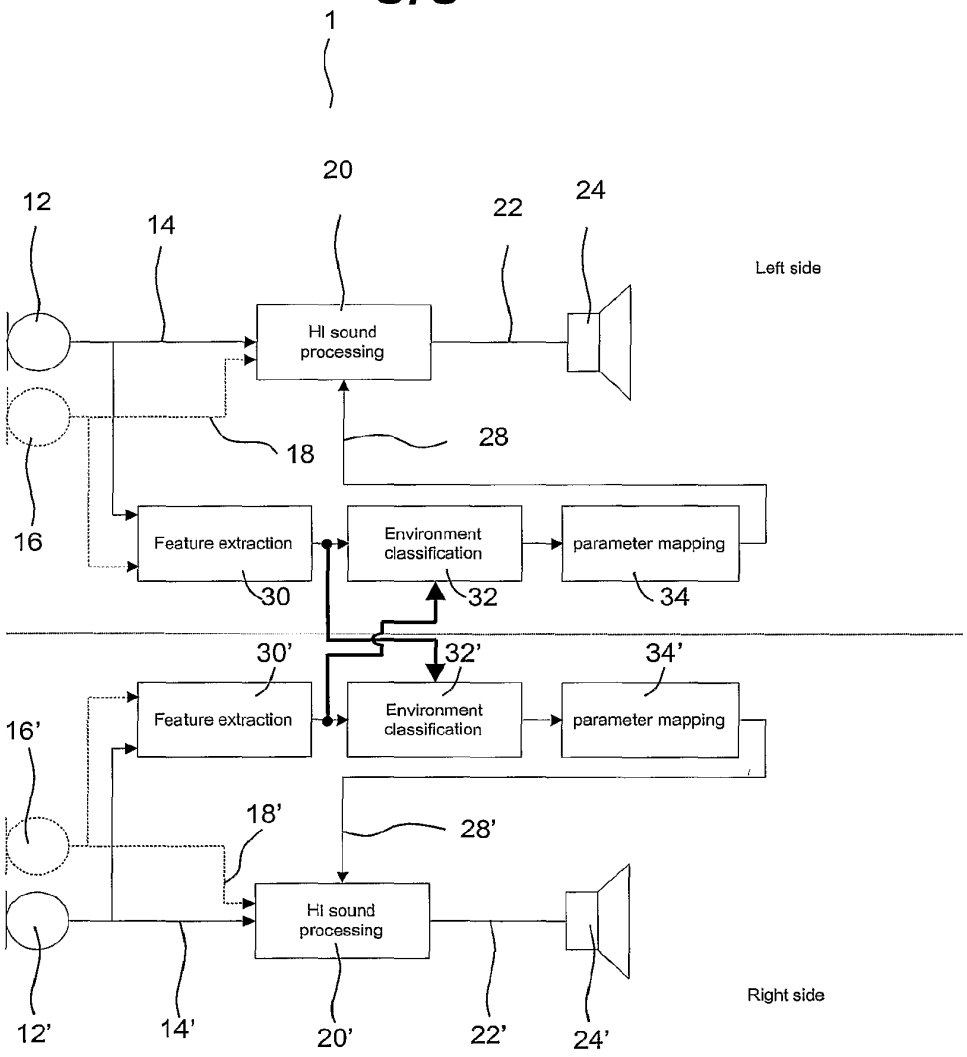

1
20
12
14
22
24
Left side
HI sound processing
18
28
16
Feature extraction
Environment classification
parameter mapping
30
32
34
30'
32'
34'
Feature extraction
Environment classification
parameter mapping
16'
18'
28'
HI sound processing
Right side
12'
14'
20'
22'
24'

Fig. 3

4/5

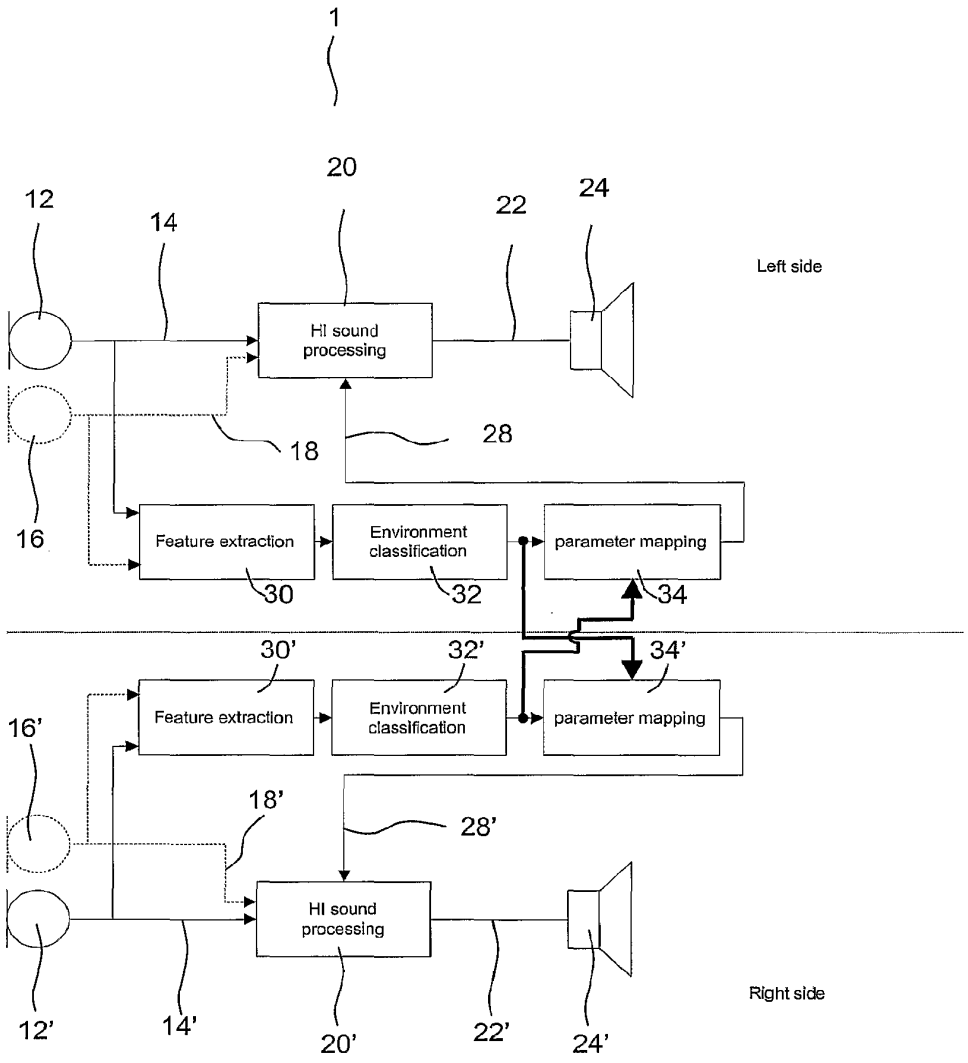

1
20
12
14
22
24
Left side
HI sound processing
18
28
16
Feature extraction
Environment classification
parameter mapping
30
32
34
30'
32'
34'
16'
Feature extraction
Environment classification
parameter mapping
18'
28'
HI sound processing
12'
14'
20'
22'
24'
Right side

Fig. 4

5/5

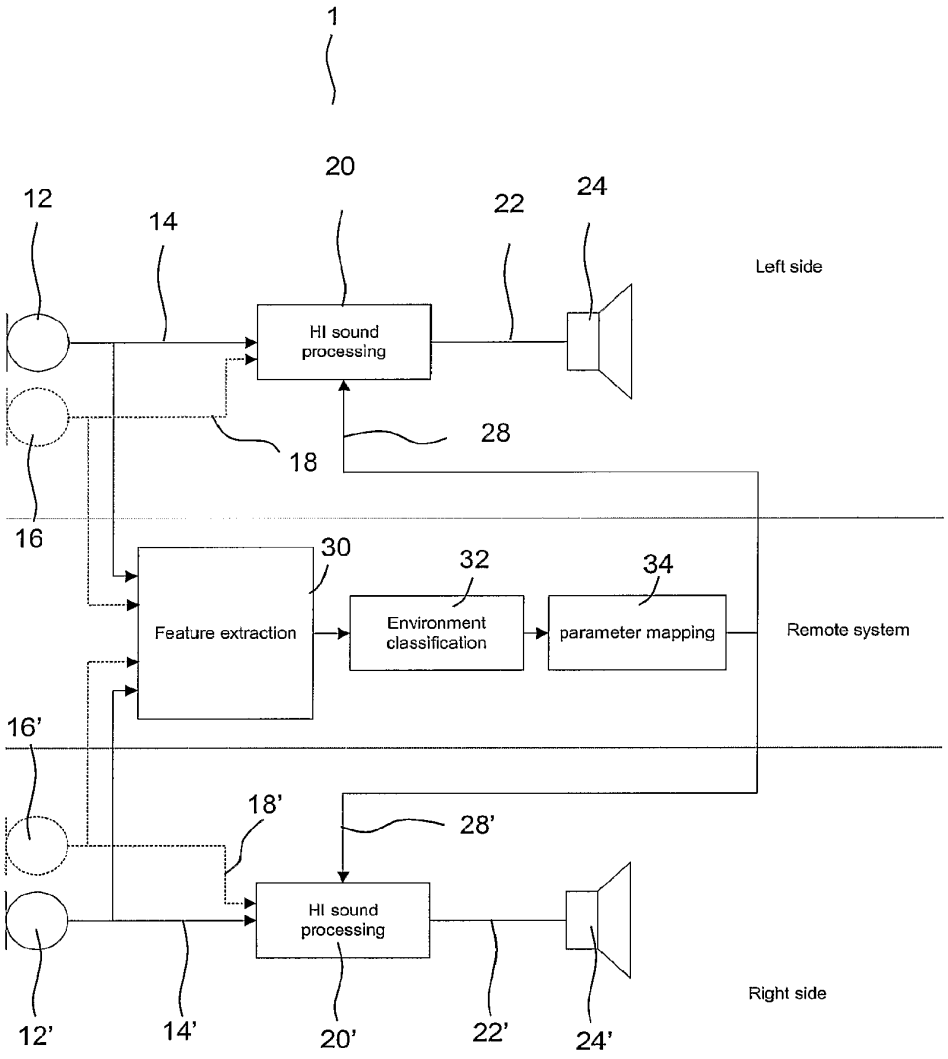

1
20
12
14
22
24
Left side
HI sound processing
28
18
16
30
32
34
Feature extraction
Environment classification
parameter mapping
Remote system
16'
18'
28'
HI sound processing
Right side
12'
14'
20'
22'
24'

Fig. 5

INTERNATIONAL SEARCH REPORT

International Application No
PCT/DK2004/000442

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04R25/00

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)
IPC 7 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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 02/28143 A (SIEMENS AUDIOLOGISCHE TECHNIK ; NIEDERDRAENK TORSTEN (DE)) 4 April 2002 (2002-04-04) the whole document	1-8
X	US 5 757 932 A (BISGAARD NIKOLAI ET AL) 26 May 1998 (1998-05-26) the whole document	1-8
A	US 6 549 633 B1 (WESTERMANN SOEREN ERIK) 15 April 2003 (2003-04-15) the whole document	1-8
A	US 2002/037087 A1 (ALLEGRO SYLVIA ET AL) 28 March 2002 (2002-03-28) the whole document	5
A	WO 02/32208 A (PHONAK AG ; ALLEGRO SILVIA (CH)) 25 April 2002 (2002-04-25)	

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
9 November 2004	18/11/2004

Name and mailing address of the ISA	Authorized officer
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Form PCT/ISA/210 (second sheet) (January 2004)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/DK2004/000442

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			CA	2439427 A1	25-04-2002
			EP	1470735 A2	27-10-2004

REQUEST FOR REFUND

Reference # *(Patent #, Application #, Trademark Serial #, Registration #, etc.)*	**Title of Invention or Mark Information**	
17/182,570; Pat. 11665493	Acoustic Sealing Analysis Sytem	
Attorney Docket # *(if applicable)*	**Payment Date** *(mm/dd/yyyy)*	**Refund Request Amount**
11472-36-3_CONT-16838277	08/24/2023	$ 800.00

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ELECTRONIC PAYMENT RECEIPT

APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/182,570	08/25/2023 04:29:44 PM ET	11472-36-3_CONT-16838277

Title of Invention

ACOUSTIC SEALING ANALYSIS SYSTEM

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	11665493
CONFIRMATION #	5671	FILED BY	Peter Chiabotti
PATENT CENTER #	62683036	AUTHORIZED BY	-
CUSTOMER #	150139	FILING DATE	02/23/2021
CORRESPONDENCE ADDRESS	-	FIRST NAMED INVENTOR	John Usher

Payment Information

PAYMENT METHOD	PAYMENT TRANSACTION ID	PAYMENT AUTHORIZED BY
DA / 500951	E20238OG30095407	Peter Chiabotti

FEE CODE	DESCRIPTION	ITEM PRICE($)	QUANTITY	ITEM TOTAL($)
2463	PETITIONS REQUIRING THE PETITION FEE SET FORTH IN 37 CFR 1.17(G) (GROUP II)	88.00	1	88.00
			TOTAL AMOUNT:	$88.00

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 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.

Application No.	:	17/182,570	Confirmation No.	:	5671
Applicant	:	John Usher			
Patent No.		11665493			
Examiner	:	Paul Kim			
Docket No.	:	11472-36-3_CONT-16838277			

37 CFR 1.59(b) Petition to Expunge Information Submitted in Incorrect Application

Applicant submits this petition to expunge information unintentionally submitted and authorizes the Patent Office to charge the 37 CFR 1.17(h) petition fees to Deposit Account 50-0951. Yesterday, a RCE, IDS and references were submitted in the incorrect file. MPEP notes "[w]here the Office can determine the correct application file that the papers were actually intended for, based on identifying information in the heading of the papers (e.g., application number, filing date, title of invention and inventor(s) name(s)), the Office will transfer the papers to the correct application file for which they were intended without the need of a petition." The MPEP describes the exact circumstances here. Applicant respectfully requests the below documents, and this petition, be expunged in their entirety.

Date	Doc Code	Doc Description	Pages
08/24/2023	RCEX	Request for Continued Examination (RCE)	3
08/24/2023	IDS	Information Disclosure Statement (IDS) Form (SB08)	2
08/24/2023	FOR	Foreign Reference	14
08/24/2023	FOR	Foreign Reference	19
08/24/2023	FOR	Foreign Reference	37
08/24/2023	N417.PYMT	Electronic Fee Payment	2
08/24/2023	FOR	Foreign Reference	27
08/24/2023	FOR	Foreign Reference	15
08/24/2023	N417	Electronic Filing System(EFS) Acknowledgment Receipt	4
08/24/2023	FOR	Foreign Reference	15
08/24/2023	FOR	Foreign Reference	24
08/24/2023	SPEC	Specification	27
08/24/2023	FOR	Foreign Reference	67
08/24/2023	FOR	Foreign Reference	19

Respectfully submitted,
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Date: August 25, 2023

UNITED STATES
PATENT AND TRADEMARK OFFICE

P.O. Box 1450
Alexandria, VA 22313 - 1450
www.uspto.gov

ELECTRONIC ACKNOWLEDGEMENT RECEIPT

APPLICATION #	RECEIPT DATE / TIME	ATTORNEY DOCKET #
17/182,570	**08/25/2023 04:29:44 PM ET**	**11472-36-3_CONT-16838277**

Title of Invention

ACOUSTIC SEALING ANALYSIS SYSTEM

Application Information

APPLICATION TYPE	Utility - Nonprovisional Application under 35 USC 111(a)	PATENT #	11665493
CONFIRMATION #	5671	FILED BY	Peter Chiabotti
PATENT CENTER #	62683036	FILING DATE	02/23/2021
CUSTOMER #	150139	FIRST NAMED INVENTOR	John Usher
CORRESPONDENCE ADDRESS	-	AUTHORIZED BY	-

Documents

TOTAL DOCUMENTS: 2

DOCUMENT	PAGES	DESCRIPTION	SIZE (KB)
Petition to Expunge.pdf	1	Petition for review by the Office of Petitions	67 KB
Refund_Request_Form.pdf	1	Refund Request	706 KB

Digest

DOCUMENT	MESSAGE DIGEST(SHA-512)
Petition to Expunge.pdf	9BCEFC2AE22D7B3198D6843CDCD52CECE1CA5B7A59A2851 ED364C0A703E5B8A7E73093BD0735AAF02D3833C245FBF7C9 602912FDC3098C6A03D800DB78CFB01C

Refund_Request_Form.pdf 87C8506092ABFE0DD78CD4BD2F3598CB638E6586A37C5BE9D47BF14A2326B2C62BE078B8A1BA995E2B86A4C6AF9E4DBC28B6C2C79DD92227843D2B3EBB2F137C

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