PTO/SB/57 (01-18)
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(Also referred to as FORM PTO-1465)

REQUEST FOR *EX PARTE* REEXAMINATION TRANSMITTAL FORM

Address to:
Mail Stop *Ex Parte* Reexam
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Attorney Docket No.: 0027071.00133

Date: February 24, 2023

1. ■ This is a request for *ex parte* reexamination pursuant to 37 CFR 1.510 of patent number 11,244,666 issued February 8, 2022. The request is made by:
   ☐ patent owner. ■ third party requester.

2. ■ The name and address of the person requesting reexamination is:
   Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc.
   c/o Arnold & Porter Kaye Scholer LLP
   601 Massachusetts Avenue, N.W., Washington, DC 20001

3. Requester ☐ asserts small entity status (37 CFR 1.27) or ☐ certifies micro entity status (37 CFR 1.29). Only a patent owner requester can certify micro entity status. Form PTO/SB/15A or B must be attached to certify micro entity status.

4. This request is accompanied by payment of the reexamination fee as set forth in:
   ■ 37 CFR 1.20(c)(2); **or**
   ☐ 37 CFR 1.20(c)(1). **In checking this box for payment of the fee set forth in 37 CFR 1.20(c)(1), requester asserts that this request has forty (40) or fewer pages and complies with all other requirements of 37 CFR 1.20(c)(1).**

   Payment of the reexamination fee is made by the method set forth below.
   a. ☐ A check in the amount of $_____ is enclosed to cover the reexamination fee;
   b. ☐ The Director is hereby authorized to charge the reexamination fee to Deposit Account No. _____;
   c. ☐ Payment by credit card. Form PTO-2038 is attached; **or**
   d. ■ Payment made via EFS-Web.

   ■ In addition, the Director is hereby authorized to charge any fee deficiencies to Deposit Account No. 50-2387.

5. ■ Any refund should be made by ☐ check or ■ credit to Deposit Account No. 50-2387. 37 CFR 1.26(c). If payment is made by credit card, refund must be to credit card account.

6. ■ A copy of the patent to be reexamined having a double column format on one side of a separate paper is enclosed. 37 CFR 1.510(b)(4).

7. ☐ CD-ROM or CD-R in duplicate, Computer Program (Appendix) or large table
   ☐ Landscape Table on CD

A Federal agency may not conduct or sponsor, and a person is not required to respond to, nor shall a person be subject to a penalty for failure to comply with an information collection subject to the requirements of the Paperwork Reduction Act of 1995, unless the information collection has a currently valid OMB Control Number. The OMB Control Number for this information collection is 0651-0064. Public burden for this form is estimated to average 18 minutes per response, including the time for reviewing instructions, searching existing data sources, gathering and maintaining the data needed, and completing and reviewing the information collection. Send comments regarding this burden estimate or any other aspect of this information collection, including suggestions for reducing this burden to the Chief Administrative Officer, United States Patent and Trademark Office, P.O. Box 1450, Alexandria, VA 22313-1450 or email InformationCollection@uspto.gov. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **If filing this completed form by mail, 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.

Samsung v. Techiya
IPR2024-01004
Exhibit 1009

8. ☐ Nucleotide and/or Amino Acid Sequence Submission
If applicable, items a. – c. are required.
   a. ☐ Computer Readable Form (CRF)
   b. Specification Sequence Listing on:
      i. ☐ CD-ROM (2 copies) or CD-R (2 copies) **or**
      ii. ☐ paper
   c. ☐ Statements verifying identity of above copies.

9. ☐ A copy of any disclaimer, certificate of correction or reexamination certificate issued in the patent is included.

10. ■ Reexamination of claim(s) 1-11 is requested.

11. ■ A copy of every patent or printed publication relied upon is submitted herewith including a listing thereof on Form PTO/SB/08, PTO-1449, or equivalent.

12. ☐ An English language translation of all necessary and pertinent non-English language patents and/or printed publications is attached.

13. ■ The attached detailed request includes at least the following items:
   a. A statement identifying each substantial new question of patentability based on prior patents and printed publications. 37 CFR 1.510(b)(1).
   b. An identification of every claim for which reexamination is requested, and a detailed explanation of the pertinency and manner of applying the cited art to every claim for which reexamination is requested. 37 CFR1.510(b)(2).

14. ☐ A proposed amendment is included (only where the patent owner is the requester). 37 CFR 1.510(e).

15. ■ It is certified that the statutory estoppel provisions of 35 U.S.C. 315(e)(1) or 35 U.S.C. 325(e)(1) do not prohibit requester from filing this *ex parte* reexamination request. 37 CFR 1.510(b)(6).

16. Service
   a. ■ It is certified that a copy of this request (if filed by other than the patent owner) has been served in its entirety on the patent owner as provided in 37 CFR 1.33(c).

      The name and address of the party served are:

      Akerman LLP

      777 South Flagler Drive, Suite 1100 West Tower

      West Palm Beach, FL 33401

      Date of Service: February 24, 2023

   OR

   b. ☐ A duplicate copy is enclosed since service on patent owner was not possible. An explanation of the efforts made to serve patent owner **is attached**. See MPEP 2220.

17. Correspondence Address: Direct all communication about the reexamination to:

☒ The address associated with Customer Number: 28381

OR

☐ Firm or Individual Name _____
(at the address identified below)

Address

City	State	Zip

Country

Telephone	Email

18. ☒ The patent is currently the subject of the following concurrent proceeding(s):

a. ☐ Copending reissue Application No. _____

b. ☐ Copending reexamination Control No. _____

c. ☐ Copending Interference No. _____

d. ☒ Copending litigation styled:

Staton Techiya, LLC v. Samsung Electronics Co., Ltd., No. 2:22-cv-00053-JRG-RSP (E.D. Tex.),

which has been consolidated with Staton Techiya, LLC v. Samsung Electronics Co., Ltd., No. 2:21-cv-00413-JRG-RSP (E.D. Tex.).

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

/Ali R. Sharifahmadian/ — Authorized Signature

February 24, 2023 — Date

Ali R. Sharifahmadian — Typed/Printed Name

48,202 — Registration No.

☐ For Patent Owner Requester ☒ For Third Party Requester

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. The United States Patent and Trademark Office (USPTO) collects the information in this record under authority of 35 U.S.C. 2. The USPTO's system of records is used to manage all applicant and owner information including name, citizenship, residence, post office address, and other information with respect to inventors and their legal representatives pertaining to the applicant's/owner's activities in connection with the invention for which a patent is sought or has been granted. The applicable Privacy Act System of Records Notice for the information collected in this form is COMMERCE/PAT-TM-7 Patent Application Files, available in the Federal Register at 78 FR 19243 (March 29, 2013). https://www.govinfo.gov/content/pkg/FR-2013-03-29/pdf/2013-07341.pdf

Routine uses of the information in this record may include disclosure to: 1) law enforcement, in the event that the system of records indicates a violation or potential violation of law; 2) a Federal, state, local, or international agency, in response to its request; 3) a contractor of the USPTO having need for the information in order to perform a contract; 4) the Department of Justice for determination of whether the Freedom of Information Act (FOIA) requires disclosure of the record; 5) a Member of Congress submitting a request involving an individual to whom the record pertains, when the individual has requested the Member's assistance with respect to the subject matter of the record; 6) a court, magistrate, or administrative tribunal, in the course of presenting evidence, including disclosures to opposing counsel in the course of settlement negotiations; 7) the Administrator, General Services Administration (GSA), or their designee, during an inspection of records conducted by GSA under authority of 44 U.S.C. 2904 and 2906, in accordance with the GSA regulations and any other relevant (i.e., GSA or Commerce) directive, where such disclosure shall not be used to make determinations about individuals; 8) 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)); 9) the Office of Personnel Management (OPM) for personnel research purposes; and 9) the Office of Management and Budget (OMB) for legislative coordination and clearance.

If you do not furnish the information requested on this form, the USPTO may not be able to process and/or examine your submission, which may result in termination of proceedings, abandonment of the application, and/or expiration of the patent.

Additional Uses

Additional USPTO uses of the information in this record may include disclosure to: 1) the International Bureau of the World Intellectual Property Organization, if the record is related to an international application filed under the Patent Cooperation Treaty; 2) the public i) after publication of the application pursuant to 35 U.S.C. 122(b), ii) after issuance of a patent pursuant to 35 U.S.C. 151, iii) 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, or iv) without publication of the application or patent under the specific circumstances provided for by 37 CFR 1.14(a)(1)(v)-(vii); and/or 3) the National Archives and Records Administration, for inspection of records.

Exhibit PA-A

US006647368B2

(12) **United States Patent**
Nemirovski

(10) **Patent No.: US 6,647,368 B2**
(45) **Date of Patent: *Nov. 11, 2003**

(54) **SENSOR PAIR FOR DETECTING CHANGES WITHIN A HUMAN EAR AND PRODUCING A SIGNAL CORRESPONDING TO THOUGHT, MOVEMENT, BIOLOGICAL FUNCTION AND/OR SPEECH**

(75) Inventor: **Guerman G. Nemirovski**, St. Petersburg, FL (US)

(73) Assignee: **Think-A-Move, Ltd.**, Cleveland, OH (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **09/897,696**

(22) Filed: **Jul. 2, 2001**

(65) **Prior Publication Data**

US 2002/0143242 A1 Oct. 3, 2002

Related U.S. Application Data

(60) Provisional application No. 60/280,282, filed on Mar. 30, 2001.

(51) **Int. Cl.**[7] **G10L 15/08**; A61B 5/00

(52) **U.S. Cl.** **704/270**; 704/271; 704/275; 600/300

(58) **Field of Search** 704/270, 271, 704/275

(56) **References Cited**

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Claims from Ser. No. 09/919,291; filed Jul. 31, 2001.
Claims from Ser. No. 09/706,091; filed Nov. 3, 2000, and as amended May 7, 2002.

(List continued on next page.)

Primary Examiner—Tālivaldis Ivars Šmits
(74) *Attorney, Agent, or Firm*—Renner, Otto, Boisselle & Sklar LLP

(57) **ABSTRACT**

A pair of sensors are used for detecting an air pressure change signal within an ear of a person caused by the person's initiating action (thought, movement, biological function and/or speech). One of the microphones is placed at least partially within an ear of the person and the other is placed adjacent to and external to the ear, to produce two electrical signals, respectively corresponding to internally detected and to externally detected changes in air pressure. Comparison of the unmodified signal strength difference between these two signals is used to distinguish an initiating action component of each signal from an external source component of each signal. The electrical signals are processed to produce an output signal corresponding to the initiating action, which signal is then recognized by a neural network or speech recognizer, and used for control or communication.

20 Claims, 8 Drawing Sheets

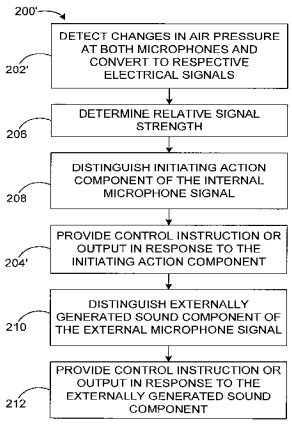


Exhibit PA-A

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Claims from Serial No. 09/919,291; Filed Jul. 31, 2001.

Claims from Serial No. 09/706,091; Filed Nov. 3, 2000, and as amended May 7, 2002.

* cited by examiner

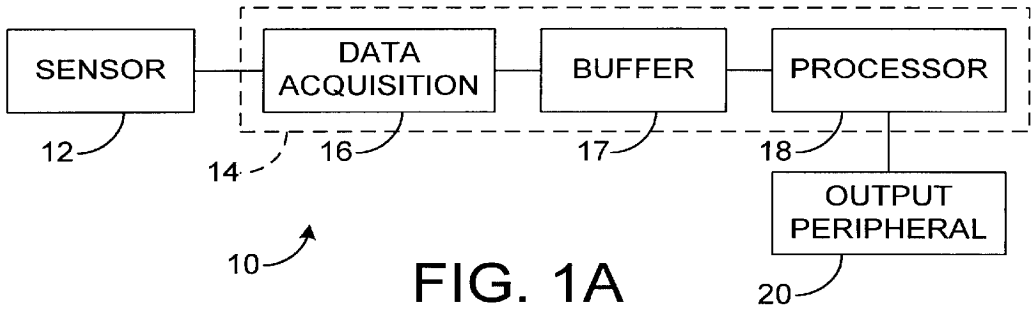


FIG. 1A

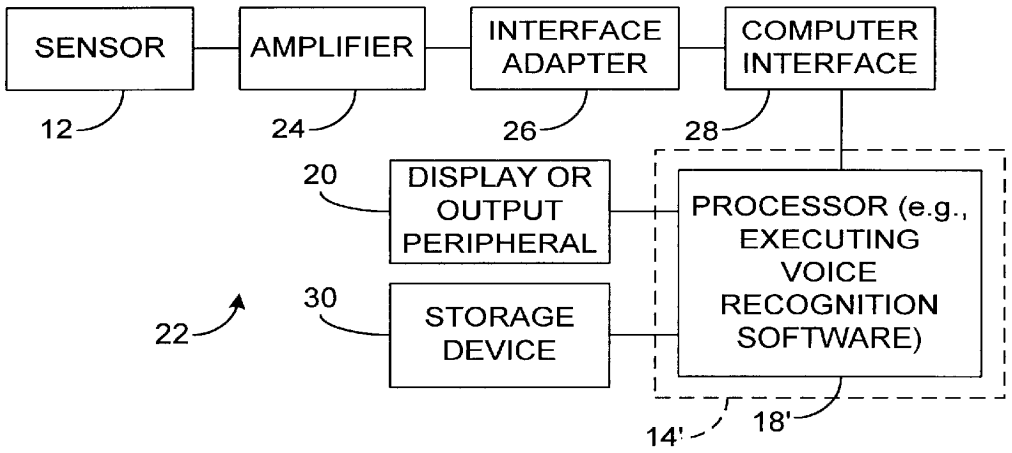


FIG. 1B

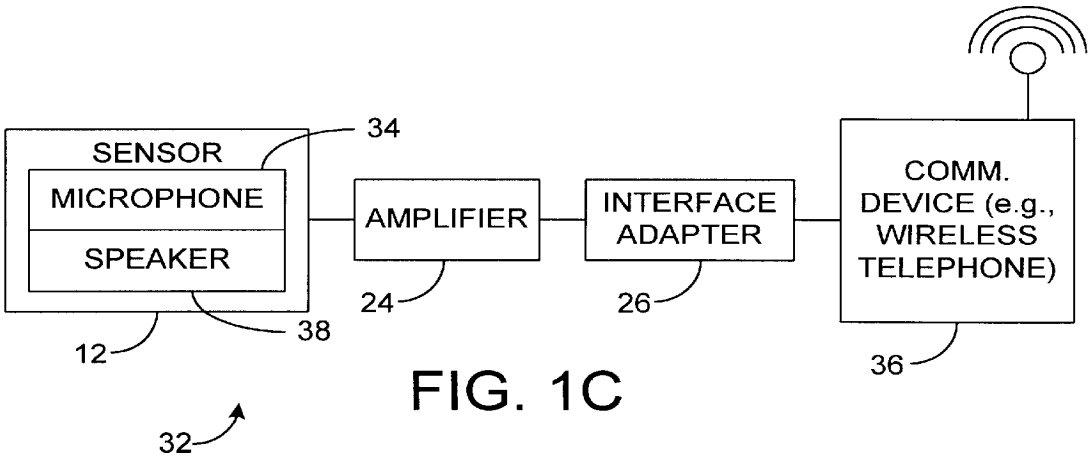


FIG. 1C

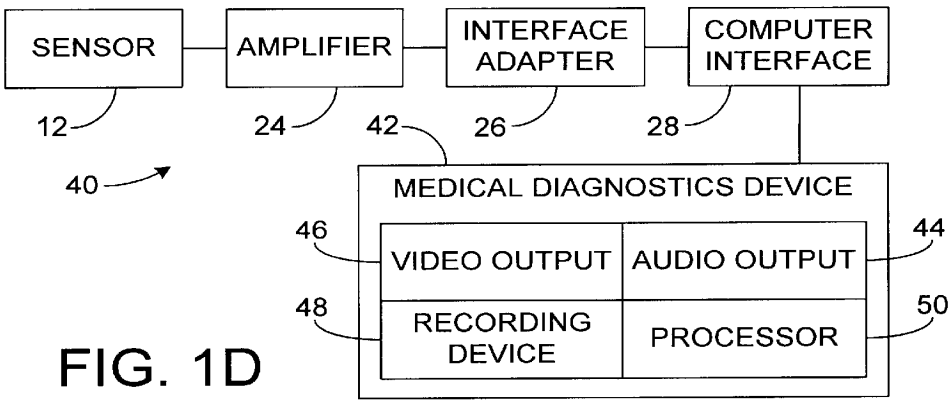


FIG. 1D

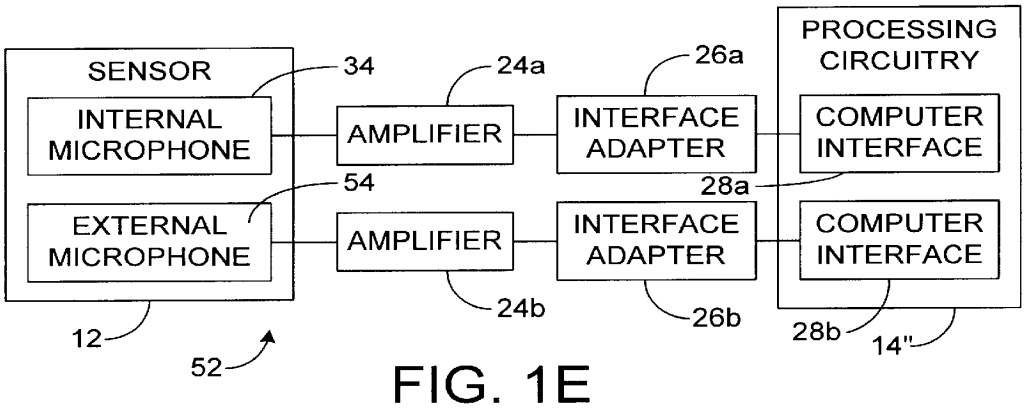


FIG. 1E

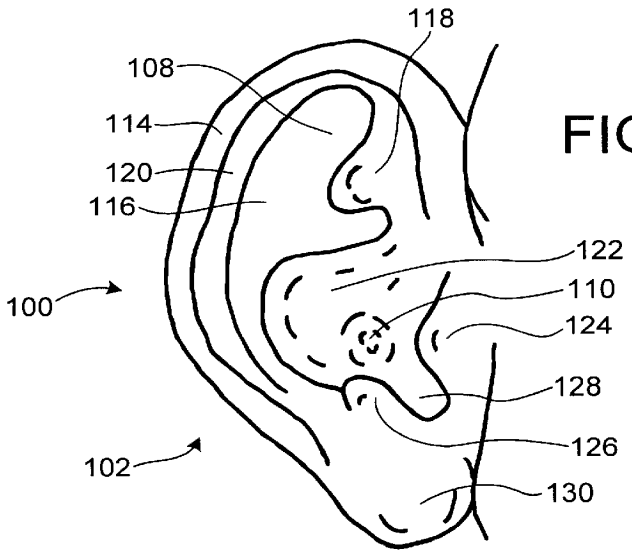


FIG. 2A

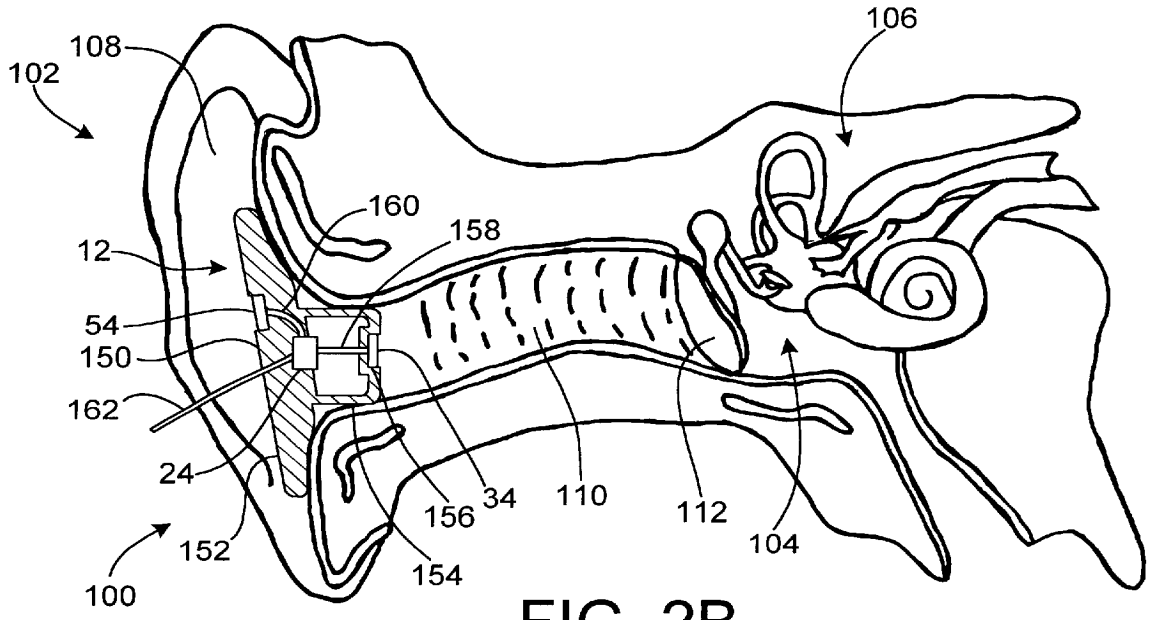

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FIG. 2B

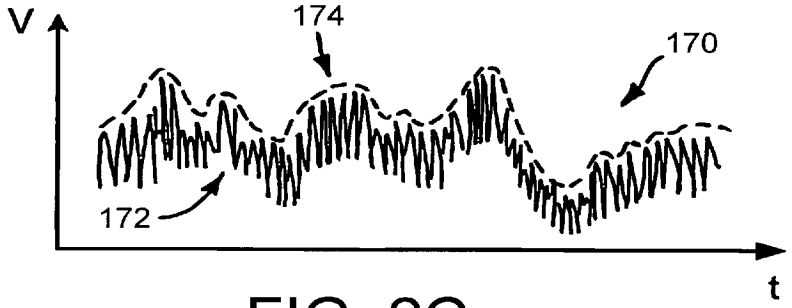

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FIG. 2C

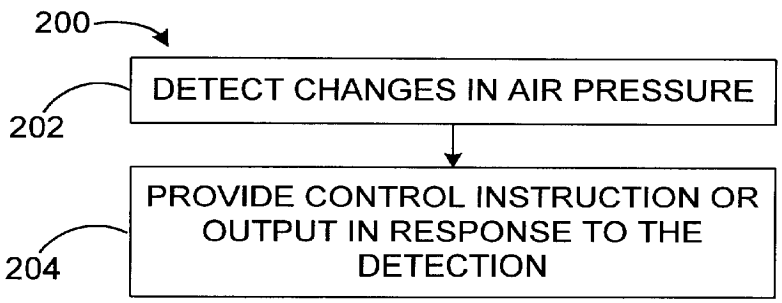

200
DETECT CHANGES IN AIR PRESSURE
202
PROVIDE CONTROL INSTRUCTION OR OUTPUT IN RESPONSE TO THE DETECTION
204

FIG. 3A

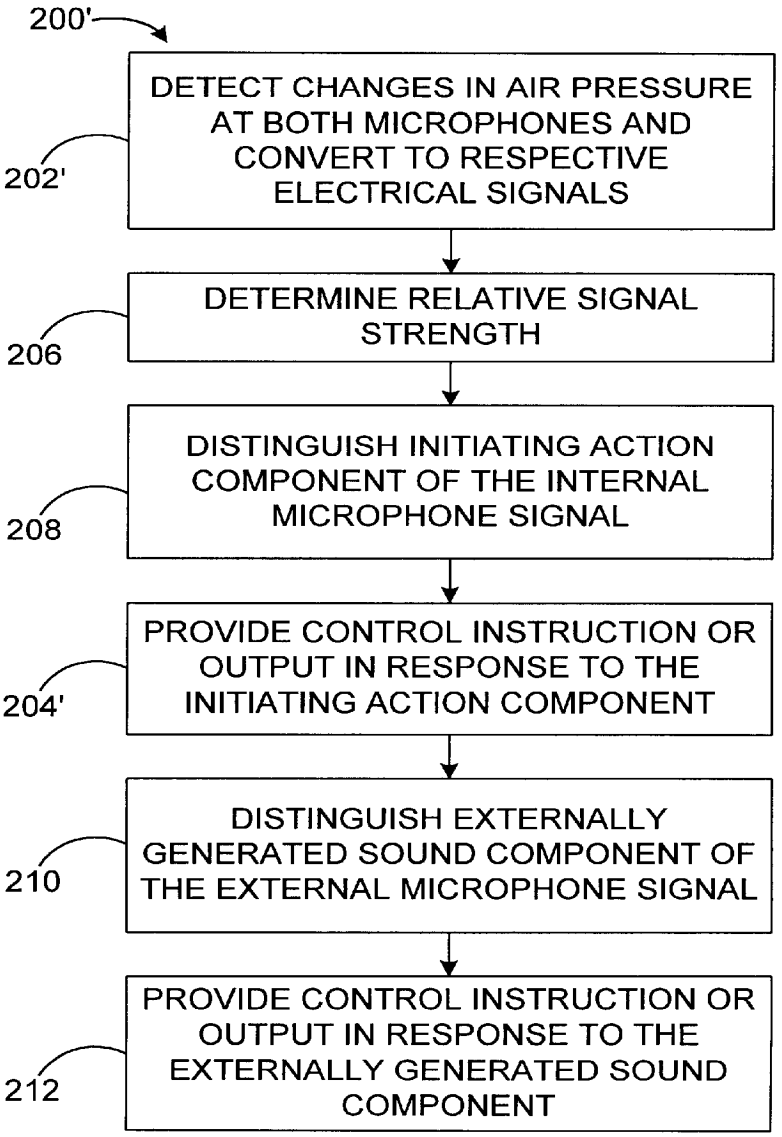

200'
DETECT CHANGES IN AIR PRESSURE AT BOTH MICROPHONES AND CONVERT TO RESPECTIVE ELECTRICAL SIGNALS
202'
DETERMINE RELATIVE SIGNAL STRENGTH
206
DISTINGUISH INITIATING ACTION COMPONENT OF THE INTERNAL MICROPHONE SIGNAL
208
PROVIDE CONTROL INSTRUCTION OR OUTPUT IN RESPONSE TO THE INITIATING ACTION COMPONENT
204'
DISTINGUISH EXTERNALLY GENERATED SOUND COMPONENT OF THE EXTERNAL MICROPHONE SIGNAL
210
PROVIDE CONTROL INSTRUCTION OR OUTPUT IN RESPONSE TO THE EXTERNALLY GENERATED SOUND COMPONENT
212

FIG. 3B

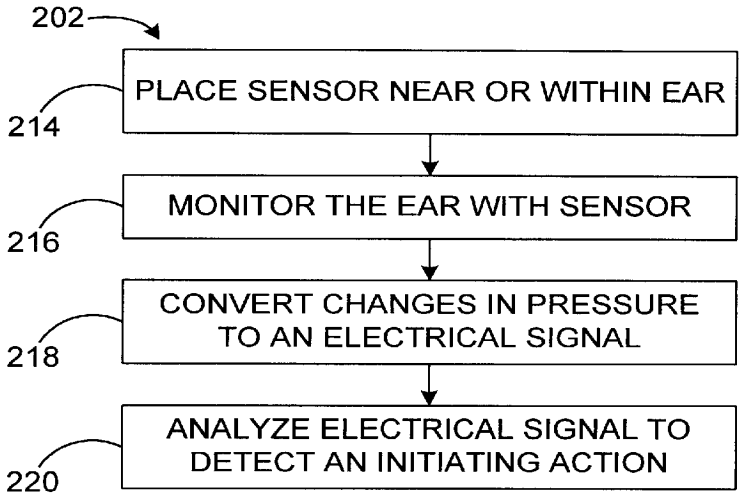


FIG. 4

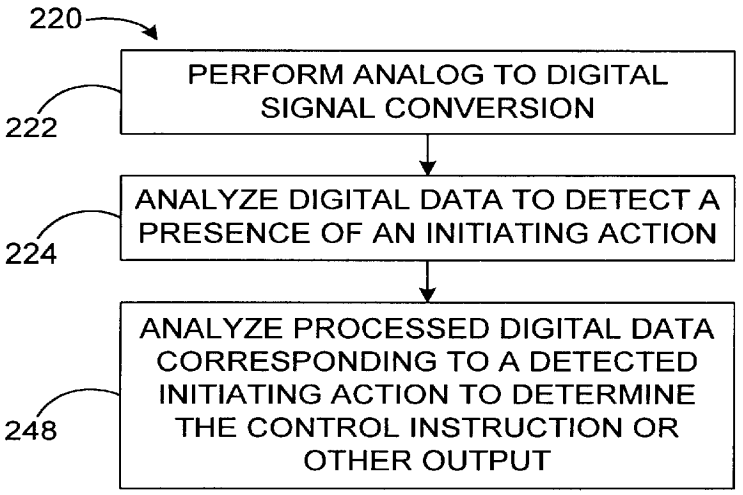


FIG. 5

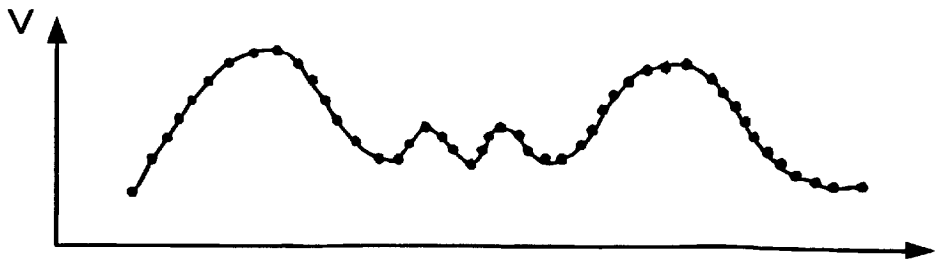


FIG. 6

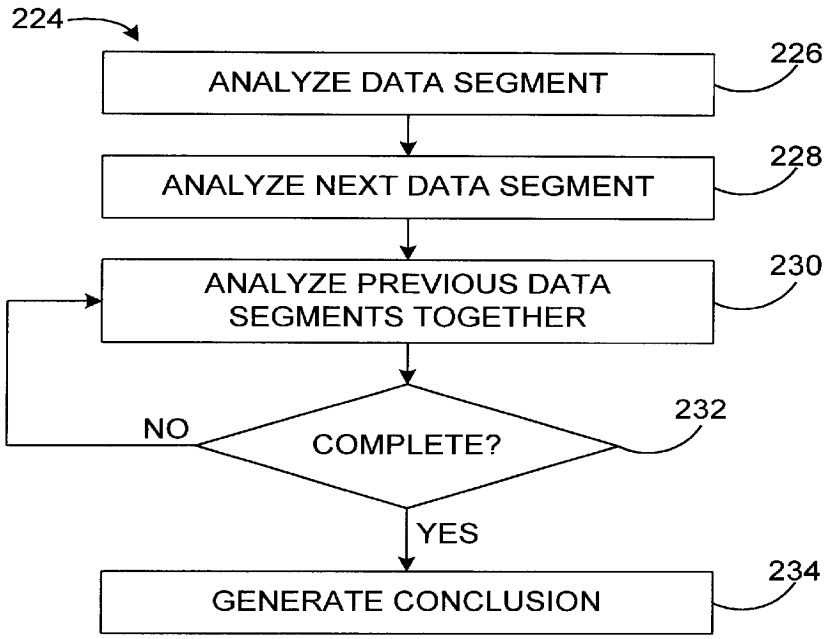

224
ANALYZE DATA SEGMENT
226
ANALYZE NEXT DATA SEGMENT
228
ANALYZE PREVIOUS DATA SEGMENTS TOGETHER
230
NO
COMPLETE?
232
YES
GENERATE CONCLUSION
234

FIG. 7

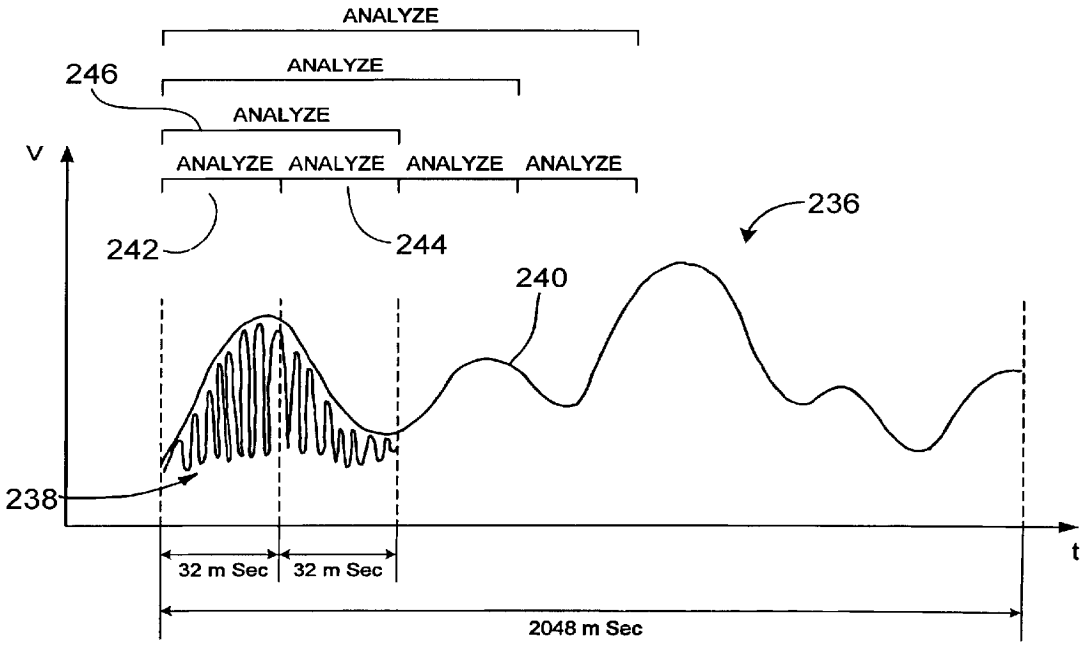

ANALYZE
ANALYZE
246
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V
236
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32 m Sec
32 m Sec
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FIG. 8

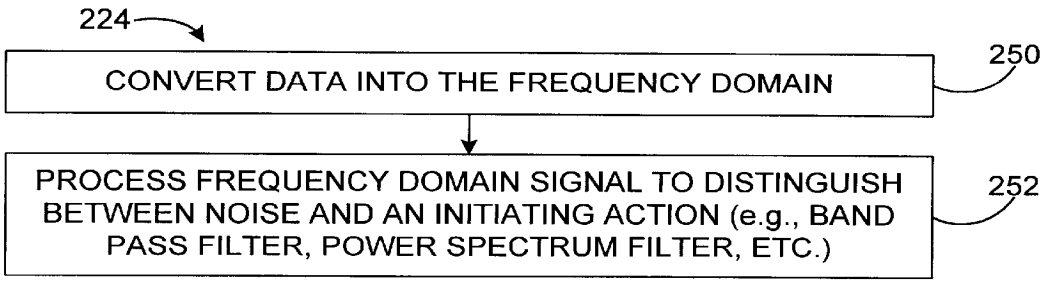

224
CONVERT DATA INTO THE FREQUENCY DOMAIN
250
PROCESS FREQUENCY DOMAIN SIGNAL TO DISTINGUISH BETWEEN NOISE AND AN INITIATING ACTION (e.g., BAND PASS FILTER, POWER SPECTRUM FILTER, ETC.)
252

FIG. 9

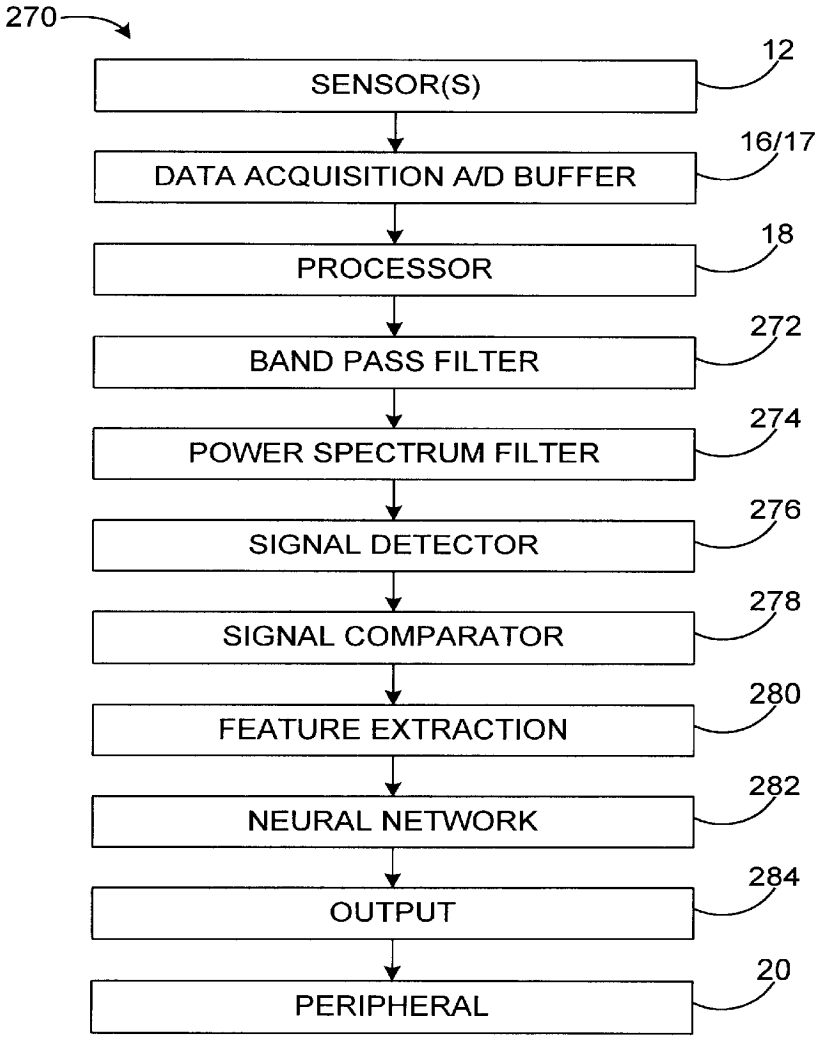

270
SENSOR(S)
12
DATA ACQUISITION A/D BUFFER
16/17
PROCESSOR
18
BAND PASS FILTER
272
POWER SPECTRUM FILTER
274
SIGNAL DETECTOR
276
SIGNAL COMPARATOR
278
FEATURE EXTRACTION
280
NEURAL NETWORK
282
OUTPUT
284
PERIPHERAL
20

FIG. 10



FIG. 11

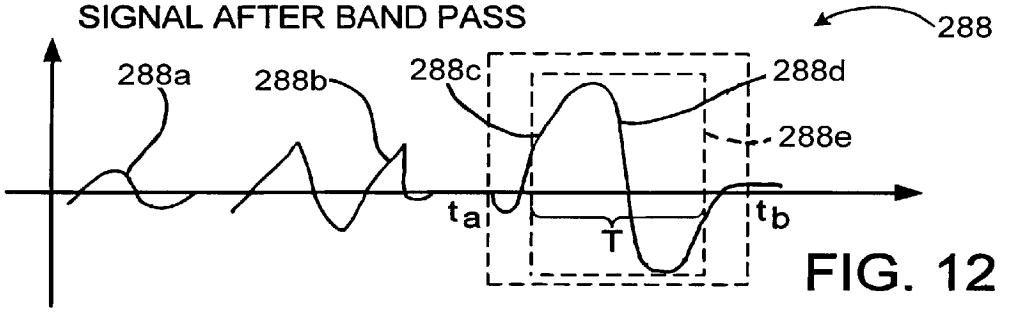


FIG. 12

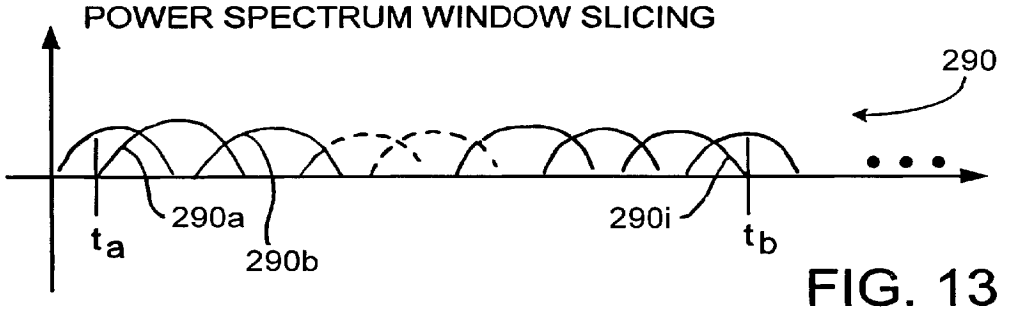


FIG. 13

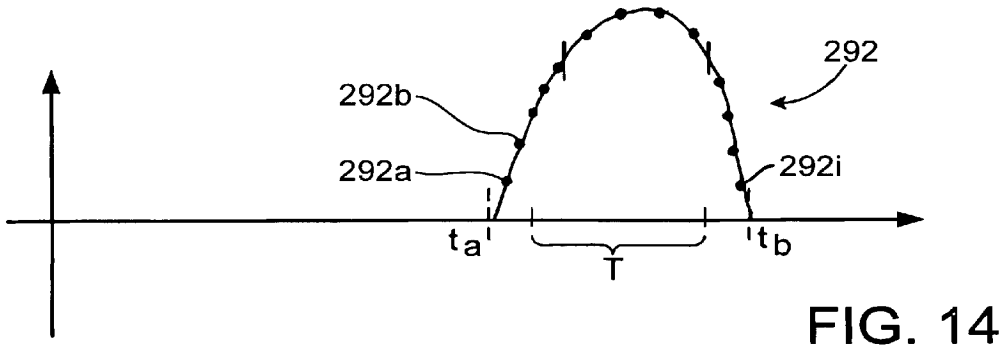


FIG. 14

SENSOR PAIR FOR DETECTING CHANGES WITHIN A HUMAN EAR AND PRODUCING A SIGNAL CORRESPONDING TO THOUGHT, MOVEMENT, BIOLOGICAL FUNCTION AND/OR SPEECH

RELATED APPLICATION(S)

This application claims the benefit of U.S. Provisional Application No. 60/280,282, filed Mar. 30, 2001, the entire disclosure of which is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention generally relates to a sensor placed with respect to an ear. More particularly, the sensor is used to detect physical or mental actions of a person such as speech, thoughts, movements of the tongue with respect to the oral cavity, biological functions, etc. and provides an output which can be converted into system control functions.

BACKGROUND OF THE INVENTION

An apparatus and method for detecting thoughts is disclosed in U.S. Pat. No. 6,024,700, the entire disclosure of which is incorporated by reference. The thought may lead to a detectable air pressure or sound at one or both ears of a user which can be detected by a detector, such as a microphone. Exemplary outputs from such system and method for detecting thoughts may be one or plural discrete outputs, such as discrete outputs representing respective alphanumeric characters, directions, etc.

An apparatus and method for detecting physical action, such as mouth and tongue movement, is disclosed in U.S. patent application Ser. No. 09/706,091, filed Nov. 3, 2000, the entire disclosure of which is incorporated by reference. The physical action leads to a detectable air pressure or sound at one or both ears of a user.

Sounds produced by a person's ears sometimes are referred to as autoaccoustic. Autoaccoustic sounds have been used in the past for diagnostics, e.g., to indicate ear function or a condition of the ear. For example, in response to a sound input to the ear, the ear has been found to provide a detectable autoaccoustic output.

At present there are available controllers to enable a handicapped person, who cannot use arms or legs, to operate a motorized wheelchair. Such a controller includes a trackball or joystick type device that is inserted into the mouth of an individual; and the individual operates the controller using mouth and/or tongue movements to provide mechanical inputs. Such device is difficult to operate and is problematic if it falls from the mouth or irritates the mouth.

Other controllers exist to enable handicapped individuals to perform functions with mechanical assists. Operation of such controllers typically requires a signal, such as an incident electromagnetic signal, e.g., infrared or ultraviolet light/radiation, which is directed at the individual; then, by a specified action or activity the individual can modulate the incident electromagnetic signal, e.g., by blinking the eyes, moving the head, or some such action. The need for incident electromagnetic signal is disadvantageous for several reasons: an extra piece of equipment is required to provide that signal, the signal must be properly directed or aligned, the detector and the modulated signal must be properly aligned, the signal should not have a detrimental affect on the user (such as causing eye injury), etc.

At present there are available voice recognition software programs which receive an input signal representative of human speech and convert that signal into text format or carry out a command upon interpreting the speech. The input signal is generated by a microphone disposed in front of the speaker's mouth.

SUMMARY OF THE INVENTION

According to one aspect of the invention, the invention is a method of detecting an air pressure change within an ear of a person. The method includes steps of placing a microphone at least partially within an ear of the person; detecting the change in air pressure within the ear of the person using the microphone, the change in air pressure corresponding to an initiating action made by person; producing an electrical signal corresponding to the internally detected change in air pressure; and processing the electrical signal to produce a corresponding output.

According to another aspect of the invention, the invention is a detection system. The detection system includes a housing positionable with respect to an ear of a person; a microphone disposed with respect to the housing for at least partial insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person makes an initiating action and to produce an electrical signal corresponding to the internally detected change in air pressure; and processing circuitry coupled to the microphone for processing the electrical signal to produce a corresponding output.

According to yet another aspect of the invention, the invention is a sensor for detecting changes in air pressure in a human ear. The sensor includes a housing having an interior portion adapted for at least partial insertion into the ear of a person and an external portion adapted to engage at least a portion of a pinna of the ear; an internal microphone dispose with respect to the interior portion of the housing to detect air pressure changes emanating from within the ear; and an exterior microphone dispose with respect to the exterior portion of the housing to detect air pressure changes emanating externally to the ear.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1**A is a block level diagram illustrating a system by which air pressure fluctuations are detected and used to provide a control function which corresponds to the detected air pressure according to the present invention;

FIG. **1**B is a block level diagram illustrating a system by which air pressure fluctuations are detected and processed, and in the illustrated embodiment, processed by a voice recognition software program to interpret speech of a person;

FIG. **1**C is a block level diagram illustrating a system by which air pressure fluctuations are detected and re-transmitted by a communications device;

FIG. **1**D is a bock level diagram illustrating a system by which air pressure fluctuations are detected and used by a medical diagnostics device;

FIG. **1**E is a block diagram illustrating a system having a sensor with an internal microphone and an external microphone;

FIG. **2**A is a schematic view of an ear;

FIG. **2**B is a cross-sectional view of a sensor embodiment for monitoring air pressure changes within the ear of a person;

FIG. **2**C is a graph illustrating an exemplary electrical signal produced by a sensor according to the present invention;

FIG. **3A** is a flow chart diagram illustrating a method of detecting changes in air pressure and providing a control instruction in response to the detected changes according to the present invention;

FIG. **3B** is a flow chart diagram illustrating another method of detecting changes in air pressure and providing a control instruction in response to the detected changes according to the present invention;

FIG. **4** is a flow chart diagram illustrating a method of monitoring according to the present invention;

FIG. **5** is a flow chart diagram illustrating a method of processing an electrical signal corresponding to a mouth/tongue action, a thought, speech and/or another type of initiating action;

FIG. **6** is a graph illustrating the conversion of an analog electrical signal to digital signal data according to the present invention;

FIG. **7** is a flow chart diagram illustrating a method of analyzing the digital signal data according to the present invention;

FIG. **8** is a graph illustrating a method of analyzing the digital signal data with the flow chart of FIG. **7** according to the present invention;

FIG. **9** is a flow chart diagram illustrating a method of processing the digital signal data according to the present invention;

FIG. **10** is a system diagram showing a system including parts and functions of the invention;

FIG. **11** is a graph representing exemplary raw signal information detected by a sensor used in the invention to detect air pressure changes;

FIG. **12** is a graph of frequency filtered signal information of FIG. **11**, the frequency filtering to remove noise, for example;

FIG. **13** is a graph representing power spectrum window slicing of the curve of FIG. **12**; and

FIG. **14** is a graph of such power spectrum.

DETAILED DESCRIPTION OF THE INVENTION

The following is a detailed description of the present invention made in conjunction with the attached figures, wherein like reference numerals will refer to like elements throughout. To illustrate the present invention in a clear and concise manner, the drawings may not necessarily be to scale and certain features may be shown in somewhat schematic form.

The present invention is directed to apparatus and method for detecting physical and/or mental actions of a person (such as actions occurring in association with the thoughts of the person, the speech of the person and/or movements associated with the oral cavity) by monitoring air pressure changes in or near a person's ear over a period of time. As an example, motion of the tongue in the oral cavity can be detected or sensed to provide a useful output. As another example, the action can be mental, as in thinking a thought, which produces a detectable air pressure change within and/or adjacent the ear as discussed in greater detail in U.S. Pat. No. 6,024,700. More specifically, the thought causes a reaction in the human anatomy which effectuates a change in the ear (perhaps by muscle contraction), thereby producing a detectable change in air pressure in or near the ear canal. An exemplary output may contain information analogous to an information signal produced by a computer mouse, the click buttons on the mouse, a computer joystick, or the like.

The present invention has been found to be more aesthetically pleasing and comfortable to the user and detects less environmental, or ambient, noise than the systems disclosed in the above-mentioned U.S. Pat. No. 6,024,700 and in U.S. patent application Ser. No. 09/706,091. As will be described in more detail below, the sensor of the present invention includes a microphone disposed with respect to an ear of a user and detects air pressure changes within the ear. This microphone will be referred to herein as an internal microphone. The air pressure changes may be caused by one or more factors, including vibrations of the ear drum, vibrations of bone within the ear, vibrations of other anatomical structures within the ear and vibrations conducted by the bones and/or tissue of the user to the ear and which invoke an air pressure change in the ear. As a result, the sensor can be used to detect a person's speech. The term speech, as used herein, is intended to include spoken words and utterances, singing and other vocalizations produced by a person. Such detected speech can be used for many purposes including, for example, processing the detected speech with voice recognition software or re-transmitting the detected speech with a communications device such as a cellular telephone (thereby allowing separation of the speech input device from the RF transceiver).

In addition to the internal microphone, the present invention can also include an external microphone. The external microphone detects air pressure changes, or sounds, originating outside the user's ear. As described below in more detail, these sounds are converted into electrical signals and compared with electrical signals corresponding to the sounds detected by the internal microphone. The respective signals can be processed to discern air pressure changes within the ear from externally produced ambient noise, speech and the like.

The present invention can also be used as a medical diagnostic tool by detecting and monitoring various biological processes, including a person's heartbeat, breathing and gastronomical events, which cause detectable changes in air pressure in or near the ear. Therefore, by monitoring air pressure adjacent or within the ear, a person's heartbeat, breathing and any irregularities associated therewith can be studied. The sounds detected by the sensor of the present invention can be amplified to a level to be studied by medical personnel and/or processed by a device programmed to analyze such biological functions.

The present invention is able to provide an output having a continuity characteristic, e.g., similar to the signal produced by moving a computer mouse along a surface. The electrical signal produced by the sensor can contain discernable information which was generated by two or three dimensional movements of certain body parts, particularly movements of the tongue, jaw, and oral and nasal cavities. The resulting signal output from the sensor is analogous to the signal produced by a computer joystick that has up, down, left, right, front and back signal outputs. Intensity or other components of the signal can be used as a measure of the action's speed characteristics.

By measuring air pressure in both ears with separate sensors, at least two electrical signals can be produced and processed, resulting in a two dimensional (or stereophonic) information pattern. The information derived from each sensor, along with similarities and differences in the data streams, can be used to discern additional information from the person. In some cases, the additional information will improve the generation of control commands, detection of thoughts, recognition of speech, detection of biological processes and other functions of the systems described herein.

The invention may be used for many additional purposes. One exemplary use is to provide an input to a computer or to some other electrical or electronic device. An exemplary input is an electrical signal, but it will be appreciated that the signal generated by the sensor used to detect air pressure changes may be other than electrical, or an electrical signal generated by the sensor can be converted into a form which is not electrical. Example alternative signal types include, for example, an optical signal, a fluidic signal, a radio frequency (RF) signal, etc. The input may control the computer or other device or it may be used for some other purpose. To facilitate and to avoid complexity of the description of the invention herein, the invention is described with respect to a use to provide an electrical input to a computer, for example, a personal computer or some other computer, whether of the digital or analog type, to control the operation of the computer. It will be appreciated that the invention may be used for many other purposes.

One example of a use of the invention in combination with a computer which is controlled using the invention is to provide functions and capabilities for a handicapped person. For example, the invention may be used to enable a handicapped individual or a non-handicapped individual to control operation of a piece of machinery in a manufacturing facility, to control operation of a motorized wheelchair or similar device, to control operation of a video game, etc. These are only examples, and it will be appreciated that the invention may be used for many other purposes.

In the present invention there is no need for the individual to be subjected to a stimulus signal, such as an incident acoustic or electromagnetic signal. Rather, in the present invention, actions of the individual (e.g., a tongue movement involving clicking of the tongue against the teeth or pallet) are used to create air pressure or sounds at one or both ears of the individual, which can be detected or sensed and used as an output of the invention to be provided as an input to a computer or to some other device. Thus, the present invention presents an unstimulated signal, namely a signal that is produced by action of the individual without the need for a stimulus to be directed at the individual.

In accordance with an embodiment of the invention a sensor senses or detects air pressure, a change in air pressure or a sound that is produced by an ear of an individual. These pressure changes may be produced by the person's ear drum or other anatomical structure within the ear. In addition, the pressure changes may originate in other parts of the person's body and conducted to the ear by bone or tissue where a vibration or air pressure change is produced and detected by the sensor.

To facilitate the description herein, the event of sensing or detecting will be referred to as detecting and that which is detected will be referred to as a change within the ear. Also, to facilitate the description herein, the action causing the detectable change will be described as an "initiating action." The initiating action can be one or more physical or mental actions, including, but not limited to, thinking a thought, speaking, movement of the tongue in the oral cavity (sometimes referred to as "tongue action"), other movements or actions in or associated with the oral or nasal cavities, clicking or other actions of the tongue against the teeth or pallet, exhaling or inhaling by the lungs, beating of the heart and actions of the nose. As an example, tongue movements lead to the detectable air pressure. These movements can be two dimensional (e.g., up/down, left/right, in/out, etc.) or three dimensional (any combination of front and back, left and right and up and down) which generate a corresponding output by the sensor, even if that output is one dimensional in the case of one sensor or two dimensional in the case of two sensors (one for each ear).

The present invention monitors changes within the human ear which occur instantaneously, or nearly instantaneously, in response to the initiating action to provide a substantially real-time detection and control system. In addition, the monitoring of the changes is passive and thus avoids potential health and/or environmental concerns related to subjecting the body to electrical signals and/or radiation. Furthermore, the changes in the ear uniquely correspond to one of a variety of initiating actions (e.g., have unique signal signatures) which allows a plurality of different initiating actions to be detected and distinguished for a system requiring multiple control functions.

According to one aspect of the present invention, the sensor is in electrical communication with processing circuitry and senses changes of the ear and converts the sensed changes into an electrical signal. The electrical signal is then processed to provide a control instruction or some other useful output, such as an output similar to that produced by a computer mouse or joystick which corresponds to the particular initiating action. Exemplary control functions may include, but are not limited to: controlling a video game display, controlling a piece of medical equipment such as a wheelchair, and controlling computer functions to effectuate a handless mouse, keyboard, joystick, etc.

Turning now to the figures, FIG. **1A** is a block level diagram which illustrates a system **10** for detecting initiating actions and providing an output which corresponds to the detected action. The system **10** includes a sensor **12** coupled to processing circuitry **14**. The sensor **12** includes an internal ear microphone as described in more detail below. The sensor **12** detects changes within the ear that result from an initiating action by the person wearing the sensor **12**. The processing circuitry **14** includes a data acquisition circuit **16**, such as a computer sound card. Other example data acquisition circuits include a PCI9118HG data acquisition card manufactured by Adlink Technology or a DAQi250 data acquisition card manufactured by Ines Company Gmbh in Germany. The data acquisition circuit can include an analog to digital (A/D) converter for converting an analog signal into digital signal data. A buffer **17** can be provided to store portions of signals from data acquisition circuit **16** (as those signals may be relatively continuous) until a filtration and signal detection can be carried out as are described further below. The processing circuitry **14** also includes a processor **18** for receiving the digital signal data from the data acquisition circuit **16** and buffer **17** and performing various signal processing functions on the digital signal data to detect the occurrence of an initiating action and determine the type of initiating action. The system **10** also includes an output peripheral **20** coupled to the processor **18** for executing one or more control instructions provided by the processor **18** which correspond to the initiating action. One skilled in the art will recognize that, depending on the application, the analog to digital converter function and the buffering function can be omitted or performed by software.

FIG. **1B** is a block level diagram which illustrates another embodiment of a system **22** for detecting initiating actions and providing an output which corresponds to the detected action. The system **22** includes the sensor **12** having an internal microphone (described in greater detail below). The electrical signal produced by the sensor **12** is optionally amplified with an amplifier **24**. However, in some applications, the amplifier **24** is not needed depending on the equipment receiving and processing the electrical signal output by the sensor **12**.

The electrical signal is then received by an interface adapter **26**. The interface adapter **26** can be a mechanical connection such as an electrical connector to be received by a corresponding jack on the device which ultimately receives and processes the electrical signal output by the sensor **12**. The interface adapter **26** can also include more complex instrumentation, such as a universal serial bus (USB) adapter. As one skilled in the art will appreciate, depending on the processor, device or application receiving the electrical signal produced by the sensor **12**, the interface adapter **26** is optional or can be any device for establishing physical and/or logical connection between the sensor **12** and the device to receive the electrical signal produced by the sensor **12**.

The electrical signal is then received by a computer interface **28**. The computer interface **28**, as with the amplifier **24** and the interface adapter **226**, is optional depending on the configuration of the system **22**. The computer interface **28** can be a mechanical connection such as a jack or receptacle, or can be more complex instrumentation, such as a sound board for a computer. In one embodiment, the amplifier **24** and/or the interface adapter **26** are omitted and the signal produced by the sensor **12** is input to processing circuitry **14'** via a conventional computer sound board, the sound board being used to convert the analog signal produced by the sensor **12** into a digital signal.

The electrical signal is then processed by processing circuitry **14'**. The processing circuitry **14'** can take on the form of the processing circuitry **14** illustrated in FIG. **1**A and can optionally include such devices such as an analog to digital converter, a buffer, and/or a processor **18'**. The processor **18'** can be configured to execute logic for a variety of applications that make use of the signal generated by the sensor **12**.

In the illustrated embodiment, the processor **18'** is programmed to execute a voice recognition software program. As is known in the art, voice recognition software converts spoken language into text which can be electronically manipulated such as with a word processing program. The text can be output to an output peripheral **20**. For example, the text can be displayed on a display or printed by a printer. The text can also be stored for later retrieval by a storage device **30** on a computer-readable medium including, but not limited to, magnetic and optical storage devices. Alternatively, the electrical signal itself or recognized speech derived therefrom can be used to control machinery such as a self-powered wheelchair, a robotic arm, a computer, transportation vehicles, domestic and industrial appliances or machinery, and the like.

FIG. **1**C is a block level diagram which illustrates a system **32** for detecting initiating actions and providing an output which corresponds to the detected action. The system **32** includes the sensor **12** having at least an internal microphone **34**. The sensor **12** and internal microphone **24** will be described in more detail below. The sensor **12** outputs an electrical signal which is optionally amplified by amplifier **24**. Next, the electrical signal is received by a communication device **36** via an appropriate interface adapter **26**, if needed. The communications device can be, for example, a telephone which is either hard-wired to a telephone system or, as shown in the illustrated embodiment, is a wireless telephone such as a cellular phone. Alternatively, the communications device **36** can be a radio, portable computing device, or the like. In one embodiment, the signal generated by the sensor **12** is input into a personal computer, personal digital assistant (PDA) or the like and is transmitted to another location or device via a network (e.g., a LAN, a WAN, the Internet, etc.). In another arrangements, the communications device can be replaced with other types of devices such as a recorder for recording the user's speech for later retrieval or an amplification system for broadcasting the speech to a live audience.

In addition, the sensor **12** can include a speaker **38** used to transmit (i.e., broadcast) sounds to the user. In this way, the sensor **12** and communication device **34** can be used as a bidirectional communication apparatus.

FIG. **1**D is a block level diagram which illustrates a system **40** for detecting initiating actions and providing an output which corresponds to the detected action. The system **40** includes the sensor **12**, which preferably includes the internal microphone **34** (FIG. **2**B) described in more detail below. The sensor **12** is connected to a medical diagnostics device **42** via, if desired, an amplifier **24**, interface adapter **26** and computer interface **28**. The medical diagnostics device **42** has an audio output **44** (e.g., speaker, stethoscope attachment, etc.) so that detections of air pressure by the sensor **12** corresponding to biological processes (heart beat, breath sounds, gastronomic sounds, etc.) of a patient can be heard and studied by medical personnel. To help the medical personnel listen to the biological processes, the medical diagnostics device can optionally include filters (in hardware or software) to isolate these sounds before broadcasting the sounds to the medical personnel. In another embodiment, the medical diagnostics device **42** has a video output **46** (e.g., an LCD or CRT screen) to display a video representation of the detected air pressure changes. For instance, the sound generated by the heart within the patient's ear and detected by the sensor **12** can be converted into a graph (e.g., amplitude versus time) and displayed on the video output **46** in similar fashion to display of an electrocardiogram.

The medical diagnostics device **42** can also include a recording device **48** (e.g., a digital or analog data storage medium) to store detections of the patient's biological processes. The recorded sounds can be compared to subsequent detections. This comparison can be made by a person or by software programmed to isolate differences between the first and second detections.

The medical diagnostics device **42** can also include a processor **50** which executes programmed logic (e.g., software) to analyze the detected air pressure changes. The software can be programmed to search the signal produced by the sensor for patterns, irregularities (including those normally undetectable by a human), differences from a baseline detection and so forth.

In addition, data acquired by the medical diagnostics device **42** from the sensor **12**, or data calculated by the medical diagnostics device **42**, can be transmitted to other locations. For example, the medical diagnostics device **42** can be placed in an ambulance and transmit information about a patient to a destination hospital. As an alternative example, the data could be gathered by one medical worker and transmitted to another medical worker, such as a specialist located in another city.

In another embodiment, the sensor can be located remotely from the medical diagnostic device **42**. For example, a person could use the sensor **12** from a location such as his/her home. The information detected by the sensor **12** could then be transmitted to a remote medical diagnostics device **42**. The transmission could be carried out over a telephone or other communications device as discussed above with respect to the system **32** illustrated in FIG. **1**C. Alternatively, the signal generated by the sensor **12** can be input to a personal computer, a PDA or the like and

transmitted to the medical diagnostics device **42** via a network (e.g., a LAN, a WAN, the Internet, etc.).

FIG. **1**E is a block diagram which illustrates a system **52** for detecting initiating actions and providing an output which corresponds to the detected action. The system **52** includes the sensor **12**, preferably having the internal sensor **34**. In the illustrated embodiment of the sensor **12**, the sensor **13** has an additional pressure detector, or external microphone **54**. The use of the external microphone **54** is optional. As will be described in more detail with respect to FIG. **2**B below, the internal microphone **34** is disposed on a housing **150** (FIG. **2**B) to be inserted at least part way into the ear of a person. That is, the internal microphone **34** is insert at least into a cavity defined by an external portion of the ear (described below in more detail with respect to FIG. **2**A). As a result, the internal microphone **34** detects air pressure changes located within the ear. These air pressure changes are the result of initiating actions and, in part, to externally generated sound, such as other people's speech, environmental noise and the like. The external microphone **54** is disposed on the housing **150** (FIG. **2**B) and generally points away from the person wearing the sensor **12**. As a result, the external microphone **54** detects air pressure changes located outside the ear. These air pressure changes are the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to initiating actions of the person wearing the sensor **12** and particularly the speech of that person.

The internal microphone **34** is connected to processing circuitry **14**" or another device used to receive the electrical signals respectively produced by the microphones **34** and **54** in response to sensed air pressure changes. Each microphone **34** and **54** is connected to the processing circuitry **14**" using separate and distinct inputs, or channels. In one embodiment, the internal microphone **34** is connected to the left channel input of a computer sound card, or computer interface **28***a*, via an amplifier **24***a* and an interface adapter **26***a*. The external microphone **54** is connected to the right channel input of the computer sound card, or computer interface **28***b*, via an amplifier **24***b* and an interface adapter **26***b*.

The processing of the unique signals generated by the microphones **34** and **54** will be described in more detail below with respect to FIG. **3**B. The unique signals can be used to distinguish between user produced initiating actions and externally produced sounds. In alternative configurations of the system **52**, the system **52** can be adapted to cancel externally generated noise from the detections of the internal microphone **34**.

Testing has shown that when the user speaks (or produces an air pressure change in the ear from another type of initiating action), the strength of the signal generated by the internal microphone **34** corresponding to the speech (or initiating action) is usually stronger than the strength of the signal generated by the external microphone **54** that corresponds to the same speech (or initiating action). Testing has also shown that when a sound is generated by a source other than the user (e.g., another person speaking), the strength of the signal generated by the external microphone **54** corresponding to the sound is usually stronger than the strength of the signal generated by the internal microphone **34** that corresponds to the sound. These sound energy differentials, or relative signal strengths, can be used to determine whether the source of the detected air pressure change is from the user or external to the user.

For instance, if the user is speaking, the internal microphone **34** may detect the user's speech and external noise (e.g., the hum of an air conditioner and a computer). In this example, the external microphone **54** may detect the user's speech, but to a smaller degree than the internal microphone **34**. The external microphone **54** will also detect external noise and that external noise will be detected to a larger degree by the external microphone **54** than by the internal microphone **34**. Accordingly, the processing circuitry **14**" can be programmed to compare the signals and discern that the source of the speech is from the user. In addition, the processing circuitry **14**" can be programmed to isolate the user's speech by comparing the two signals to identify the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be used in connection with voice recognition software as indicated above. In alternative embodiments, the foregoing technique can be used to distinguish between the user's speech and another individual's speech. Both person's speech can then be analyzed as desired. As one skilled in the art will appreciate, the foregoing technique can be applied to the detection of other type of initiating actions, including thoughts, movements related to the oral cavity, biological processes and the like.

Referring to FIGS. **2**A and **2**B, an external view and a cross-sectional view of an ear **100** are respectively illustrated. FIG. **2**B also shows the sensor **12** in cross-section. According to Henry Gray's famous text "Anatomy", the human ear is divided into three parts, including the external ear **102**, the middle ear (or tympanum) **104** and the internal ear (or labyrinth) **106**. The middle ear **104** and the internal ear **106** will not be described in great detail herein. The external ear **102** includes an expanded portion, or a pinna **108** (also referred to as an auricle), and an ear canal **110** (also referred to as a meatus or auditory canal). The pinna **108** serves to collect vibrations of the air surrounding the person's head. The ear canal **110** conducts those vibrations to the tympanum, or ear drum **112**.

The pinna **108** has a generally ovoid form with a larger end directed upward and having an outer surface that is irregularly concave and directed slightly forward. The pinna **108** has a number of eminences and depressions. Typically, the ear **100** has a prominent and curved rim, or helix **114**. Generally parallel to the helix **114** is another curved prominence, or antihelix **116**. The antihelix **116** bifrucates to form a triangular depression, or a fossa of the antihelix **118** (also referred to as a fossa triangularis). A narrow, curved depression located between the helix **114** and antihelix **116** is referred to as fossa of the helix, or scapha **120**. The antihelix **116** also curves around a deep, capacious cavity, or the concha **122** (the concha **122** being divided by the commencement of the helix **114**, or crus helicis, into an upper part, termed the cymba conchae, and a lower part, termed the cavum conchae). The concha **122** leads inward to an opening of the ear canal **110**. In front of the concha **122** and projecting backward (usually over the opening of the ear canal **110**) is a pointed eminence, or tragus **124**. Opposite the tragus **124** is a tubercle, or antitragus **126**. A notch-like depression, or incisura intertragica **128**, is disposed between the tragus **124** and antitragus **126**. A lobule **130** is present under the tragus **124** and antitragus **126**.

The ear canal **110** is an oval cylindrical passage extending from a bottom of the concha **122** to the ear drum **112**. The ear canal **110** is about an inch and a half in length when measured from the tragus **124** to the ear drum **112**. When measured from the bottom of the concha **122** to the ear drum **112**, the ear canal is about an inch long. The ear canal **110** forms a gradual "S-shaped" curve and is directed, at first, inward, forward and slightly upward (i.e., pars externa). The

ear canal **110** then passes inward and backward (i.e., pars media) and then passes inward, forward and slightly downward (i.e., pars interna).

It is not certain what physical, chemical or neural mechanism causes or generates the changes in air pressure in or near the ear in response to various initiating actions. However, due to the connection of the oral cavity to the ear via the eustachian tube, movements and speech may cause an air pressure, a change in air pressure or an air flow to or from the ear leading to a detectable air pressure which can be detected by the sensor **12**. Regardless of the exact physical, chemical or neural mechanism, empirical testing has confirmed that initiating actions described herein generate pressure changes in or near the ear of the person and that the air pressure changes have substantially their own signature and are thus substantially unique for a given initiating action of an individual. Consequently, the air pressure changes can be monitored near the ear and used to detect the initiating action of a user.

The present invention uses various forms of the terms "changes in air pressure" and "changes within the ear" in their broadest sense to characterize the parameter being measured. Changes in air pressure may alternatively be characterized as sound waves. These sound waves (or vibrations) may propagate through mediums other than air, such as bone and tissue. As is well known by those skilled in the art, as a sound wave spreads out from its source its intensity falls off (the energy per unit area decreases with the inverse square of the distance), but the total energy is constant. Consequently, it is desirable to have the microphone **34** sufficiently close to the source of the sound wave so that the intensity level of the detected changes will be as large as possible.

The frequency range at which sound waves are audible by humans is about 20 Hz to about 20 KHz, however, the present invention is not concerned with whether the changes within the ear are audible since the microphone **34** is selected to be sufficiently sensitive and to have a frequency detection range which is sufficient to detect air pressure changes at high or low frequencies. Therefore, any frequency range may be monitored which is suitable to detect initiating action (e.g., about 10 Hz to about 20 KHz) and such variations are contemplated as failing within the scope of the present invention.

It has been found that many initiating actions (including, for example, thoughts and tongue movements) generate air pressure changes having a frequency in the range of about 10 Hz to about 300 Hz. Other initiating actions, such as speaking and biological processes, generate air pressure changes in the range of about 20 Hz to about 20 KHz. Also, it is noted here again that the reference to detecting air pressure may mean detecting air pressure, a difference in air pressure and/or a sound. In addition, some initiating actions, such as movements of the tongue, can produce a detectable pressure wave that has a strength corresponding to the direction, speed and/or intensity of the movement. Although the invention may use a single sensor **12** in one ear **100**, two sensors **12** (one in each ear **100**) can be used to obtain additional information whereby each sensor **12** detects different information and/or cumulative information, which can be correlated by the processor.

FIG. **26** illustrates the sensor **12** inserted at least partially into the ear **100** of a person (i.e., at least within the cavity defined by the pinna **108**, if not deeper within the ear **1000** such as within the concha **122**, at the opening of the ear canal **110** or slightly into the ear canal **110**). The sensor **12** includes a housing **150**, the internal microphone **34** and, optionally, an external microphone **54**. In the illustrated embodiment the housing **150** has an exterior portion **152** and an interior portion **154**. The housing **150**, however, can take on a number of different configurations. The housing preferably resembles the housing of a hearing aid and particularly a digital hearing aid for similar insertion, or partial insertion, into the ear **100** and/or ear canal **110**. The exterior portion **152** is optional. The illustrated external portion **152** is made from a material such as plastic and is wider than an opening of the ear canal **110** to engage the pinna **108** or, alternatively, to engage the ear **100** to cover the whole pinna **108** (e.g., similar to that of a common headphone assembly). In one embodiment, the external portion **152** fits within the concha **122** and is retained, at least in part, by the tragus **124** and/or the antitragus **126**. Such arrangement at least partially insulates the interior microphone **34** from externally generated noise and air pressure changes.

The interior portion **154** can be made from a pliable material, such as rubber, and is sized and shaped to snugly fit within the ear **100**. Such material is sometimes referred to in the art as being "soft-solid." Optionally, the entire housing **150**, or portions thereof, can be made of pliable material, sound absorbing (or sound proofing) material and/or include sound insulating material such as foam.

The internal microphone **34** is disposed on the interior portion **154** of the housing **150**. Accordingly, the interior portion **154** is sized and shaped to position the internal microphone **34** with respect tot he ear **100** and/or ear canal **100** as desired. On one embodiment, the interior portion **154** positions the internal microphone **34** in the concha **122**. In another embodiment, the interior portion **154** positions the internal microphone **34** at the opening of the ear canal **110** where the ear canal **110** meets the concha **122**. It should be appreciated that the interior portion **154** and/or internal microphone **34** need not enter the ear canal **110**. In yet another embodiment, and as illustrated by example in FIG. **2**B, the interior portion **154** positions the internal microphone **154** within the ear canal **110**. When positioned within the ear canal **110**, the internal microphone **34** enters the ear canal **110** at a depth of about one tenth of an inch to about a half of an inch measured from the opening of the ear canal **110**.

The housing **150**, including both the exterior portion **152** and the interior portion **154**, can be custom designed for the individual to form a close and comfortable fit with the ear of the individual. Alternatively, the housing can have a standard design for all individuals which is fabricated in a number of sizes. As one skilled in the art will appreciate, many alternative configurations for the housing **150** are possible and each are considered to fall within the scope of the present invention.

The interior portion **154** of the housing **150** has a recess **156** in which the microphone **34** is placed. Preferably, the air adjacent the ear drum **112** and/or the air in the ear canal **110** will be in fluid communication with the air adjacent the microphone **34**. In this way, sounds or air pressure changes generated within the ear **100** will be transmitted directly to the microphone **34** through an air medium. In one embodiment, the microphone **34** is held in place with an adhesive. The microphone **34** in one embodiment is about one mm to about six mm in diameter. One skilled in the art, however, will appreciate that the microphone **34** is not necessarily round. In an alternative embodiment, the microphone **34** is disposed within a hollow cavity defined by the housing **150** and an opening in a wall of the internal portion **154** of the housing **150** is provided to establish fluid com-

munication of the air inside the housing **150** and the air inside the ear canal **110**.

By inserting the microphone **34** in the ear **100** and/or ear canal **110**, the microphone **34** is shielded from environmental noise. More specifically, the housing **150** and the head of the user at least partially blocks externally generated sound waves before reaching the microphone **34**.

In one embodiment, the microphone **34** is a directional microphone adapted to detect air pressure changes from a particular direction. The directional microphone is aimed as desired to detect various different air pressure changes generated by different type of initiating actions. For example, the microphone can be aimed towards a desired source, such as the ear drum **112**, the ear canal **110** wall, or the bone structure surrounding the opening of the ear canal **110**. Since the microphone is directional, the microphone will detect less environmental noise (the source of that noise is outside the ear and from a direction which does not typically coincide with the directional sensitivity of the microphone). In other embodiments of the present invention, the microphone **34** is a noise canceling microphone to further reduce the detection of environmental noise.

It should be appreciated that the microphone **34** can be moved closer to or further away from various anatomical structures within the ear **100** as desired for factors such as comfort, detection of certain air pressure changes caused by a specific type of initiating action, and the like.

The microphones **34** and **54** can take on a number of different forms, such as a silicon or semiconductor microphone (also referred to as a pressure sensor) such as the devices disclosed in U.S. Pat. Nos. 4,533,795, 4,922,471, and 6,156,585, each of which are incorporated herein by reference in their entireties. It is noted that the foregoing devices are exemplary and other electroacoustic transducers from a number of different vendors can be used.

In the illustrated embodiment, when the sensor **12** is inserted into the ear **100**, a slight positive pressure is created by compression of the air within the ear canal **110**. It is believed that this slight positive pressure, which is undetectable to the individual, improves the overall operating characteristics and sensitivity of the sensor **12**.

The microphone **34** is capable of detecting various forms of initiating action, including, for example, the speech of a person speaking at a variety of volumes, physical movements of the person such as clicking the person's tongue against the roof of the person's mouth, opening the person's jaw, the person's breathing, the person's heartbeat, etc. In addition, pressure changes in the ear caused by thoughts can be discerned by detecting the changes within the ear when a trained individual thinks a thought.

It has been shown that the use of one sensor **12** with only the internal microphone **34** (i.e., without an external microphone **54**) in one of the ears **100** of an individual can be used to detect the initiating actions described herein. However, empirical testing has shown that by using two such sensors **12**, one in each of the ears of an individual, the performance of systems such as the systems **10**, **22**, **32**, **40** and **52** improves by about six times due to the detection of more information. It is noted that in certain circumstances, the use of a sensor **12** in both ears is discouraged (such as when using a cellular telephone while driving a car). It is noted that the processing and logic routines described herein can be modified to process signals from a second sensor **12** (the sensors **12** with or without an external microphone **54**). Such modifications will be apparent to one of ordinary skill in the art.

Although not illustrated in FIG. **2**B, a speaker **38** (FIG. **1**C) can be placed on or within the housing **150** of the sensor **12** to transmit sounds to the individual. For example, when using the sensor **150** as a microphone for a communications device **36**, the sensor **12** can also be used to broadcast received sounds to the user.

As indicated, the sensor **12** can also include the external microphone **54**. The external microphone **54** is disposed on the exterior portion **152** of the housing **150** and is pointed away from the wearer of the sensor **12** assembly. The external microphone **54** can be secured to the housing **150** with adhesive and can be disposed in a recesses defined by the exterior portion **152**. Similar to the internal microphone **34**, the external microphone **54** is small (e.g., about one mm to about six mm, but need not be round).

The position of the external microphone **54** allows the external microphone **54** to detect air pressure changes present near the wearer's ear but have a source external to the wearer of the sensor **12**. In one embodiment, the external microphone **54** can optionally be a noise canceling microphone as is known in the art.

The internal microphone **34** is electrically coupled to the amplifier **24** with a conductor **158** and the exterior microphone **54** is electrically coupled to the amplifier **24** with a conductor **160**. Although illustrated as a single unit in FIG. **2**B, the amplifier **24** contains separate amplifying circuitry **24***a* and **24***b* (FIG. **1**E) for each signal respectively generated by the internal microphone **34** and the external microphone **54**. The amplifier **24** is electrically coupled to the interface adapter **26** (not shown in FIG. **2**B) with a conductor **162**. More specifically, the conductor **162** carries the amplified signals from the microphones **34** and **54** separately to separate interface adapters **26***a* and **26***b*. As one skilled in the art will appreciate, if the amplifier **24** is omitted the microphones **34** and **54** will be coupled directly to the interface adapter(s) **26** by way of a conductor or conductors. It is also noted that if two sensors **12** are used by placing one sensor **12** in each ear **100** of a person, the processing circuitry or other device receiving signals from the sensors **12** will receive four signals corresponding respectively to the internal microphone **34** and the external microphone **54** of each sensor **12**.

The microphones **34** and **54** of the sensor **12** monitor the changes in air pressure and convert the pressure changes to an analog electrical signal **170**, as illustrated in FIG. **2**C. In the signal **170** there are at least two signal components, a high frequency component **172** and a low frequency component **174**. In addition, other frequencies may also exist within the electrical signal **170** and the present invention can be adapted to analyze the various signal frequencies, as described herein.

In an alternative embodiment, the internal microphone **34** and the external microphone **54** are mounted to a headphone style assembly having a band disposed around the user's head and hubs disposed over one or both ears **100** of the user. One exemplary headphone and internal microphone assembly which may be used is the Model 4190 microphone manufactured by Brüel & Kjaer in Denmark.

A method **200** for carrying out the present invention is disclosed in FIG. **3**A. The method **200** includes detecting an initiating action by monitoring a change in the ear **100** at step **202** with the internal microphone **34** of the sensor **12**. The external microphone **54** is optional in carrying out the method **200**. Once the initiating action is detected at step **202**, one or more control instructions or other output(s) which correspond to the detected initiating action is provided at step **204** to effectuate the desired output function.

A method **200'** for carrying out the present invention is disclosed in FIG. **3**B. The method **200'** includes the detection of changes in air pressure with both the internal microphone **34** and the external microphone **54**. More specifically, in step **202'**, changes in air pressure within the ear **100** and/or ear canal **110** are detected by the internal microphone **34**. These detections can be the result of an initiating action, but can also include a sound component generated external to the person wearing the sensor **12** and/or a sound component generated by speech of the wearer and traveling from the wearer's mouth, through the air and to the wearer's ear **100** area. Also in step **202'**, changes in air pressure near, but outside, the ear **100** of the person wearing the sensor **12** are detected by the external microphone **54**. These detections are mainly the result of sounds generated external to the person wearing the sensor **12** and/or a sound component generated by speech of the wearer and traveling from the wearer's mouth, through the air and to the wearer's ear **100** area. The detections are converted by the microphones **34** and **54** into respective electrical signals, herein referred to as an internal signal and an external signal.

Next, in step **206**, the internal signal and the external signal are compared. When the user speaks or produces an air pressure change in the ear **100** from another type of initiating action, the strength of the signal generated by the internal microphone **34** corresponding to the initiating action is usually stronger than the strength of the corresponding signal generated by the external microphone **54**. Testing has also shown that when a sound is generated by a source other than the user (e.g., another person speaking), the strength of the signal generated by the external microphone **54** corresponding to the sound is usually stronger than the strength of the corresponding signal generated by the internal microphone **34**.

Using the signal strength differential, the source of the detected air pressure changes (i.e., from the user or external to the user) can be reliably determined. From further comparison of the internal signal and the external signal, two components from each signal are derived. The two components for each signal include a component relating to air pressure changes caused by the initiating action and a component relating to externally generated noise and/or sound, which may or may not include speech of the user detected by the external microphone **54** depending on the system in which the sensor **12** is being used. It is noted that for initiating actions other than speech, the initiating action component of the external signal may be nearly undetectable.

Subsequent processing of the internal signal and external signal will depend on the system in which the sensor **12** is being used and which types of initiating actions are being detecting. For example, the signals can be used by the processor or other device receiving the signals to invoke a corresponding output signal or signals. However, in most applications, it will be advantageous to isolate the component of the internal signal generated by the initiating action and, if desired, eliminate to the fullest extent possible any externally generated component of the internal signal. Therefore, in step **208**, the method **200'** distinguishes the initiating action component of the internal microphone signal. In another embodiment of step **208**, the method **200'** can cancel noise (or all sounds) detected by the external microphone **54** from the detections of the internal microphone **34**. More specifically, the externally generated component of internal signal is filtered from or mathematically removed from the internal signal. The resulting signal from step **208** is used in step **204'** to provide the control instruction or output response appropriate for the initiating action.

Similar processing to process the external signal can be carried out in step **210** to derive a signal of externally generated sounds, such as a to isolate another person's speech from noise and from speech of a person wearing the sensor **12**. Thereafter, in step **212**, a control instruction or output response appropriate for the externally generated sound can be produced.

A method of detecting changes in the ear **100** (step **202**) is illustrated in FIG. **4**. The microphone **34** is placed in the ear **100** of the user who's initiating actions are to be detected at step **214**. However, initiating actions may result in changes at or near other parts of the body. Therefore, it is contemplated that in alternative embodiments of the present invention, the sensor **12** may be located on, in or near other parts of the body and any detection of initiating actions by analyzing physiological or air pressure changes is contemplated as falling within the scope of the present invention.

The changes in the ear **100** are monitored with the microphone **34** at step **216** and are converted to an electrical signal at step **218** for subsequent analysis. After conversion into an electrical signal at step **218**, the electrical signal is analyzed to detect the initiating action at step **220**. Although it is conceivable that the initiating action may be detected at step **220** simply by analyzing the signal corresponding to changes in air pressure or changes in the ear **100** without additional data processing, step **220** typically includes data processing in conjunction with the signal analysis. It is noted that when a sensor is used which has an external microphone **54**, then the foregoing logic can be modified to include the capture of an electrical signal corresponding to air pressure changes outside the ear **100** and using that signal in the analysis of step **220** as described above.

An exemplary method of analyzing and processing the electrical signal which corresponds to the monitored pressure (step **220**) is illustrated in FIG. **5**. Again, the analyzing logic can be adapted as described herein to include processing of a second electrical signal derived from an external microphone **54**. The analog electrical signal, an example of which is illustrated in FIG. **2**C, is converted into a digital signal at step **222**, as illustrated by example in FIG. **6**. As is well known by those skilled in the art, an analog signal may be converted into a digital signal by sampling the analog signal at a selected frequency and identifying the signal amplitude at each sampling point. Each sampled data point is then saved as a digital word in a memory and used for further analysis. In FIG. **6**, a sampled analog signal is illustrated in which the dotted line illustrates the exemplary analog signal for a particular time period and the plurality of points on the dotted line represent sampled amplitude values which are saved in the memory. It is desirable that the sampling frequency be sufficient to capture enough data points to adequately represent the analog signal. As an example, the sampling rate of the may be 32 KHz and the total signal time length to be analyzed may be 2048 mSec. Alternatively, however, other sampling rates and data acquisition time frames may be utilized and such variations are contemplated as falling within the scope of the present invention.

Once the analog signal has been converted into digital signal data at step **222**, the digital data is analyzed and processed by, for example, a signal processor to detect the presence of an initiating action at step **224**. As indicated above, step **224** may include comparison of the internal and external signals, and filtering or mathematical manipulation of the signals based on the comparison to isolate a desired component(s) of the detections and/or to remove an undesired component(s) of the detections. In addition, or as an

alternative, the analysis and processing of the data may be performed in a plurality of segments, for example, as illustrated in FIGS. **7** and **8**. As illustrated in FIG. **7**, the processing of step **224** can include analysis of a first data segment in step **226**, followed by the analysis of a second data segment at step **228**. Once various data segments have been analyzed separately, the data segments are analyzed together at step **230**. If all the data segments have not yet been analyzed at step **232**, the method returns to step **228** and the next data segment is analyzed, after which all previous segments are then analyzed together at step **230**. The process continues until all the data segments have been analyzed at step **232**, thus allowing a conclusion to be generated using the analyzed data segments at step **234**.

The data segment analysis may be seen graphically in FIG. **8**, wherein digital signal data **236** is illustrated as being continuous for the sake of simplicity. The total length of data for analysis may be separated into **64** segments that are each 32 mSec in length. Note that the signal **236** contains both a high frequency component **238** and a low frequency component **240**. Since data relating to an initiating action potentially may be found in either component or the initiating action data may span multiple data segments, the data segments may be analyzed separately as well as together. Thus, at step **226** of FIG. **7**, the first data segment is analyzed (region **242**), at step **228** the second data segment is then analyzed (region **244**) and at step **230** both data segments are analyzed together (region **246**). The process then continues for all the data segments; consequently, the data analysis of the present invention may analyze both the high frequency and low frequency signals to detect the initiating action.

Returning back to FIG. **5**, once the data considered to be initiatory data has been found in the data signal from the sensor at step **224**, subsequent analysis is performed to determine the type of initiating action at step **246**. Such analysis can include feature extraction and analysis using a neural net trained to recognize respective signals/features of such signals. If the neural net determines that the signal represents an initiating action of a particular character, then it provides an output or effects the production of an output representative thereof for use as described herein. Alternatively, however, other techniques may be utilized to identify the initiating action type once the initiating action is detected and any such technique is contemplated as falling within the scope of the present invention.

One exemplary method of analyzing the digital signal in data segments (step **224**) is illustrated in FIG. **9**. For each data segment of 32 mSec, the data is converted from the time domain to the frequency domain at step **250** using, for example, a Fast Fourier Transform (FFT) as is well known by those skilled in the art. As is well known, a time domain signal f(t) is linked with the frequency domain f(jω) according to the following equation:

$$F(f(t))=\int f(t)e^{-j\omega t}dt=f(j\omega),$$

wherein F(f(t)) is a traditional Fourier transform. As is well known by those skilled in the art, a Fast Fourier Transform is related to the traditional Fourier transform since the Fast Fourier Transform is an efficient algorithm for computing discrete Fourier transforms. After the digital signal data is converted into the frequency domain via the Fast Fourier Transform, the frequency domain data is processed to distinguish data relating to initiating actions from noise data at step **252**. As is well known by those skilled in the art, the separation of data from noise is often simplified in the frequency domain because unlike noise, the data signal has some physical characteristics. Though the data signal in the time domain has an amplitude which is less than the noise, the data signal has a greater amplitude than the noise in the frequency domain. Therefore the Fast Fourier Transform is a typical method for noise separation. In addition, or as an alternative, signal data from the external microphone **54** can be used to derive signal components as described in greater detail above.

The details surrounding the data processing of the digital signal data may be accomplished through a variety of data processing techniques as is well known by those skilled in the art and any data processing methodology is contemplated as falling within the scope of the present invention. Although many different data processing methodologies may be employed, an example methodology is disclosed below in conjunction with the following method and system.

Turning to FIG. **10**, a system **270** and method are depicted. One or more sensors **12** sense changes in the ear(s) **100** of the user and, if desired, external to the ears(s) **100** of the user. The analog signal(s) from the sensor(s) is provided via a data acquisition card, analog to digital converter, and buffer **16**/**17** to the processor **18**. The signal received from the sensors is substantially continuous at least over respective time periods while the initiating action is being performed; and, therefore, the buffer **17** is able to store sufficient portions of the signal for analysis by the processor **18**.

In one embodiment, the processor **18** can include a band pass filter **272**, power spectrum filter **274**, signal detector **276**, signal comparator **278**, feature extraction portion **280**, and neural network **282**. The system **270** also includes an output **284** which may be coupled to a desired output peripheral **20**.

Summarizing operation of the overall system **270**, the sensor **12** and data acquisition, A/D, and buffer **16**/**17** supply a signal to the band pass filter **272**. An example of a raw data or raw signal supplied to the band pass filter **272** is illustrated at **286** in the graph of FIG. **11**. The raw signal is presented to the band pass filter and is operated on by the band pass filter in the frequency domain using fast Fourier transform techniques. The fast Fourier transform transforms the digital signal to the frequency domain. In the band pass filter frequencies that are above or below the frequency range representing initiating action, for example, are removed from the signal. The frequencies which are above and below the frequency range of interest typically are due to noise. An example of the signal after being filtered by band pass filtration is illustrated at **288** in the graph of FIG. **12**. Various band pass filters may be used to provide the band pass filtration to strip from the signal frequencies above and below the desired frequency range.

Portions of the signal represented at **288**, such as portions **288***a* and **288***b* may not be representative of initiating action, e.g., since they are relatively low power and, therefore, still may be noise. Therefore, the power spectrum filter **274** does a further filtration according to power of the various portions of the signal to determine portions which are due to initiating action and, accordingly, should be considered for further analysis. The power filtration may be carried out by slicing the signal of FIG. **12** to multiple windows or segments, such as those represented at **290***a*, **290***b*, . . . **290***i* in FIG. **13**. It will be appreciated that the graph of FIGS. **12** and **13** are of different scales. Thus, the windows **290***a*–**290***i* slice a time frame between times t_a and t_b, which is the same time frame between times ta and tb illustrated for the signal portion **288***c* in FIG. **12**.

Power filtration suppresses or eliminates those frequencies where the power of the signal does not exceed a

prescribed threshold or value and, thus, likely represent noise. By eliminating those noise components the signal representing the initiating action can be found. To perform the power filtration, for example, a power spectrum for the signal is calculated using the following formula:

$$P_f = \text{Mag}_f^2 + \text{Phase}_f^2$$

where f=$F_0 \ldots F_i$, and [F_0; F_i] represent the frequency range in which the signal resulting from tongue action is expected.

In words, the power is a function of the magnitude squared plus the phase squared. The power as calculated above for respective windows **290***a,* **290***b* . . . **290***i* is reconstructed to make a graph **292** shown in FIG. **14**. The graph **292** shows values **292***a,* **292***b* . . . **292***i* from the mentioned computation. The curve **292** is relatively smooth and it represents power spectrum where a signal caused by initiating action may be found in the graph **288** of FIG. **12**. Usually the signal caused by initiating action is of relatively high power and occurs in time approximately at the middle of the graph **292**, e.g., over the time frame T shown in FIG. **13**. It is the time at which this time frame occurs and the duration of it that detects the signal or locates when in time the signal **288** represents the initiating action. Therefore, the signal detector component **276** of the system **270** in effect is the combination of the graphs of FIGS. **12** and **13** and the analysis just described to provide the signal detection function. The time frame T also is shown in the graph of FIG. **12** and, thus designates that portion **288***d* (in the illustrated blocked window **288***e*) of the signal **288***c* which is to be considered produced by the initiating action.

As an alternative to, or in addition to, the filtration and detection operations of the power spectrum filter **274**, the signal detector **276** and the feature extraction **278**, the signal comparator **278** can carry out method **200'** to produce a signal indicative of the initiating action, or signal portion **288***d.* The signal portion **288***d* or at least a feature of that signal then may be supplied to the neural network **282**. Using feature extraction principles a feature of the signal portion **288***d* is supplied to the neural network. In the neural network **282** the feature is compared with other features on which the neural network has been trained. Training of neural networks is a well known procedure. The neural network can determine whether an input feature, such as that representative of the signal portion **288***d,* is an acceptable match or correlates acceptably to the training of the neural network; and, if affirmative, then the neural network causes production of the output **284**. The output **284** may be an electrical signal which is analogous to the type of signal produced by a computer mouse, joystick or other device. Thus, in one example, a tongue action whereby the tongue of the user is moved from the front bottom left of the mouth to the back top right of the mouth may cause the production of an output signal analogous to a movement of a mouse from the lower left to the upper right of a mouse pad.

The feature which is extracted from the signal **288**, for example, may be features which designate some characteristic or feature which differentiates the curve representing the signal from other things. Various features are used in feature extraction methods, which are well known. In the present invention an exemplary feature is the smoothed power spectrum of the signal portion **288***d.* Other features may be the logarithm of the power spectrum, the power spectrum itself, etc.

The neural network **282** may be a conventional neural network trained in a typical fashion. The neural network **282** can compare the feature of the signal portion **288***d* with its training and can determine whether the feature is more like one trained feature or another trained feature. If the input feature of signal portion **288***d* is sufficiently like one trained feature, then the neural network causes production of an appropriate signal at the output, as was described above, for example. If the input feature is not sufficiently like any of the trained features or is relatively similar to two or more trained features, then the neural network will not cause the output to produce an output signal, for the uncertainty may otherwise cause production of an undesirable output signal.

As was described above, if two sensors **12** are used, then the system **270** may operate sequentially on each signal from respective sensors **12** or may include parallel processing capability so that the signals from both sensors can be processed substantially simultaneously. The output **284** may reflect the results of signal analysis of the signals originating from both sensors and may be used to provide an output signal at the output **284** that reflects in effect three dimensional data.

The diagrams described herein show the architecture, functionality and operation of various systems for detecting changes within a human ear and producing an output signal or control instruction corresponding to an initiating action. Each functional block may represent a module, segment, or portion of software code that contains one or more executable instructions to implement the specified logical function (s).

Although the block diagrams and flowcharts show a specific order of execution, it is understood that the order of execution may differ from that depicted. For example, the order of execution of two or more blocks may be altered relative to the order shown. Also, two or more blocks shown in succession may be executed concurrently or with partial concurrence. In addition, various blocks may be omitted. It is understood that all such variations are within the scope of the present invention.

Although the invention has been shown and described with respect to a certain preferred embodiment or embodiments, it is obvious that equivalent alterations and modifications will occur to others skilled in the art upon the reading and understanding of this specification and the annexed drawings. In particular regard to the various functions performed by the above described components (assemblies, devices, circuits, etc.), the terms (including a reference to a "means") used to describe such components are intended to correspond, unless otherwise indicated, to any component which performs the specified function of the described component (i.e., that is functionally equivalent), even though not structurally equivalent to the disclosed structure which performs the function in the herein illustrated exemplary embodiments of the invention. In addition, while a particular feature of the invention may have been disclosed with respect to only one of several embodiments, such feature may be combined with one or more other features of the other embodiments as may be desired and advantageous for any given or particular application.

What is claimed is:

1. A method of detecting an air pressure change within an ear of a person, comprising the steps of:

placing a first microphone at least partially within an ear of the person;

detecting the change in air pressure within the ear of the person using the microphone, the change in air pressure including an aspect corresponding to an initiating action made by person;

producing an electrical signal corresponding to the internally detected change in air pressure;

detecting an air pressure change adjacent and external to the ear of the person with a second microphone;

producing an electrical signal corresponding to the externally detected change in air pressure;

comparing a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the elctrical signal corresponding to the externally detected change in air pressure to distinguish an initiating action component of each signal from an external source component of each signal; and

processing the electrical signals to produce and output signal corresponding to the initiating action.

2. The method according to claim **1**, wherein the first microphone is placed in a concha of the ear.

3. The method according to claim **1**, wherein the first microphone is placed in an ear canal of the ear.

4. The method according to claim **1**, wherein the initiating action is speech and the step of processing includes performing automated speech recognition to produce the output signal.

5. The method according to claim **1**, wherein the initiating action is speech and the method further includes the step of transmitting the output signal to a remote location over a communications medium.

6. The method according to claim **1**, wherein the initiating action is a thought and the output signal is a control instruction associated with the thought.

7. The method according to claim **6**, further comprising the step of training a neural network to recognize a signal pattern associated with the thought and contained within at least one of the electrical signals.

8. The method according to claim **1**, wherein the initiating action is a non-vocal voluntary physical movement associated with the oral cavity and the output signal is a control instruction associated with the movement.

9. The method according to claim **8**, further comprising the step of training a neural network to recognize a signal pattern associated with the movement and conained within at least one of the elctrical signals.

10. The method according to claim **1**, further comprising the step of removing at least an external source component of the electrical signal corresponding to the internally detected change in air pressure.

11. A detection system, comprising:

a housing positionable with respect to an ear of a person;

a first microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person makes an initiating action and to produce an electrical signal corresponding to the internally detected change in air pressure;

a second microphone disposed with respect ot the housing and located adjacent to the ear and operable to detect a chnage in air pressure external to the ear of the person and to produce an elctrical signal corresponding to the externally detected change in air pressure; and

processing circuitry that executes logic to compare a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the electrical signal corresponding to the externally detected change in air pressure to distinguish an initiating action component of each signal from an external source component of each signal and the processing circuitry coupled to the microphone for processing the processing the processes the elctrial signals to produce an output signal corresponding to the initiating action.

12. The system according to claim **11**, wherein the first microphone is disposed for placement in a concha of the ear.

13. The method according to claim **11**, wherein the first microphone is disposed for placement in an ear canal of the ear.

14. A speech recognition system according to claim **11**, wherein the initiating action is speech and the processing circuitry executes logic to perform automated speech recognition to produce the output signal.

15. A communication system according to claim **11**, wherein the initiating action is speech and the system includes a communications device for transmitting the output signal to a remote location.

16. The system according to claim **11**, wherein the initiating action is a thought and the processing circuitry executes logic to recognize the thought from the electrical signals and the output signal is a control instruction for controlling a computing device or item of machinery.

17. The system according to claim **16**, wherein the recogniton logic includes neural network matching logic.

18. The system according to claim **11**, wherein the initiating action is a non-vocal voluntary physical movement associated with the oral cavity and the processing circuitry executes logic to recognize the movement from the electrical signals and the output signal is a control instruction for controlling a computing device or item of machinery.

19. The system according to claim **18**, wherein the recognition logic includes neural network matching logic.

20. The system according to claim **11**, wherein the processing circuitry executes logic to remove at least an external source component of the electrical signal corresponding to the internally detected change in air pressure.

* * * * *

Exhibit PA-A

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,647,368 B2
DATED : November 11, 2003
INVENTOR(S) : Nemirovski

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 11,
Line 44, replace "failing" with -- falling --
Line 63, replace "FIG. 26" with -- FIG. 2B --

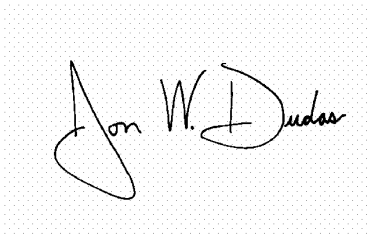
Column 20,
Line 64, replace "elctrical" with -- electrical --

Column 21,
Line 37, replace "conained" with -- contained --
Line 38, replace "elctrical" with -- electrical --

Column 22,
Line 1, replace "ot" with -- to --
Line 2, replace "adjacent to the ear" with -- adjacent the ear --
Line 3, replace "chnage" with -- change --
Line 4, replace "elctrical" with -- electrical --
Lines 14 - 16, replace "processing circuitry coupled to the microphone for processing the processing the processes the elctrial" with -- processing circuitry processes the electrical --

Signed and Sealed this

Twenty-sixth Day of October, 2004



JON W. DUDAS
Director of the United States Patent and Trademark Office

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 #
90/019,169	**02/24/2023 05:27:37 PM ET**	**0027071.00133**

Title of Invention

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

Application Information

APPLICATION TYPE	Utility / ex parte reexam	PATENT #	11244666
CONFIRMATION #	7859	FILED BY	Sandra Tartisel
PATENT CENTER #	61645467	FILING DATE	-
CUSTOMER #	150139	FIRST NAMED INVENTOR	Steven Wayne Goldstein
CORRESPONDENCE ADDRESS	-	AUTHORIZED BY	Ali Sharifahmadian

Documents

TOTAL DOCUMENTS: 43

DOCUMENT	PAGES	DESCRIPTION	SIZE (KB)
Exhibit_PA_A.pdf	22	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3273 KB
Exhibit_PA_B.pdf	16	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2767 KB
Exhibit_PA_C.pdf	15	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1398 KB
Exhibit_PA_D.pdf	10	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1838 KB

Exhibit_PA_E.pdf	21	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1536 KB
Exhibit_PA_F.pdf	15	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2476 KB
Exhibit_PA_G.pdf	33	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3049 KB
Exhibit_PA_H.pdf	11	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2107 KB
Exhibit_PA_I.pdf	17	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2476 KB
Exhibit_PA_J.pdf	12	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2342 KB
Exhibit_PA_K.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3053 KB
Exhibit_PA_L.pdf	28	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2868 KB
Exhibit_PA_M.pdf	10	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2011 KB
Exhibit_PA_N.pdf	12	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	981 KB

Exhibit_PA_O.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2516 KB
Exhibit_PA_P.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2467 KB
Exhibit_PA_Q.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	505 KB
Exhibit_PA_R.pdf	8	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1944 KB
Exhibit_PA_S.pdf	13	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1959 KB
Exhibit_PA_T.pdf	11	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2090 KB
Exhibit_PA_U.pdf	13	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2348 KB
Exhibit_PA_V.pdf	11	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2264 KB
Exhibit_PA_W.pdf	14	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	10976 KB
Exhibit_PAT_A.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2785 KB

Exhibit_PAT_B.pdf	281	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	12065 KB
Exhibit_PAT_C.pdf	196	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	8305 KB
Exhibit_PAT_D.pdf	254	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	12645 KB
Exhibit_PAT_E.pdf	70	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3214 KB
Exhibit_PAT_F.pdf	58	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2625 KB
Exhibit_PAT_G.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2642 KB
Exhibit_PAT_H.pdf	19	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2875 KB
Exhibit_PAT_I.pdf	19	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2864 KB
Exhibit_PAT_J.pdf	18	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2789 KB
Exhibit_PAT_K.pdf	39	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3060 KB

Exhibit_OTH_C.pdf		83	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	1027 KB
Exhibit_OTH_D.pdf		34	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	2587 KB
Exhibit_OTH_E.pdf		31	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	3481 KB
Exhibit_PA_1449.pdf		2	Reexam - Info Disclosure Statement Filed by 3rd Party	655 KB
Exhibit_OTH_A.pdf		220	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	5299 KB
Exhibit_OTH_B.pdf		5	Reexam - Affidavit/Declaration/Exhibit Filed by 3rd Party	673 KB
Transmittal_Request_EPR_USP11244666.pdf		4	Transmittal Letter	285 KB
Request_for_EPR_US11244666.pdf		148	-	1897 KB
Request_for_EPR_US11244666-RXOSUB.R.pdf	(1-147)	147	Receipt of Original Ex Parte Request by Third Party	1894 KB
Request_for_EPR_US11244666-RXC/SR.pdf	(148-148)	1	Reexam Certificate of Service	132 KB

Digest

DOCUMENT	MESSAGE DIGEST(SHA-512)

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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.

Exhibit PA-V


US009135797B2

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

(10) **Patent No.: US 9,135,797 B2**
(45) **Date of Patent: Sep. 15, 2015**

(54) **AUDIO DETECTION USING DISTRIBUTED MOBILE COMPUTING**

(75) Inventors: **Christopher C. Couper**, Shingle Springs, CA (US); **Neil A. Katz**, Parkland, FL (US); **Victor S. Moore**, Lake City, FL (US)

(73) Assignee: **INTERNATIONAL BUSINESS MACHINES CORPORATION**, Armonk, NY (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 1046 days.

(21) Appl. No.: **11/616,973**

(22) Filed: **Dec. 28, 2006**

(65) **Prior Publication Data**

US 2008/0162133 A1 Jul. 3, 2008

(51) **Int. Cl.**

G10L 15/00	(2013.01)
G10L 21/00	(2013.01)
H04R 5/00	(2006.01)
H04R 29/00	(2006.01)
G08B 13/16	(2006.01)
G10L 15/20	(2006.01)
G10L 21/0208	(2013.01)
G08B 25/08	(2006.01)
G08B 25/10	(2006.01)
H04R 5/027	(2006.01)
H04R 5/04	(2006.01)

(52) **U.S. Cl.**

CPC ***G08B 13/1672*** (2013.01); ***G08B 25/08*** (2013.01); ***G08B 25/10*** (2013.01); ***G10L 15/00*** (2013.01); ***G10L 15/20*** (2013.01); ***G10L 21/00*** (2013.01); ***G10L 21/0208*** (2013.01); *H04R 5/00* (2013.01); *H04R 5/027* (2013.01); *H04R 5/04* (2013.01)

(58) **Field of Classification Search**

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See application file for complete search history.

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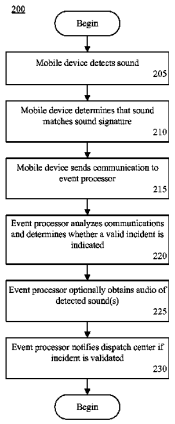
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Primary Examiner — Pierre-Louis Desir
Assistant Examiner — David Kovacek
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(57) **ABSTRACT**

A method of identifying incidents using mobile devices can include receiving a communication from each of a plurality of mobile devices. Each communication can specify information about a detected sound. Spatial and temporal information can be identified from each communication as well as an indication of a sound signature matching the detected sound. The communications can be compared with a policy specifying spatial and temporal requirements relating to the sound signature indicated by the communications. A notification can be selectively sent according to the comparison.

17 Claims, 2 Drawing Sheets



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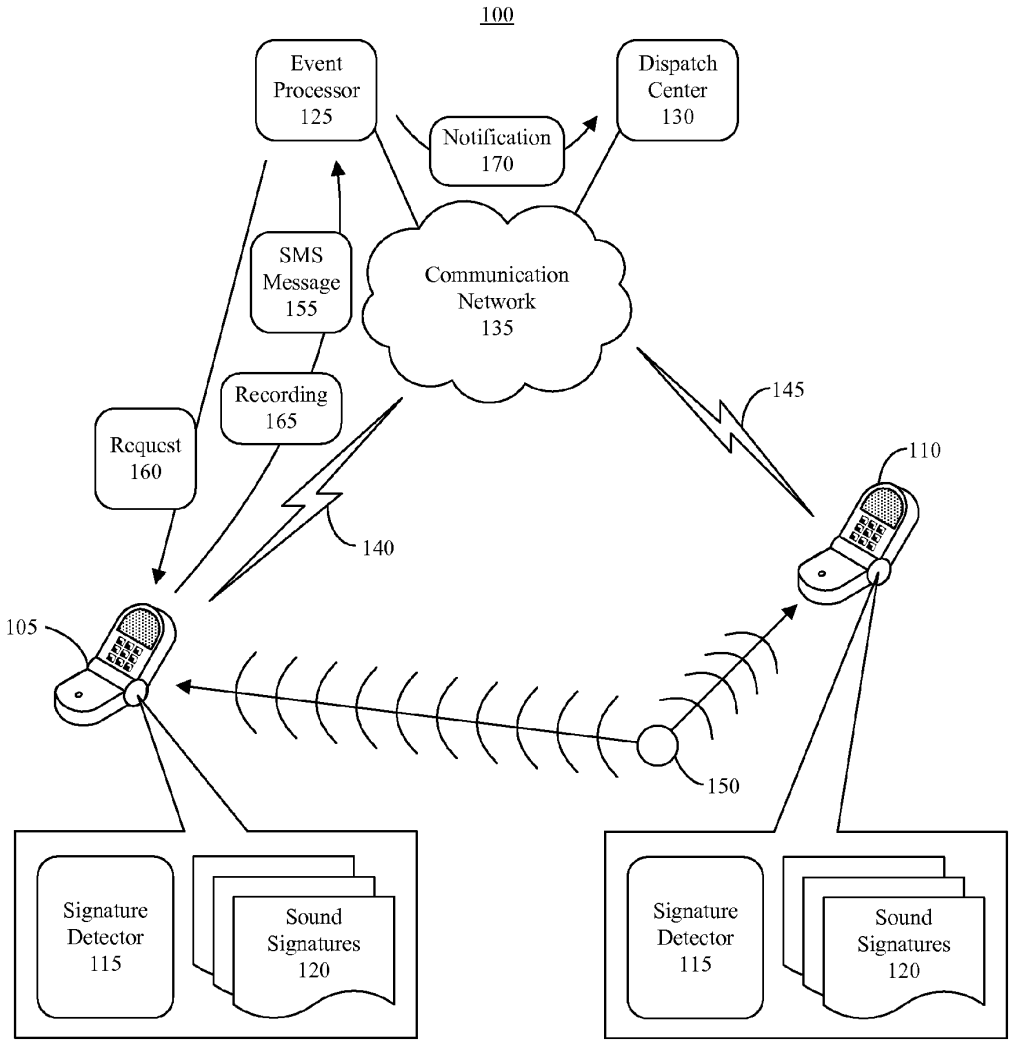

100
Event Processor 125
Dispatch Center 130
Notification 170
SMS Message 155
Communication Network 135
Recording 165
Request 160
145
110
140
105
150
Signature Detector 115
Sound Signatures 120
Signature Detector 115
Sound Signatures 120

FIG. 1

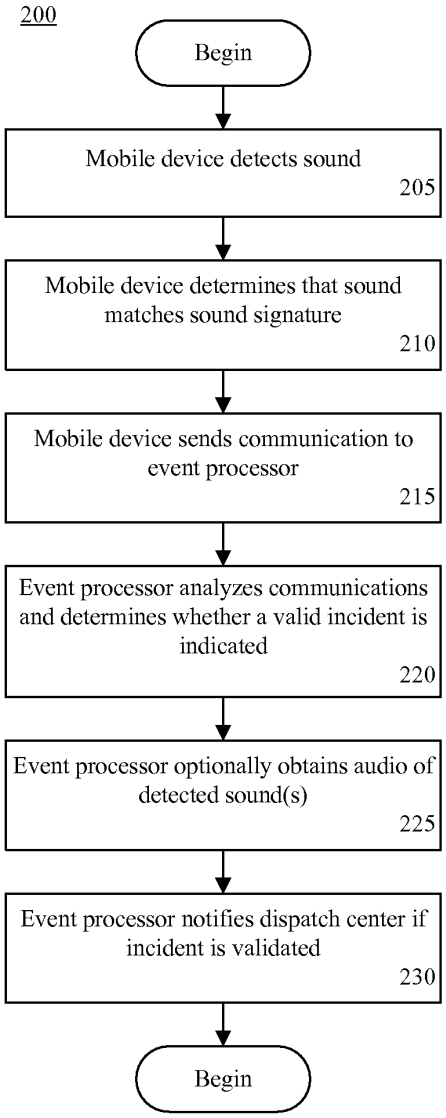

200
Begin
Mobile device detects sound
205
Mobile device determines that sound matches sound signature
210
Mobile device sends communication to event processor
215
Event processor analyzes communications and determines whether a valid incident is indicated
220
Event processor optionally obtains audio of detected sound(s)
225
Event processor notifies dispatch center if incident is validated
230
Begin

FIG. 2

AUDIO DETECTION USING DISTRIBUTED MOBILE COMPUTING

BACKGROUND OF THE INVENTION

Some municipalities have come to rely upon sound detection as a tool for crime prevention. Fixed-location audio sensors are distributed throughout a geographic area, such as a neighborhood, a town, or a city. The audio sensors are networked with a central processing system. The central processing system continually monitors the sounds provided by the various sensors to determine whether any detected sound is indicative of a potential crime.

Audio provided from the sensors to the central processing system is compared with signatures of various sounds. For example, audio from the sensors can be compared with signatures for gunshots, breaking glass, or the like. If a portion of audio matches the signature of one, or more, of the sounds, the central processing system can determine that the event, e.g., a gunshot, a window being broken, likely happened in the vicinity of the sensor that sent the audio.

When such a sound is detected, a response team can be dispatched to the location at which the sound was detected. While this sort of system has been successfully used to reduce crime, it can be costly to deploy. The system requires the installation of specialized audio sensors and networking equipment throughout a geographic area. The cost of installing the audio sensors alone can be significant even before the other components of the system are considered.

BRIEF SUMMARY OF THE INVENTION

The present invention relates to a method of identifying incidents using mobile devices. A communication from each of a plurality of mobile devices can be received. Each communication can specify information about a detected sound. Spatial and temporal information can be identified from each communication as well as an indication of a sound signature matching the detected sound. The spatial and temporal information from the communications can be compared with a policy specifying a spatial constraint and a temporal constraint, each of the constraints relating to the sound signature indicated by the communications and varying according to the sound signature. A notification can be selectively sent according to the comparison.

The present invention also relates to a method of identifying incidents using mobile devices including receiving a communication from each of a plurality of mobile devices. The communications can be compared with a validation policy that specifies a spatial constraint and a temporal constraint, each of which relates to a sound signature matching the detected sound and varies according to the sound signature. A notification can be selectively sent according to the comparison.

The present invention also relates to a computer program product including a computer-usable medium having computer-usable program code that, when executed by an information processing system, performs the various steps and/or functions disclosed herein.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWINGS

FIG. **1** is a block diagram illustrating a system in accordance with one aspect of the present invention.

FIG. **2** is a flow chart illustrating a method in accordance with another aspect of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

As will be appreciated by one skilled in the art, the present invention may be embodied as a method, system, or computer program product. Accordingly, the present invention may take the form of an entirely hardware embodiment, an entirely software embodiment, including firmware, resident software, micro-code, etc., or an embodiment combining software and hardware aspects that may all generally be referred to herein as a "circuit", "module", or "system".

Furthermore, the invention may take the form of a computer program product accessible from a computer-usable or computer-readable medium providing program code for use by, or in connection with, a computer or any instruction execution system. For the purposes of this description, a computer-usable or computer-readable medium can be any apparatus that can contain, store, communicate, propagate, or transport the program for use by, or in connection with, the instruction execution system, apparatus, or device.

Any suitable computer-usable or computer-readable medium may be utilized. For example, the medium can include, but is not limited to, an electronic, magnetic, optical, electromagnetic, infrared, or semiconductor system (or apparatus or device), or a propagation medium. A non-exhaustive list of exemplary computer-readable media can include an electrical connection having one or more wires, an optical fiber, magnetic storage devices such as magnetic tape, a removable computer diskette, a portable computer diskette, a hard disk, a rigid magnetic disk, an optical storage medium, such as an optical disk including a compact disk-read only memory (CD-ROM), a compact disk-read/write (CD-R/W), or a DVD, or a semiconductor or solid state memory including, but not limited to, a random access memory (RAM), a read-only memory (ROM), or an erasable programmable read-only memory (EPROM or Flash memory).

A computer-usable or computer-readable medium further can include a transmission media such as those supporting the Internet or an intranet. Further, the computer-usable medium may include a propagated data signal with the computer-usable program code embodied therewith, either in baseband or as part of a carrier wave. The computer-usable program code may be transmitted using any appropriate medium, including but not limited to the Internet, wireline, optical fiber, cable, RF, etc.

In another aspect, the computer-usable or computer-readable medium can be paper or another suitable medium upon which the program is printed, as the program can be electronically captured, via, for instance, optical scanning of the paper or other medium, then compiled, interpreted, or otherwise processed in a suitable manner, if necessary, and then stored in a computer memory.

Computer program code for carrying out operations of the present invention may be written in an object oriented programming language such as Java, Smalltalk, C++ or the like. However, the computer program code for carrying out operations of the present invention may also be written in conventional procedural programming languages, such as the "C" programming language or similar programming languages. The program code may execute entirely on the user's computer, partly on the user's computer, as a stand-alone software package, partly on the user's computer and partly on a remote computer or entirely on the remote computer or server. In the latter scenario, the remote computer may be connected to the user's computer through a local area network (LAN) or a

wide area network (WAN), or the connection may be made to an external computer (for example, through the Internet using an Internet Service Provider).

A data processing system suitable for storing and/or executing program code will include at least one processor coupled directly or indirectly to memory elements through a system bus. The memory elements can include local memory employed during actual execution of the program code, bulk storage, and cache memories which provide temporary storage of at least some program code in order to reduce the number of times code must be retrieved from bulk storage during execution.

Input/output or I/O devices (including but not limited to keyboards, displays, pointing devices, etc.) can be coupled to the system either directly or through intervening I/O controllers. Network adapters may also be coupled to the system to enable the data processing system to become coupled to other data processing systems or remote printers or storage devices through intervening private or public networks. Modems, cable modems, and Ethernet cards are just a few of the currently available types of network adapters.

The present invention is described below with reference to flowchart illustrations and/or block diagrams of methods, apparatus (systems) and computer program products according to embodiments of the invention. It will be understood that each block of the flowchart illustrations and/or block diagrams, and combinations of blocks in the flowchart illustrations and/or block diagrams, can be implemented by computer program instructions. These computer program instructions may be provided to a processor of a general purpose computer, special purpose computer, or other programmable data processing apparatus to produce a machine, such that the instructions, which execute via the processor of the computer or other programmable data processing apparatus, create means for implementing the functions/acts specified in the flowchart and/or block diagram block or blocks.

These computer program instructions may also be stored in a computer-readable memory that can direct a computer or other programmable data processing apparatus to function in a particular manner, such that the instructions stored in the computer-readable memory produce an article of manufacture including instruction means which implement the function/act specified in the flowchart and/or block diagram block or blocks.

The computer program instructions may also be loaded onto a computer or other programmable data processing apparatus to cause a series of operational steps to be performed on the computer or other programmable apparatus to produce a computer implemented process such that the instructions which execute on the computer or other programmable apparatus provide steps for implementing the functions/acts specified in the flowchart and/or block diagram block or blocks.

The present invention relates to detecting sounds that are indicative of incidents including, but not limited to, crime, safety hazards, terrorist threats, or any other event for which a response team may be dispatched or needed. Mobile devices can be loaded with audio analysis software that can recognize particular sounds. This allows the mobile devices to be leveraged throughout a geographic area as sound sensors using the built in audio detection capabilities of the mobile devices. Upon detecting a selected sound, the mobile device can send a communication to an event processor. The communication can specify information relating to the detected sound.

The event processor can evaluate communications received from one or more mobile devices to determine or validate whether an incident associated with the detected sounds has occurred or is occurring. If the information identified from the communications conforms to predetermined criteria, for example, as specified within an incident validation policy, the event processor can take further action. For example, the event processor can provide a notification to an emergency services dispatch center indicating a given sound or incident has been detected.

FIG. **1** is a block diagram illustrating a system **100** in accordance with one aspect of the present invention. The system **100** can include a plurality of mobile devices **105** and **110**, an event processor **125**, as well as a dispatch center **130**. The mobile devices **105** and **110**, the event processor **125**, and the dispatch center **130** can be communicatively linked via the communication network **135**.

The communication network **135** can be implemented as, or include, without limitation, a WAN, a LAN, the Public Switched Telephone Network (PSTN), the Web, the Internet, and one or more intranets. The communication network **135** further can include one or more wireless networks, whether short or long range. For example, in terms of short range wireless networks, the communication network **135** can include a local wireless network built using Bluetooth or one of the IEEE 802 wireless communication protocols, e.g., 802.11a/b/g/i, 802.15, 802.16, 802.20, Wi-Fi Protected Access (WPA), or WPA2. In terms of long range wireless networks, the communication network **135** can include a mobile, cellular, and or satellite-based wireless network and support voice, video, text, and/or any combination thereof, e.g., GSM, TDMA, CDMA, and/or WCDMA network.

The mobile devices **105** and **110** can be implemented as mobile phones, personal digital assistants, or any other device capable of sending and receiving data over wireless communication links **140** and **145** via the communication network **135**. For example, each of the mobile devices **105** and **110** can include a microphone, a processor, and memory. The mobile devices **105** and **110** further can include a wireless transceiver capable of establishing the wireless communication links **140** and **145** with the communication network **135**. The wireless transceiver can support one or more of the various communication protocols noted herein, though the present invention is not intended to be limited by the type of communication scheme or channel used.

The mobile devices **105** and **110** can include a signature detector **115** and one or more sound signatures **120**. The signature detector **115** can be a computer program that is executed by each respective mobile device **105** and **110**. The signature detector **115** can cause each mobile device **105** and **110** to perform the various functions to be described herein. In one embodiment, the mobile devices **105** and **110** can be shipped with the signature detector **115** and the sound signatures **120**. In another embodiment, the signature detector **115** and/or the sound signatures **120** can be installed on the mobile devices **105** and **110** at some other point in time, for example, after purchasing the device by downloading the computer programs and data via a wireless connection. In this regard, it should be appreciated that additional sound signatures **120** can be downloaded over time and that existing sound signatures **120** can be updated and/or deleted. The signature detector **115** also can be updated in this manner.

In general, the signature detector **115** can compare audio that is received by the internal microphone of the mobile device **105** with the sound signatures **120**. The sound signatures **120** are audio profiles of sounds that have been determined to be indicative of an incident. For example, the sound signatures **120** can specify audio profiles for sounds including, but not limited, to, gunshots, explosions, sirens, alarms, breaking glass, auto accidents, yelling or screaming, calls for

help, etc. The potential range of audio events for which a sound signature **120** can be included in a mobile device is limited only by the available memory or storage capacity of the respective mobile device.

The signature detector **115** can extract information from the detected sound and compare that information to the sound signatures **120**. Examples of the sort of data that can be extracted or determined from the detected sound can include, but are not limited to, volume or sound-pressure-level information, the frequency range of the detected sound, the amount of energy detected in different frequency bands of interest, a spectrum analysis, transient characteristics, the actual waveform of the detected sound, a Fourier Transform or FFT information, formant information, or the like. These parameters further can be measured over time. The various parameters listed herein are intended as examples only and, as such, are not intended to limit the present invention in any way.

In addition to comparing received, or detected, sounds with the sound signatures **120**, the signature detector **115** can control one or more other functions of the mobile devices **105** and **110**. For example, the signature detector **115** can cause the mobile devices **105** and **110** to keep the microphone active so that sounds are continually monitored, recorded, and compared with the stored sound signatures **120**. The mobile devices **105** and **110** further can communicate with the event processor **125** under the control of the signature detector **115**, interpret communications received from the event processor **125**, as well as respond to requests from the event processor **125**, such as providing recorded audio.

The event processor **125** can be implemented as an information processing system executing suitable operational software, e.g., a server. The event processor **125** can receive information from the mobile devices **105** and **110**, analyze that information, and based upon the analysis, contact the dispatch center **130**. The event processor **125** further can query the mobile devices **105** and **110** for additional information as described herein.

The dispatch center **130** can include any of a variety of communication systems capable of receiving information from the event processor **125**. The dispatch center **130**, for example, can be a police dispatch center, an emergency services dispatch center, 911 call center, or the like. The dispatch center **130** can be any sort of facility that is linked with the event processor **125** that, for example, can dispatch resources to address an incident detected using the mobile devices **105** and **110**.

In operation, the mobile devices **105** and **110** can detect an audio event, or sound, **150**. The signature detector **115** within each respective mobile device **105** and **110** can process the detected sound **150** and compare the sound **150** against the sound signatures **120** stored within the mobile devices **105** and **110**. It should be appreciated that as the location of each mobile device **105** and **110** will be different, the characteristics of the detected sound **150**, e.g., volume, frequency range, and the like, may differ as well, particularly if one or both of the mobile devices **105** and **110** is in motion.

In any case, each mobile device **105** and **110** can detect the sound **150** independently of the other and perform its own independent analysis. If the signature detector **115** within either one or both of mobile devices **105** and **110** can determine that the sound **150** matches a sound signature **120**, a communication can be sent to the event processor **125** from that mobile device. It should be appreciated that since each mobile device **105** and **110** performs its own analysis, mobile device **105** may determine that the sound **150** matches a sound signature **120**, while mobile device **110** determines that the sound **150** does not match any sound signatures **120** or possibly a different sound signature **120**. Such can be the case, as noted, due to distance, motion, or the possibility that one mobile device is located in a noisy environment, while the other is not.

While mobile device **105** is depicted as being in communication with the event processor **125**, it should be appreciated that communication device **110** also can conduct the same sort of information exchange with the event processor **125** as is described herein with respect to communication device **105**. Further, though only two mobile devices are shown, it should be appreciated that many more mobile devices can detect the sound **150** and perform the various processing functions disclosed herein. Such mobile devices also can communicate with the event processor **125**. The present invention lends itself to having many mobile devices, dispersed throughout an area and potentially in motion, continually detecting sounds.

Upon determining that the sound **150** matches a sound signature **120**, the mobile device **105** can send a communication to the event processor **125**. In one embodiment, the communication can be a Short Message Service (SMS) message **155**. The communication can provide one or more items of information relating to the detected sound **150**. For example, one or more parameters of the detected sound **150** that may be determined or extracted for comparison with the sound signatures **120** can be sent. Other data generated by the signature detector **115** also can be sent such as, for instance, a measure of how closely the sound **150** matches the sound signature **120**. The information sent within the communication, i.e., SMS message **155**, can include any information determined by the signature detector **115** and, in one embodiment, initially exclude any audio that have been recorded or collected by the mobile device **105**.

Spatial information, such as the location of the mobile device **105** when the sound **150** is detected can be included in the communication. Such information can be determined through conventional mobile triangulation techniques, using a Global Positioning System (GPS) receiver included in the mobile device **105**, or the like. Regardless of how location information is ascertained, spatial information can be determined by the mobile device **105** and inserted into the communication. Temporal information, e.g., a timestamp specifying the time of day and the date, when the sound **150** is detected also can be specified within the communication. Further, an identifier capable of uniquely specifying the particular sound signature **120** that was matched to the sound **150** can be included in the communication.

The event processor **125** can receive communications, such as SMS **155**, from any mobile device that detects the sound **150**, or any other sound matching a sound signature **120**, for that matter. The event processor **125** can analyze the received information from the various mobile devices and compare that information with an incident validation policy to determine when a valid incident has taken place or is taking place.

The event processor **125** can be programmed to identify communications corresponding to a same detected sound. In some cases, the sound signatures indicated by such messages will be the same. Further, the temporal and spatial information indicated by the communications will be close. That is, the mobile devices that detected the sound **150** will have been located within a predetermined distance of one another as determined from the spatial information in the communications from each respective mobile device. Further, the time at which each mobile device detected the sound **150** will have

been within a predetermined amount of time of one another as determined from the temporal information specified within the communications.

It should be appreciated the interpretation of spatial and temporal information can vary according to the particular sound signatures that are detected. The temporal and/or spatial constraints used to interpret data received from mobile devices can vary according to the particular sound signatures detected. This can be specified as part of the validation policy, for example.

In illustration, if two mobile devices detect glass breaking, but are located more than a mile apart when the sound is detected, the event processor can determine that the mobile devices detected two separate instances of glass breaking. The same can be determined from temporal information, e.g., if two detections of glass breaking occur more than 5 or 10 seconds apart, the two detections can be interpreted as separate incidents of glass breaking. If, however, an explosion is detected by two mobile devices located approximately one mile apart, the event processor can apply different spatial constraints and determine that one explosion likely occurred, but was detected by two mobile devices.

In some cases, however, the two mobile devices **105** and **110** may detect the same audio event, but interpret the audio event differently. One mobile device, for example, can match the audio event with an incorrect sound signature **120**. In that case, the event processor **125** can determine that the audio events are either the same audio event or are at least related, for example, if the audio events were detected by each respective mobile device within a predetermine amount of time of one another and the mobile devices that detected the audio event were within a predetermined distance of one another. In such cases, the event processor **125** can determine that one of the devices interpreted the detected sound incorrectly. Again, the rules applied can vary according to the particular sound signatures detected.

In other cases, the event processor **125** can determine that the sounds are actually different, but likely relate to a same incident. For example, mobile device **105** detects glass breaking and mobile device **110** detects a siren. If both devices detect the sounds within a predetermine amount of time of one another and are located within a predetermined distance of one another, the event processor **125** can determine that although two different audio events are detected, the audio events are indicative of a single incident, e.g., a burglary.

The communications can be compared with a policy that determines when, and under what circumstances, a notification **170** can be provided to the dispatch center or that a valid incident is or has occurred. For example, the validation policy can specify a minimal number of communications (mobile devices) that must detect a particular sound signature before a valid incident is determined to have occurred and a notification **170** is sent. The validation policy further can specify a minimal confidence score that the event processor **125** must calculate from information specified within the communications to determine that a valid incident has occurred. The policy further can specify spatial and/or location proximity guidelines for sound detections by multiple mobile devices that are indicative of a valid incident or indicate that multiple sounds relate to a same incident.

In one embodiment, the event processor can directly send the notification **170** to the dispatch center. In another embodiment, the event processor **125** can send a request **160** to the mobile device **105** asking for recorded audio of the detected sound **150**. The request **160** can be formatted and/or sent as any of a variety of different communication, e.g., an SMS message. In one aspect, the validation policy can specify when further information is to be requested from a mobile device **105** or **110**.

The mobile device **105** can send a recording **165** of the detected sound over the voice channel of the mobile network. This embodiment presumes that the mobile device **105** and **110** can be configured to continually record audio. In one embodiment, for example, the mobile device can record a continuous loop and, if a detected sound matches a sound signature **120**, the signature detector **115** can prevent that audio from being recorded over until the mobile device either provides the audio to the event processor **125** or determines that the audio is not needed by the event processor **125**.

In either case, any information either received by the event processor **125** and/or determined by the event processor **125** can be forwarded to the dispatch center **130** within the notification **170**. For example, the notification **170** can indicate the various locations at which mobile devices detected sounds, the sound signatures that were detected, the time at which each sound signature was detected, or the like. If recordings are received by the event processor **125**, then recordings can be sent to the dispatch center **130** as well.

In another embodiment, the event processor **125** can provide a summary of the information received from the mobile devices **105** and **110** and/or any determinations that were made in terms of calculating that a valid incident occurred. For example, the event processor **125** can send a particular location to the dispatch center **130** rather than all locations from which a mobile device detected the sound signature **120**. In another example, if multiple sound signatures are determined to be related by the event processor **125**, a single notification **170** can be sent rather than one for each type of sound signature detected.

The particular information provided from the event processor **125** to the dispatch center **130** is not intended to limit the present invention, as any information collected and/or determined by the event processor can be made available to the dispatch center **130**. Once the notification is received by the dispatch center **130**, personnel within that center can cause response personnel to be sent or dispatched to a location related to the detected sounds.

FIG. **2** is a flow chart illustrating a method **200** in accordance with another aspect of the present invention. The method **200** can be performed by a system such as the system described with reference to FIG. **1**. In step **205**, one or more mobile devices can detect a sound through the internal microphone included as part of each respective mobile device.

In step **210**, each mobile device can analyze the detected sound and compare the detected sound against one or more sound signatures stored within the mobile device. The mobile device(s) can determine that the sound matches one of the sound signatures stored within the mobile device. In step **215**, the mobile device, upon determining that the detected sound matches one of the stored sound signatures, can send a communication to the event processor.

In step **220**, the event processor can analyze any communications received from mobile devices. As noted, the event processor can analyze received communications for spatial and temporal proximity, the sort of sound signature indicated within the communications, and the like. The event processor, based upon the analysis of the various messages received from mobile devices, can determine whether a valid incident has been detected. As noted, the data obtained from the communications from the mobile devices can be compared with an incident validation policy to determine whether a valid incident has been detected.

In step **225**, the event processor optionally can request audio from one or more of the mobile devices. In one embodiment, audio can be requested from each mobile device from which a communication was received regarding a detected sound. For example, the event processor can request audio from one or more mobile devices as a matter of course. In another aspect, when the event processor cannot determine whether an event is valid from the communications from the mobile devices, the event processor can request audio from one or more mobile devices. In that case, the information specified in the communications is not sufficient to validate the occurrence of an event or is otherwise ambiguous. Obtaining actual audio and subjecting the audio to further analysis can aid the event processor in validating the occurrence of an incident.

In another embodiment, the event processor can select one or more mobile devices from which audio will be requested. For example, the event processor can review information about the detected sound such as the volume and select the mobile device that detected the sound with the greatest volume. The event processor can query that mobile device to send recorded audio of the detected sound under the presumption that the recording will be of better quality than recordings from other mobile devices.

In another aspect, the event processor can request audio from the mobile device that first detected the audio, last detected the audio, or sequentially obtain audio from the mobile devices in chronological order according to the time each respective mobile device detected the sound. Any audio parameter provided by the mobile devices can serve as a means of selecting the particular mobile device from which the recording will be obtained. For example, the communication from each mobile device can indicate the signal-to-noise ratio, background noise levels of the audio environment in which the mobile device is disposed, or other noise information pertaining to the recording of the sound made by that mobile device. For example, the event processor can select the mobile device with the recording considered to be of better fidelity in terms of signal-to-noise, or any other combination or parameters, whether spatial, temporal, or audio quality.

In any case, the recording can be received from the mobile device(s). As noted, the event processor can perform its own analysis and comparison of the detected sound, from the recording(s) furnished by the mobile device(s), with one or more sound signatures. The event processor, for example, can include more robust and/or sophisticated audio analysis software than can reasonably be included or executed in a mobile device.

In step **230**, if the event is validated, a notification can be sent to a dispatch center. As noted, personnel in the dispatch center can scramble personnel as may be required to address or respond to the incident.

The flowchart and block diagrams in the figures illustrate the architecture, functionality, and operation of possible implementations of systems, methods and computer program products according to various embodiments of the present invention. In this regard, each block in the flowchart or block diagrams may represent a module, segment, or portion of code, which comprises one or more executable instructions for implementing the specified logical function(s). It should also be noted that, in some alternative implementations, the functions noted in the block may occur out of the order noted in the figures. For example, two blocks shown in succession may, in fact, be executed substantially concurrently, or the blocks may sometimes be executed in the reverse order, depending upon the functionality involved. It will also be noted that each block of the block diagrams and/or flowchart illustration, and combinations of blocks in the block diagrams and/or flowchart illustration, can be implemented by special purpose hardware-based systems that perform the specified functions or acts, or combinations of special purpose hardware and computer instructions.

The terminology used herein is for the purpose of describing particular embodiments only and is not intended to be limiting of the invention. As used herein, the singular forms "a", "an", and "the" are intended to include the plural forms as well, unless the context clearly indicates otherwise. It will be further understood that the terms "comprises" and/or "comprising", when used in this specification, specify the presence of stated features, integers, steps, operations, elements, and/or components, but do not preclude the presence or addition of one or more other features, integers, steps, operations, elements, components, and/or groups thereof.

The corresponding structures, materials, acts, and equivalents of all means or step plus function elements in the claims below are intended to include any structure, material, or act for performing the function in combination with other claimed elements as specifically claimed. The description of the present invention has been presented for purposes of illustration and description, but is not intended to be exhaustive or limited to the invention in the form disclosed. Many modifications and variations will be apparent to those of ordinary skill in the art without departing from the scope and spirit of the invention. The embodiment was chosen and described in order to best explain the principles of the invention and the practical application, and to enable others of ordinary skill in the art to understand the invention for various embodiments with various modifications as are suited to the particular use contemplated.

Having thus described the invention of the present application in detail and by reference to the embodiments thereof, it will be apparent that modifications and variations are possible without departing from the scope of the invention defined in the appended claims.

What is claimed is:

1. A method of identifying incidents using mobile devices comprising:

receiving a communication from each of a plurality of mobile devices, wherein each communication specifies information about a first detected sound;

identifying spatial and temporal information from each communication and an indication of a sound signature matching the first detected sound;

comparing the spatial and temporal information from the communications with a policy specifying a spatial constraint and a temporal constraint, each of the spatial constraint and the temporal constraint selected according to the sound signature indicated by the communications and varying according to the sound signature;

selectively sending a notification according to the comparison;

responsive to comparing the communications with the policy, requesting a recording of the first detected sound from at least one of the plurality of mobile devices; and

receiving the recording of the first detected sound.

2. The method of claim **1**, wherein receiving a communication further comprises receiving a recording of the sound from at least one of the plurality of mobile devices.

3. The method of claim **2**, further comprising first selecting a mobile device from which the recording of the first detected sound is requested from the plurality of mobile devices.

4. The method of claim 1, further comprising:

controlling a function of at least one of the mobile devices to cause the mobile device to keep its microphone active so that additional sounds are continually monitored and compared with the sound signature.

5. The method of claim 1, further comprising:

determining that the communication from a first of the mobile devices specifies information about the first detected sound;

determining that the communication from a second of the mobile devices specifies information about a second detected sound; and

determining that the first detected sound and the second detected sound are actually different, but likely relate to a same incident.

6. The method of claim 5, wherein determining that the first detected sound and the second detected sound are actually different, but likely relate to the same incident, comprises:

determining that the second mobile device detects the second detected sound within a predetermined amount of time of when the first mobile device detects the first detected sound; and

determining that the first mobile device and the second mobile device are located within a predetermined distance of one another.

7. A method of identifying incidents using mobile devices comprising:

receiving a communication from each of a plurality of mobile devices, wherein each communication specifies a first detected sound;

comparing the communications with a validation policy that specifies a spatial constraint and a temporal constraint, each of the spatial constraint and the temporal constraint selected according to a sound signature matching the first detected sound and varying according to the sound signature;

selectively sending a notification according to the comparison; and

selecting the mobile device from which the recording of the first detected sound is requested from the plurality of mobile devices.

8. The method of claim 7, further comprising providing the recording of the first detected sound with the notification.

9. The method of claim 7, further comprising:

controlling a function of at least one of the mobile devices to cause the mobile device to keep its microphone active so that additional sounds are continually monitored and compared with the sound signature.

10. The method of claim 7, further comprising:

determining that the communication from a first of the mobile devices specifies information about the first detected sound;

determining that the communication from a second of the mobile devices specifies information about a second detected sound; and

determining that the first detected sound and the second detected sound are actually different, but likely relate to a same incident.

11. The method of claim 10, wherein determining that the first detected sound and the second detected sound are actually different, but likely relate to the same incident, comprises:

determining that the second mobile device detects the second detected sound within a predetermined amount of time of when the first mobile device detects the first detected sound; and

determining that the first mobile device and the second mobile device are located within a predetermined distance of one another.

12. A computer program product comprising:

a computer-readable storage device having computer-usable program code stored thereon, the computer-usable program code causing the computer to execute in order to identify incidents using mobile devices, said computer program product including:

computer-usable program code that receives a communication from each of a plurality of mobile devices, wherein each communication specifies information about a first detected sound;

computer-usable program code that identifies spatial and temporal information from each communication and an indication of a sound signature matching the first detected sound;

computer-usable program code that compares the spatial and temporal information from the communications with a policy specifying a spatial constraint and a temporal constraint, each of the spatial constraint and the temporal constraint selected according to the sound signature indicated by the communications and varying according to the sound signature;

computer-usable program code that selectively sends a notification according to the comparison;

computer-usable program code that, responsive to comparing the communications with the policy, requests a recording of the sound from at least one of the plurality of mobile devices; and

computer-usable program code that receives the recording of the first detected sound, wherein

the computer-readable storage device is not a transitory, propagating signal per se.

13. The computer program product of claim 12, wherein computer-usable program code that receives a communication further comprises computer-usable program code that receives a recording of the sound from at least one of the plurality of mobile devices.

14. The computer program product of claim 13, further comprising computer-usable program code that selects the mobile device from which the recording of the first detected sound is requested from the plurality of mobile devices.

15. The computer program product of claim 12, further comprising:

computer-usable program code that controls a function of at least one of the mobile devices to cause the mobile device to keep its microphone active so that additional sounds are continually monitored and compared with the sound signature.

16. The computer program product of claim 12, further comprising:

computer-usable program code that determines that the communication from a first of the mobile devices specifies information about the first detected sound;

computer-usable program code that determines that the communication from a second of the mobile devices specifies information about a second detected sound; and

computer-usable program code that determines that the first detected sound and the second detected sound are actually different, but likely relate to a same incident.

17. The computer program product of claim 16, wherein the computer-usable program code that determines that the first detected sound and the second detected sound are actually different, but likely relate to the same incident, comprises:

Exhibit PA-V

computer-usable program code that determines that the second mobile device detects the second detected sound within a predetermined amount of time of when the first mobile device detects the first detected sound; and

computer-usable program code that determines that the first mobile device and the second mobile device are located within a predetermined distance of one another.

* * * * *

Exhibit PAT-I


US010535334B2

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

(10) **Patent No.: US 10,535,334 B2**
(45) **Date of Patent: *Jan. 14, 2020**

(54) **METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

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Related U.S. Application Data

(63) Continuation of application No. 14/574,589, filed on Dec. 18, 2014, now Pat. No. 10,134,377, which is a continuation of application No. 12/017,878, filed on Jan. 22, 2008, now Pat. No. 8,917,894.

(60) Provisional application No. 60/885,917, filed on Jan. 22, 2007.

(51) **Int. Cl.**

G10K 11/178	(2006.01)
H04R 3/00	(2006.01)
H04R 1/10	(2006.01)
G10K 11/00	(2006.01)
H04R 29/00	(2006.01)

(52) **U.S. Cl.**

CPC ***G10K 11/17827*** (2018.01); ***G10K 11/002*** (2013.01); ***H04R 1/1016*** (2013.01); ***H04R 1/1083*** (2013.01); ***H04R 3/002*** (2013.01); ***H04R 3/005*** (2013.01); ***H04R 29/001*** (2013.01); *H04R 2225/41* (2013.01); *H04R 2460/05* (2013.01); *H04R 2499/11* (2013.01)

(58) **Field of Classification Search**

CPC G10K 11/17827; G10K 11/002; H04R 29/001; H04R 1/1026; H04R 1/1083; H04R 3/002; H04R 3/005; H04R 2225/41; H04R 2460/05; H04R 2499/11

See application file for complete search history.

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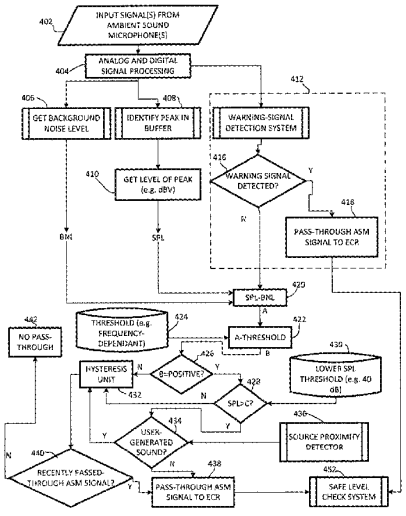
Primary Examiner — Andrew L Sniezek

(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti; Mammen (Roy) P. Zachariah, Jr.

(57) **ABSTRACT**

Earpieces and methods for acute sound detection and reproduction are provided. A method can include measuring an external ambient sound level (xASL), monitoring a change in the xASL for detecting an acute sound, estimating a proximity of the acute sound, and upon detecting the acute sound and its proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece. Other embodiments are disclosed.

20 Claims, 9 Drawing Sheets



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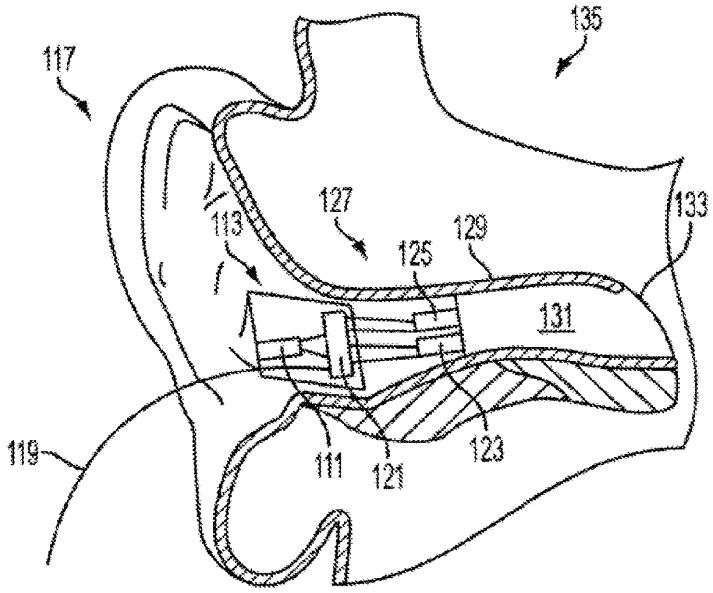


FIG. 1 100

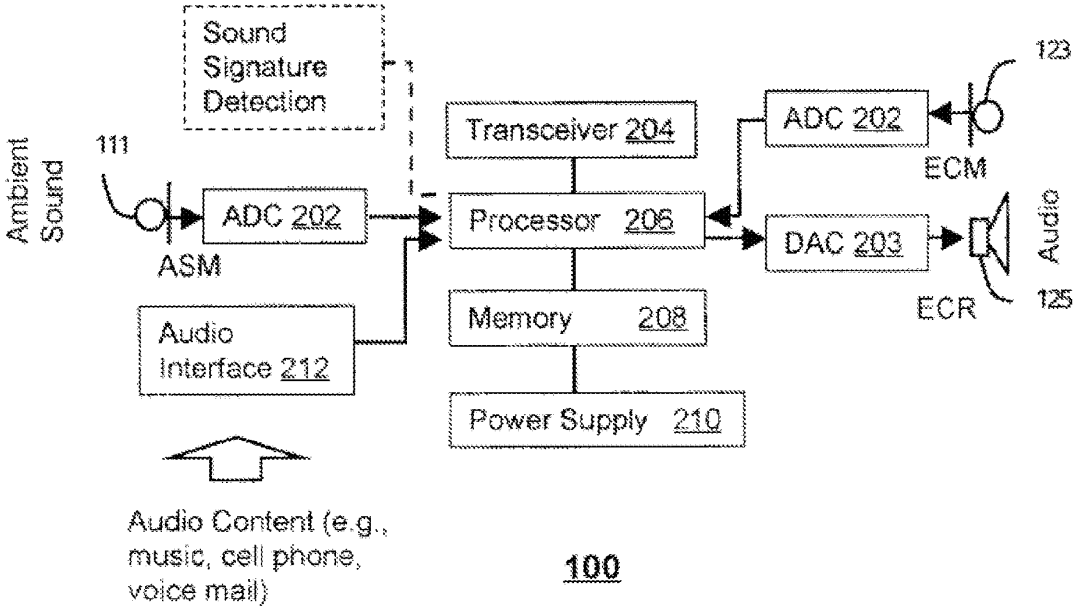
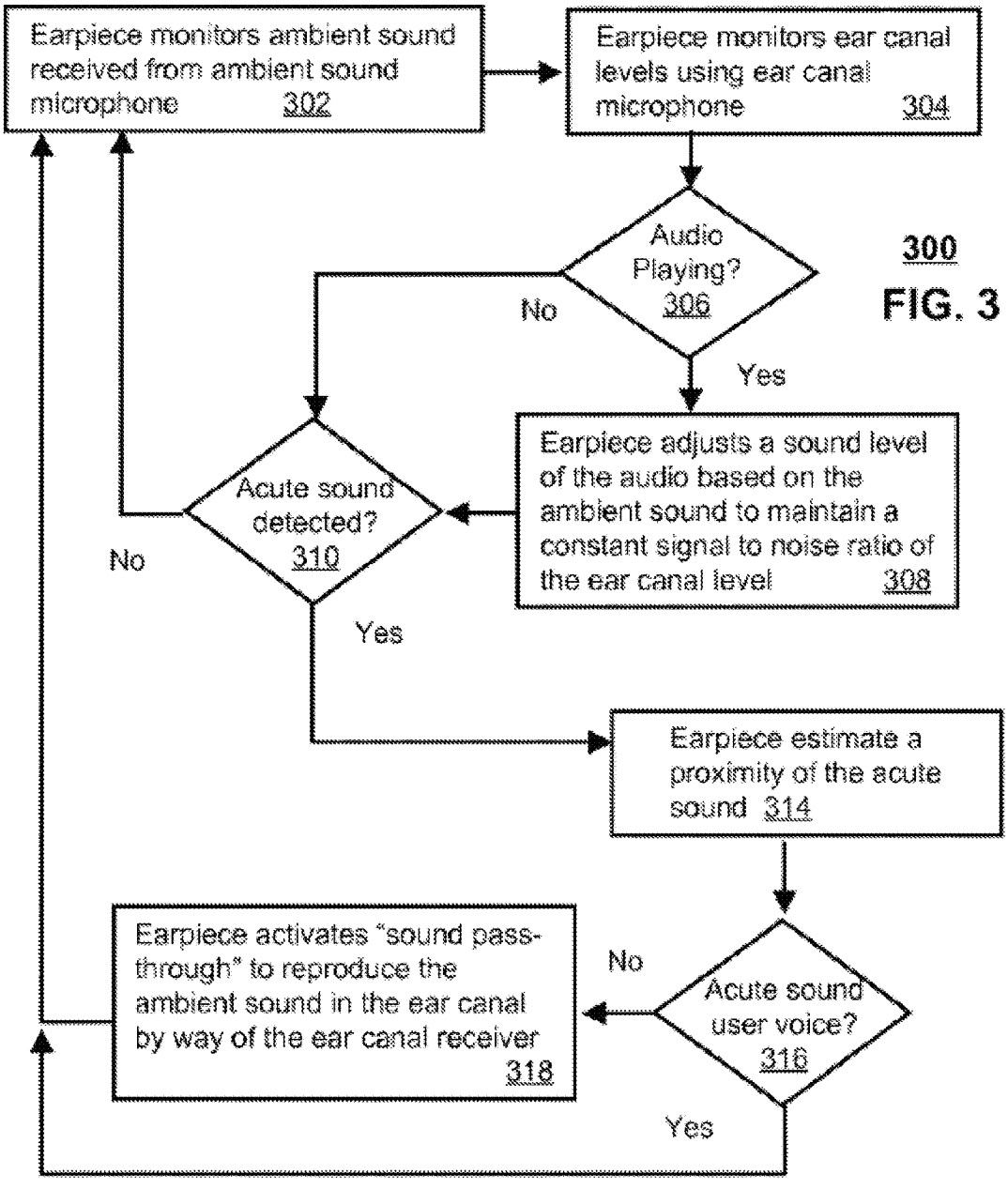


FIG. 2


Earpiece monitors ambient sound received from ambient sound microphone 302
Earpiece monitors ear canal levels using ear canal microphone 304
Audio Playing? 306
No
Yes
Earpiece adjusts a sound level of the audio based on the ambient sound to maintain a constant signal to noise ratio of the ear canal level 308
Acute sound detected? 310
No
Yes
Earpiece estimate a proximity of the acute sound 314
Acute sound user voice? 316
No
Yes
Earpiece activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ear canal receiver 318

300

FIG. 3

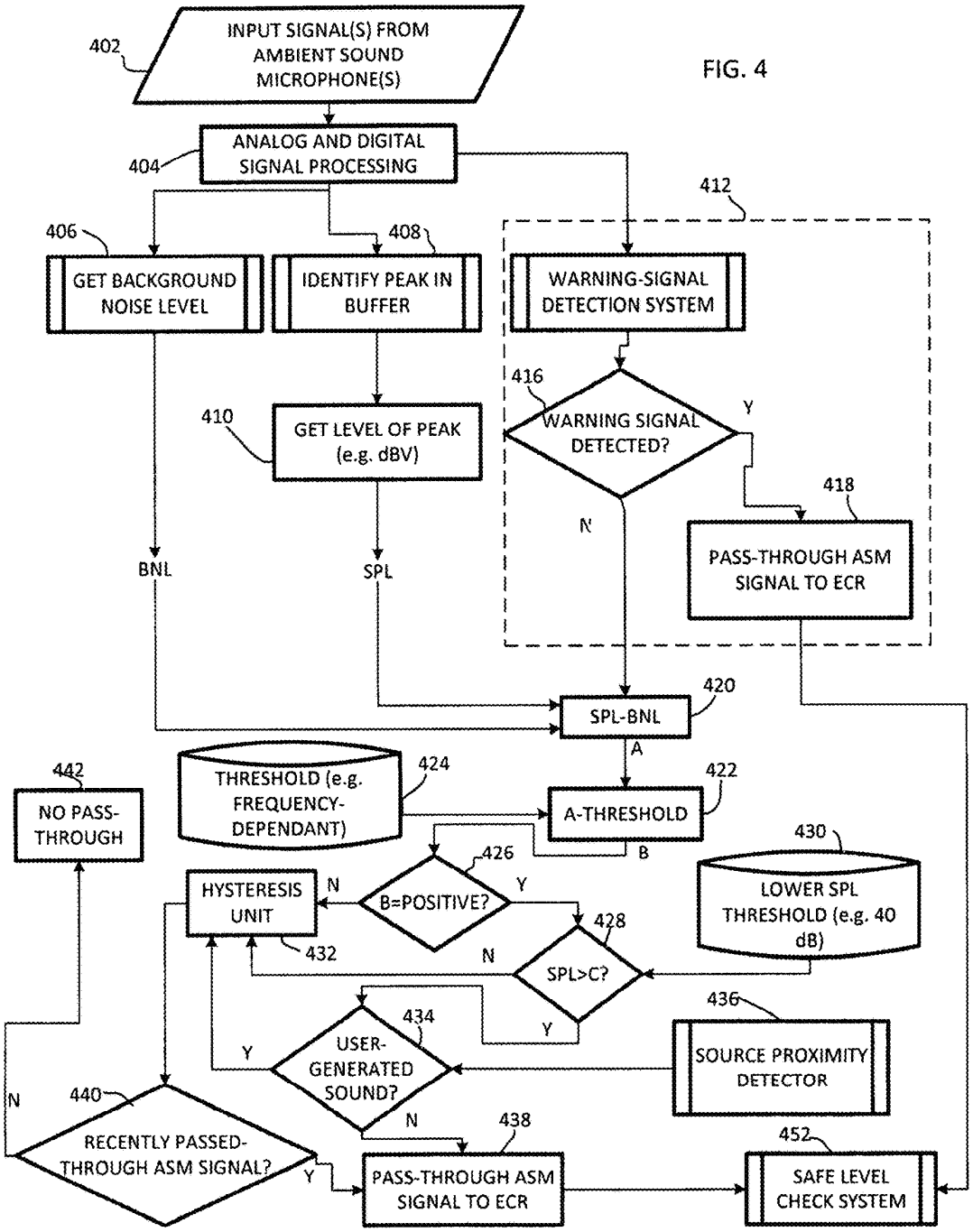
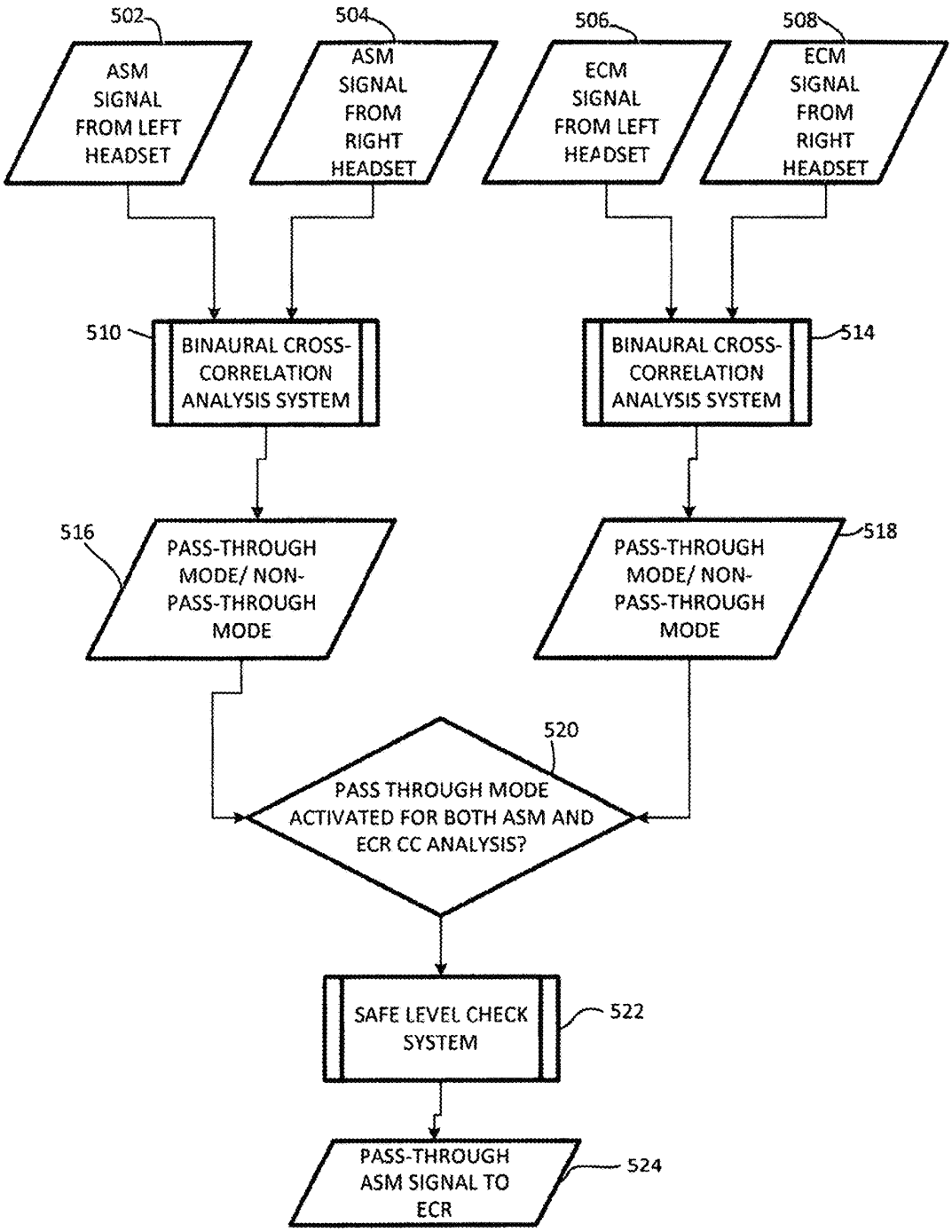

402
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONE(S)
404
ANALOG AND DIGITAL SIGNAL PROCESSING
406
GET BACKGROUND NOISE LEVEL
408
IDENTIFY PEAK IN BUFFER
410
GET LEVEL OF PEAK (e.g. dBV)
412
WARNING-SIGNAL DETECTION SYSTEM
416
WARNING SIGNAL DETECTED?
Y
N
418
PASS-THROUGH ASM SIGNAL TO ECR
BNL
SPL
420
SPL-BNL
A
422
A-THRESHOLD
B
424
THRESHOLD (e.g. FREQUENCY-DEPENDANT)
426
B=POSITIVE?
N
Y
428
SPL>C?
N
Y
430
LOWER SPL THRESHOLD (e.g. 40 dB)
432
HYSTERESIS UNIT
434
USER-GENERATED SOUND?
Y
N
436
SOURCE PROXIMITY DETECTOR
438
PASS-THROUGH ASM SIGNAL TO ECR
440
RECENTLY PASSED-THROUGH ASM SIGNAL?
N
Y
442
NO PASS-THROUGH
452
SAFE LEVEL CHECK SYSTEM

FIG. 4

FIG. 5


502
ASM SIGNAL FROM LEFT HEADSET
504
ASM SIGNAL FROM RIGHT HEADSET
506
ECM SIGNAL FROM LEFT HEADSET
508
ECM SIGNAL FROM RIGHT HEADSET
510
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
514
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
516
PASS-THROUGH MODE/ NON-PASS-THROUGH MODE
518
PASS-THROUGH MODE/ NON-PASS-THROUGH MODE
520
PASS THROUGH MODE ACTIVATED FOR BOTH ASM AND ECR CC ANALYSIS?
SAFE LEVEL CHECK SYSTEM
522
PASS-THROUGH ASM SIGNAL TO ECR
524

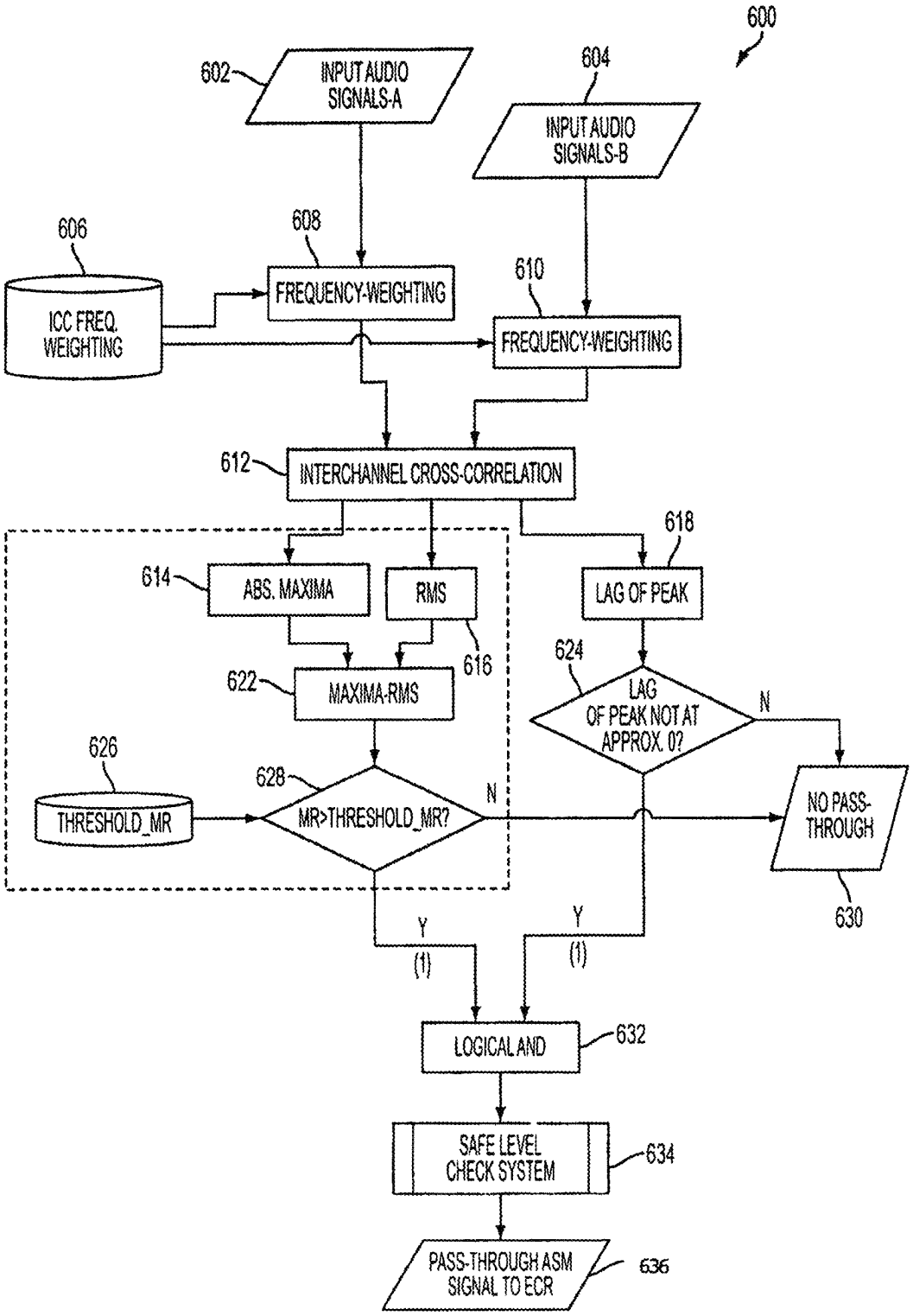
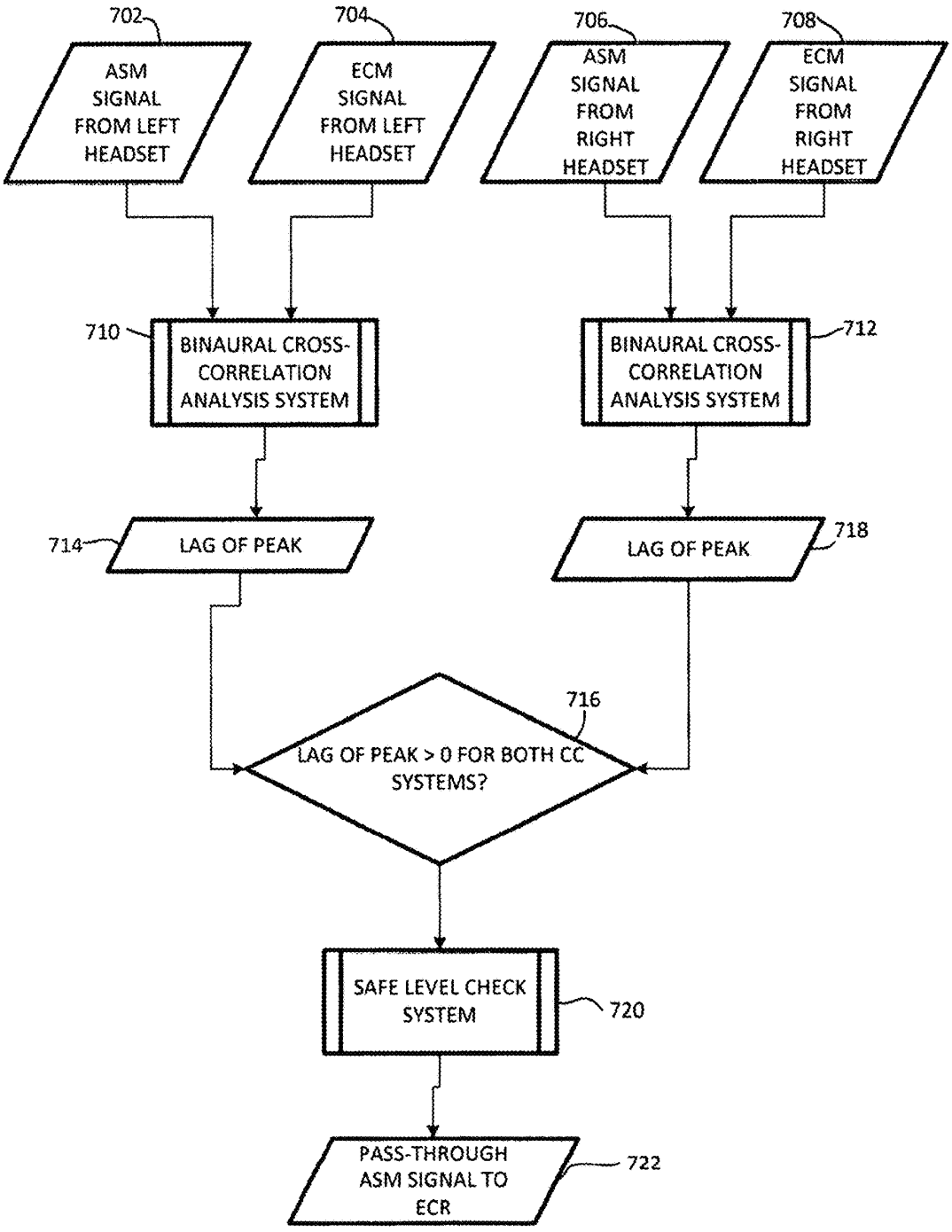

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH
630
Y
(1)
Y
(1)
LOGICAL AND
632
SAFE LEVEL CHECK SYSTEM
634
PASS-THROUGH ASM SIGNAL TO ECR
636

FIG. 6

FIG. 7


702
ASM SIGNAL FROM LEFT HEADSET
704
ECM SIGNAL FROM LEFT HEADSET
706
ASM SIGNAL FROM RIGHT HEADSET
708
ECM SIGNAL FROM RIGHT HEADSET
710
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
712
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
714
LAG OF PEAK
718
LAG OF PEAK
716
LAG OF PEAK > 0 FOR BOTH CC SYSTEMS?
SAFE LEVEL CHECK SYSTEM
720
PASS-THROUGH ASM SIGNAL TO ECR
722

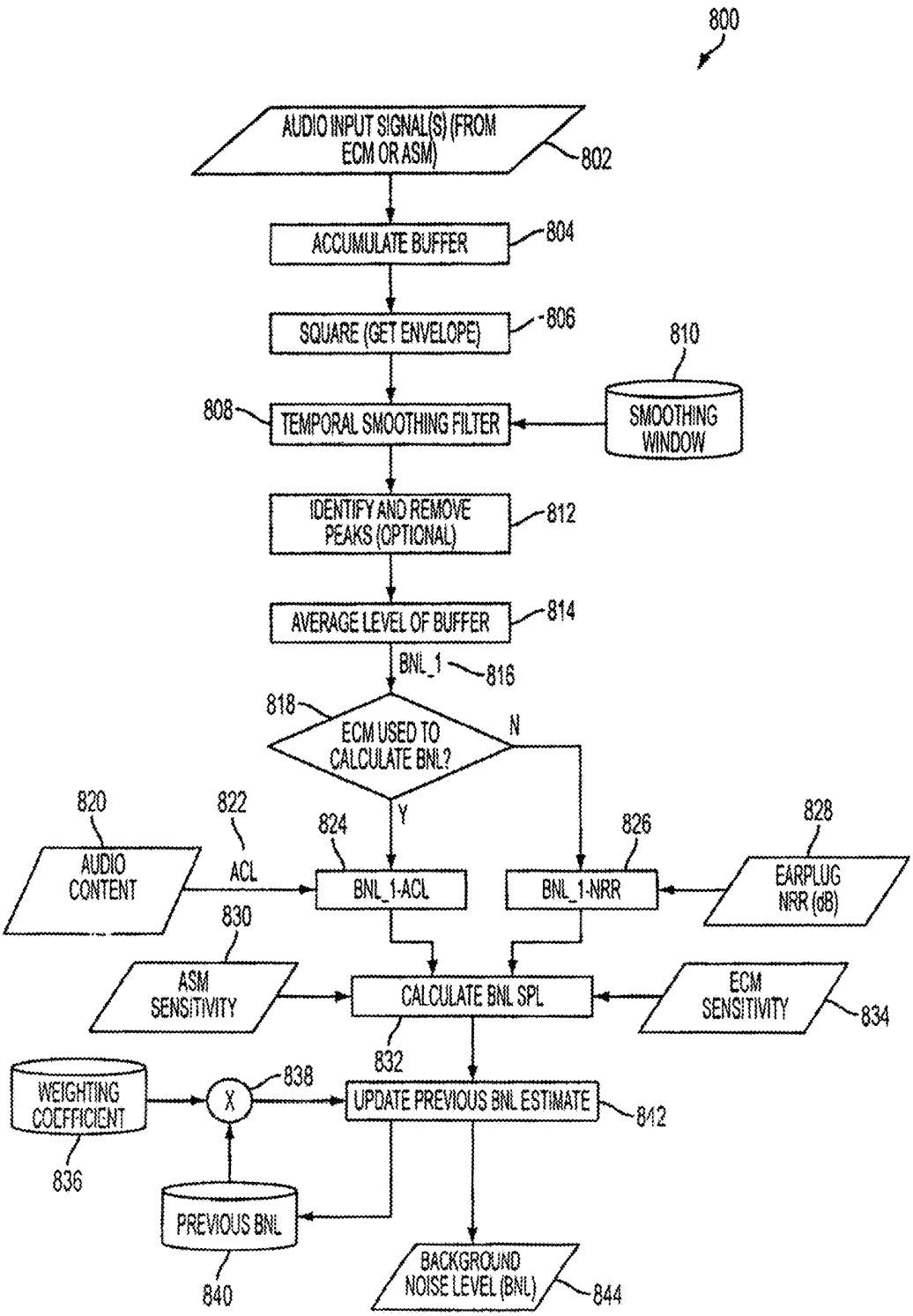

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT
822
ACL
824
BNL_1-ACL
826
BNL_1-NRR
828
EARPLUG NRR (dB)
830
ASM SENSITIVITY
CALCULATE BNL SPL
ECM SENSITIVITY
834
832
WEIGHTING COEFFICIENT
838
X
UPDATE PREVIOUS BNL ESTIMATE
842
836
PREVIOUS BNL
840
BACKGROUND NOISE LEVEL (BNL)
844

FIG. 8

FIG. 9

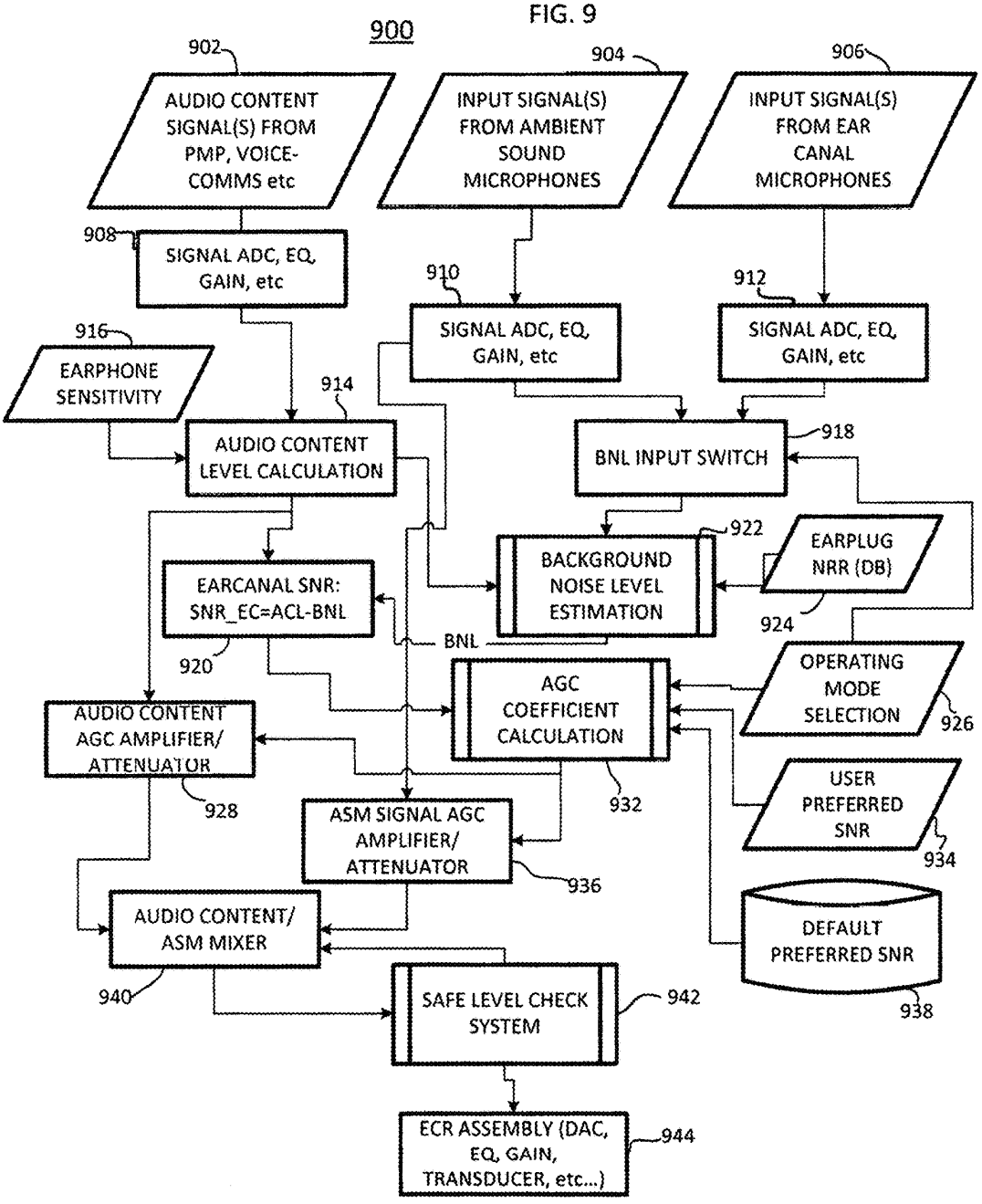
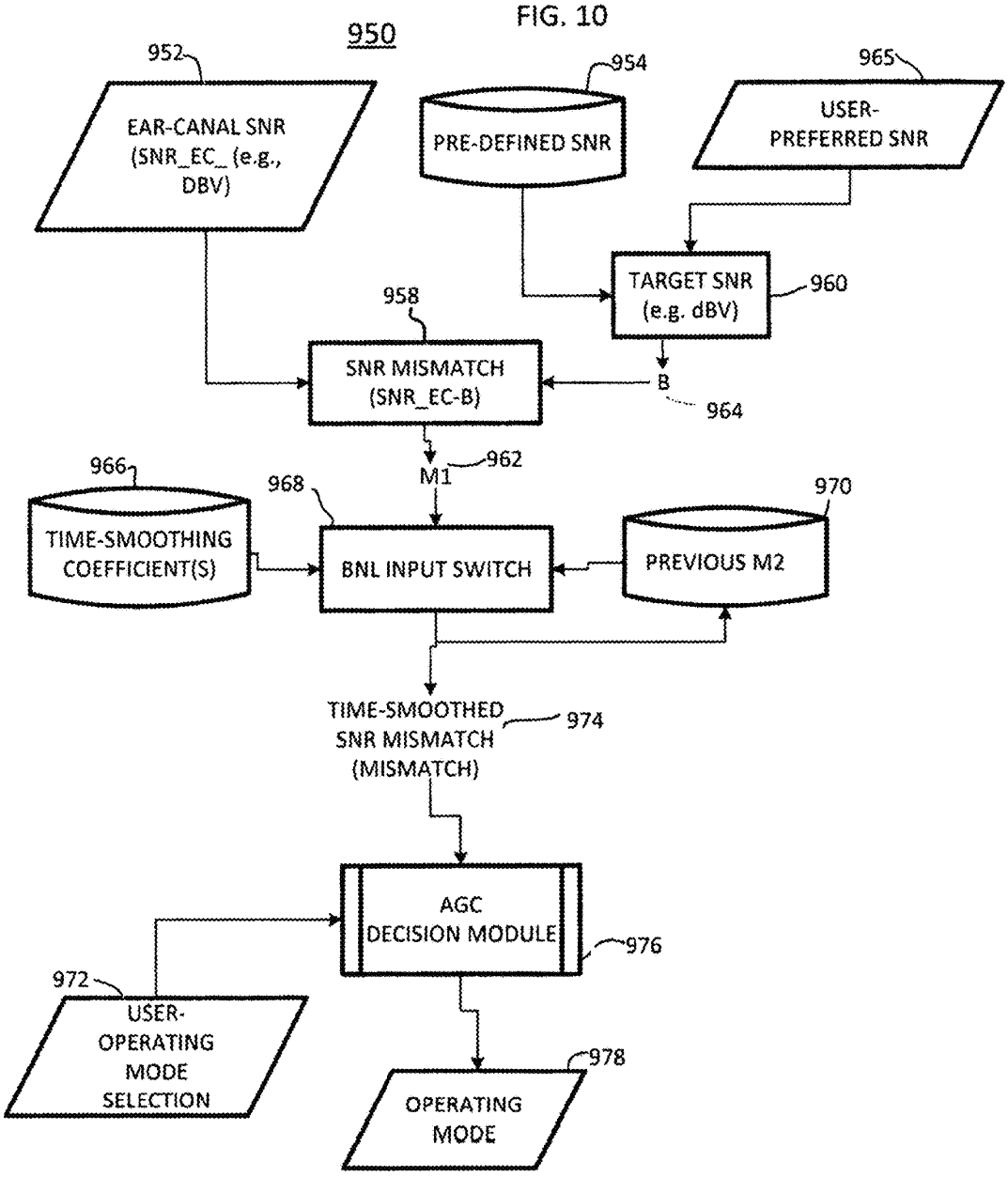

900
902
AUDIO CONTENT SIGNAL(S) FROM PMP, VOICE-COMMS etc
904
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES
906
INPUT SIGNAL(S) FROM EAR CANAL MICROPHONES
908
SIGNAL ADC, EQ, GAIN, etc
910
SIGNAL ADC, EQ, GAIN, etc
912
SIGNAL ADC, EQ, GAIN, etc
916
EARPHONE SENSITIVITY
914
AUDIO CONTENT LEVEL CALCULATION
BNL INPUT SWITCH
918
922
BACKGROUND NOISE LEVEL ESTIMATION
EARPLUG NRR (DB)
924
EARCANAL SNR: SNR_EC=ACL-BNL
920
BNL
OPERATING MODE SELECTION
926
AUDIO CONTENT AGC AMPLIFIER/ ATTENUATOR
928
AGC COEFFICIENT CALCULATION
932
USER PREFERRED SNR
934
ASM SIGNAL AGC AMPLIFIER/ ATTENUATOR
936
AUDIO CONTENT/ ASM MIXER
940
DEFAULT PREFERRED SNR
938
SAFE LEVEL CHECK SYSTEM
942
ECR ASSEMBLY (DAC, EQ, GAIN, TRANSDUCER, etc...)
944

FIG. 10


950
952
EAR-CANAL SNR (SNR_EC_ (e.g., DBV)
954
PRE-DEFINED SNR
965
USER-PREFERRED SNR
TARGET SNR (e.g. dBV)
960
958
SNR MISMATCH (SNR_EC-B)
B
964
M1
962
966
TIME-SMOOTHING COEFFICIENT(S)
968
BNL INPUT SWITCH
970
PREVIOUS M2
TIME-SMOOTHED SNR MISMATCH (MISMATCH)
974
AGC DECISION MODULE
976
972
USER-OPERATING MODE SELECTION
978
OPERATING MODE

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

CROSS-REFERENCE TO RELATED APPLICATION(S)

This application is a continuation of and claims priority to U.S. patent application Ser. No. 14/574,589, filed on Dec. 18, 2014, which claims priority to and is a continuation of U.S. patent application Ser. No. 12/017,878, filed on Jan. 22, 2008, now U.S. Pat. No. 8,917,894, which claims the benefit of U.S. Provisional Patent Application Ser. No. 60/885,917, filed on Jan. 22, 2007, all of which are herein incorporated by reference in their entireties.

FIELD OF THE INVENTION

The present invention relates to a device that monitors sound directed to an occluded ear, and more particularly, though not exclusively, to an earpiece and method of operating an earpiece that detects acute sounds and allows the acute sounds to be reproduced in an ear canal of the occluded ear.

BACKGROUND

Since the advent of industrialization over two centuries ago, the human auditory system has been increasingly stressed to tolerate high noise levels to which it had hitherto been unexposed. Recently, human knowledge of the causes of hearing damage have been researched intensively and models for predicting hearing loss have been developed and verified with empirical data from decades of scientific research. Yet it can be strongly argued that the danger of permanent hearing damage is more present in our daily lives than ever, and that sound levels from personal audio systems in particular (i.e. from portable audio devices), live sound events, and the urban environment are a ubiquitous threat to healthy auditory functioning across the global population.

Environmental noise is constantly presented in industrialized societies given the ubiquity of external sound intrusions. Examples include people talking on their cell phones, blaring music in health clubs, or the constant hum of air conditioning systems in schools and office buildings. Excess noise exposure can also induce auditory fatigue, possibly comprising a person's listening abilities. On a daily basis, people are exposed to various environmental sounds and noises within their environment, such as the sounds from traffic, construction, and industry.

To combat the undesired cacophony of annoying sounds, people are arming themselves with portable audio playback devices to drown out intrusive noise. The majority of devices providing the person with audio content do so using insert (or in-ear) earbuds. These earbuds deliver sound directly to the ear canal at high sound levels over the background noise even though the earbuds generally provide little to no ambient sound isolation. Moreover, when people wear earbuds (or headphones) to listen to music, or engage in a call using a telephone, they can effectively impair their auditory judgment and their ability to discriminate between sounds. With such devices, the person is immersed in the audio experience and generally less likely to hear warning sounds within their environment. In some cases, the user may even turn up the volume to hear their personal audio over environmental noises. It also puts them at high sound exposure risk which can potentially cause long term hearing damage.

With earbuds, personal audio reproduction levels can reach in excess of 100 dB. This is enough to exceed recommended daily sound exposure levels in less than a minute and to cause permanent acoustic trauma. Furthermore, rising population densities have continually increased sound levels in society. According to researchers, 40% of the European community is continuously exposed to transportation noise of 55 dBA and 20% are exposed to greater than 65 dBA. This level of 65 dBA is considered by the World Health Organization to be intrusive or annoying, and as mentioned, can lead to users of personal audio devices increasing reproduction levels to compensate for ambient noise.

A need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties.

SUMMARY

Embodiments in accordance with the present invention provide a method and device for acute sound detection and reproduction.

In a first embodiment, an earpiece can include an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change. The acute sound can be reproduced within the ear canal via the ECR responsive to detecting the acute sound.

The processor can pass (transmit) sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal. In one arrangement, the processor can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal. In one arrangement, the processor can measure an external ambient sound level (xASL) of the ambient sound with the ASM and subtract an attenuation level of the earpiece from the xASL to estimate the internal ambient sound level (iASL) within the ear canal.

The earpiece can further include an Ear Canal Microphone (ECM) to measure an ear canal sound level (ECL) within the ear canal. In this configuration, the processor can estimate the internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL. For instance, the processor can measure a voltage level of the audio content sent to the ECR, and apply a transfer function of the ECR to convert the voltage level to the ACL. The processor can be located external to the earpiece on a portable computing device.

In a second embodiment, an earpiece can comprise an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal, an audio interface operatively coupled to the processor to receive audio content, and a processor operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change, adjust an audio content level (ACL) of the audio content delivered to the ear canal, and reproduce the acute sound within the ear canal via the ECR responsive to detecting the acute sound and based on the ACL.

The audio interface can receive the audio content from at least one among a portable music player, a cell phone, and

a portable communication device. During operation, the processor can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal. In one arrangement, the processor can mute the audio content and pass the acute sound to the ECR for reproducing the acute sound within the ear canal. In another arrangement, the processor can amplify the acute sound with respect to the audio content level (ACL).

In a third embodiment, a method for acute sound detection and reproduction can include the steps of measuring an ambient sound level (xASL) of ambient sound external to an ear canal at least partially occluded by the earpiece, monitoring a change in the xASL for detecting an acute sound, and reproducing the acute sound within the ear canal responsive to detecting the acute sound. The reproducing can include enhancing the acute sound over the ambient sound. The step of reproducing can produce sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal.

The method can further include receiving audio content from an audio interface that is directed to the ear canal, and maintaining an approximately constant ratio between a level of the audio content (ACL) and a level of an internal ambient sound level (iASL) measured within the ear canal. The ACL can be determined by measuring a voltage level of the audio content sent to the ECR, and applying a transfer function of the ECR to convert the voltage level to the ACL. The method can further include measuring an Ear Canal Level (ECL) within the ear canal, and subtracting the ACL from the ECL to estimate the iASL. The iASL can be estimated by subtracting an attenuation level of the earpiece from the xASL.

In a fourth embodiment, a method for acute sound detection and reproduction suitable for use with an earpiece can include the steps of measuring an external ambient sound level (xASL) in an ear canal at least partially occluded by the earpiece, monitoring a change in the xASL for detecting an acute sound, estimating a proximity of the acute sound, and reproducing the acute sound within the ear canal responsive to detecting the acute sound based on the proximity. The step of estimating a proximity can include performing a cross correlation analysis between at least two microphones, identifying a peak in the cross correlation and an associated time lag, and determining the direction from the associated time lag. The method can further include identifying whether the acute sound is a vocal signal produced by a user operating the earpiece or a sound source external from the user.

In a fifth embodiment, a method for acute sound detection and reproduction suitable for use with an earpiece can include measuring an external ambient sound level (xASL) due to ambient sound outside of an ear canal at least partially occluded by the earpiece, measuring an internal ambient sound level (iASL) due to residual ambient sound within the ear canal at least partially occluded by the earpiece, monitoring a high frequency change between the xASL and the iASL with respect to a low frequency change between the xASL and the iASL for detecting an acute sound, and reproducing the xASL within the ear canal responsive to detecting the high frequency change. The method can further include determining a proximity of a sound source producing the acute sound.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

FIG. **2** is a block diagram of the earpiece in accordance with an exemplary embodiment;

FIG. **3** is a flowchart of a method for acute sound detection in accordance with an exemplary embodiment;

FIG. **4** is a more detailed approach to the method of FIG. **3** in accordance with an exemplary embodiment;

FIG. **5** is a flowchart of a method for acute sound source proximity in accordance with an exemplary embodiment;

FIG. **6** is a flowchart of a method for binaural analysis in accordance with an exemplary embodiment;

FIG. **7** is a flowchart of a method for logic control in accordance with an exemplary embodiment;

FIG. **8** is a flowchart of a method for estimating background noise level in accordance with an exemplary embodiment;

FIG. **9** is a flowchart of a method for maintaining constant audio content level (ACL) to internal ambient sound level (iASL) in accordance with an exemplary embodiment; and

FIG. **10** is a flowchart of a method for adjusting audio content gain in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

The following description of at least one exemplary embodiment is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

Processes, techniques, apparatus, and materials as known by one of ordinary skill in the relevant art may not be discussed in detail but are intended to be part of the enabling description where appropriate, for example the fabrication and use of transducers. Additionally in at least one exemplary embodiment the sampling rate of the transducers can be varied to pick up pulses of sound, for example less than 50 milliseconds.

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.

Note 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 for following figures.

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.

At least one exemplary embodiment of the invention is directed to an earpiece for ambient sound monitoring and warning detection. Reference is made to FIG. **1** in which an earpiece device, generally indicated as earpiece **100**, is constructed in accordance with at least one exemplary embodiment of the invention. As illustrated, earpiece **100** depicts an electro-acoustical assembly **113** for an in-the-ear acoustic assembly, as it would typically be placed in the ear canal **131** of a user **135**. The earpiece **100** can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, open-fit device, or any other suitable earpiece type. The earpiece **100** can be partially or fully occluded in the ear canal, and is suitable for use with users having healthy or abnormal auditory functioning.

Earpiece **100** includes an Ambient Sound Microphone (ASM) **111** to capture ambient sound, an Ear Canal Receiver (ECR) **125** to deliver audio to an ear canal **131**, and an Ear Canal Microphone (ECM) **123** to assess a sound exposure level within the ear canal. The earpiece **100** can partially or fully occlude the ear canal **131** to provide various degrees of

acoustic isolation. The assembly is designed to be inserted into the user's ear canal **131**, and to form an acoustic seal with the walls **129** of the ear canal at a location **127** between the entrance **117** to the ear canal and the tympanic membrane (or ear drum) **133**. Such a seal is typically achieved by means of a soft and compliant housing of assembly **113**. Such a seal is pertinent to the performance of the system in that it creates a closed cavity **131** of approximately 5 cc between the in-ear assembly **113** and the tympanic membrane **133**. As a result of this seal, the ECR (speaker) **125** is able 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 **133** resulting from the sound field at the entrance to the ear canal. This seal is also the basis for the sound isolating performance of the electro-acoustic assembly **113**.

Located adjacent to the ECR **125**, is the ECM **123**, which is acoustically coupled to the (closed) ear canal cavity **131**. One of its functions is that of measuring the sound pressure level in the ear canal cavity **131** 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 itself and the ECR. The ASM **111** is housed in an assembly **113** and monitors sound pressure at the entrance to the occluded or partially occluded ear canal. All transducers shown can receive or transmit audio signals to a processor **121** that undertakes audio signal processing and provides a transceiver for audio via the wired or wireless communication path **119**.

Referring to FIG. **2**, a block diagram of the earpiece **100** in accordance with an exemplary embodiment is shown. As illustrated, the earpiece **100** can include a processor **206** operatively coupled to the ASM **111**, ECR **125**, and ECM **123** via one or more Analog to Digital Converters (ADC) **202** and Digital to Analog Converters (DAC) **203**. The processor **206** can monitor the ambient sound captured by the ASM **111** for acute sounds in the environment, such as an abrupt high energy sound corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., "help", "stop", "police", etc.), or specific noise type (e.g., breaking glass, gunshot, etc.). The processor **206** can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated storage memory **208** such as Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device **100**. The memory **208** can store program instructions for execution on the processor **206** as well as captured audio processing data.

The earpiece **100** can include an audio interface **212** operatively coupled to the processor **206** to receive audio content, for example from a media player or cell phone, and deliver the audio content to the processor **206**. The processor **206** responsive to detecting acute sounds can adjust the audio content and pass the acute sounds directly to the ear canal. For instance, the processor can lower a volume of the audio content responsive to detecting an acute sound for transmitting the acute sound to the ear canal. The processor **206** 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.

The earpiece **100** can further include a transceiver **204** 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 **204** can also provide support for dynamic downloading over-the-air to the earpiece **100**. It should be noted also that next generation access technologies can also be applied to the present disclosure.

The power supply **210** can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece **100** and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply **210** 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.

The earpiece **100** can further represent 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 **100** can be reused in different form factors for the master and slave devices.

FIG. **3** is a flowchart of a method **300** for acute sound detection and reproduction in accordance with an exemplary embodiment. The method **300** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **300**, reference will be made to components of FIG. **2**, although it is understood that the method **300** can be implemented in any other manner using other suitable components. The method **300** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

The method **300** can start in a state wherein the earpiece **100** has been inserted and powered on. As shown in step **302**, the earpiece **100** can monitor the environment for ambient sounds received at the ASM **111**. Ambient sounds correspond to sounds within the environment such as the sound of traffic noise, street noise, conversation babble, or any other acoustic sound. Ambient sounds can also correspond to industrial sounds present in an industrial setting, such as factory noise, lifting vehicles, automobiles, and robots to name a few.

Although the earpiece **100** when inserted in the ear can partially occlude the ear canal, the earpiece **100** may not completely attenuate the ambient sound. During the monitoring of ambient sounds in the environment, the earpiece **100** also monitors ear canal levels via the ECM **123** as shown in step **304**. The passive aspect of the physical earpiece **100**, due to the mechanical and sealing properties, can provide upwards of a 22-26 dB noise reduction. However, portions of ambient sounds higher than 26 dB can still pass through the earpiece **100** into the ear canal. For instance, high energy low frequency sounds are not completely attenuated. Accordingly, residual sound may be resident in the ear canal and heard by the user.

Sound within the ear canal **131** can also be provided via the audio interface **212**. The audio interface **212** can receive the audio content from at least one among a portable music player, a cell phone, and a portable communication device. The audio interface **212** responsive to user input can direct sound to the ECR **125**. For instance, a user can elect to play music through the earpiece **100** which can be audibly presented to the ear canal **131** for listening. The user can also elect to receive voice communications (e.g., cell phone, voice mail, messaging) via the earpiece **100**. For instance, the user can receive audio content for voice mail or a phone call directed to the ear canal via the ECR **125**. As shown in

step **304**, the earpiece **100** an monitor ear canal levels due to ambient sound and user selected sound via the ECM **123**.

If at step **306**, audio is playing (e.g., music, cell phone, etc.), the earpiece **100** adjusts a sound level of the audio based on the ambient sound to maintain a constant signal to noise ratio with respect to the ear canal level at step **308**. For instance, the processor **206** can selectively amplify or attenuate audio content received from the audio interface **212** before it is delivered to the ECR **125**. The processor **206** estimates a background noise level from the ambient sound received at the ASM **111**, and adjusts the audio level of delivered audio content (e.g., music, cell phone audio) to maintain a constant signal (e.g., audio content) to noise level (e.g., ambient sound). By way of example, if the background noise level increases due to traffic sounds, the earpiece **100** automatically increases the volume of the audio content. Similarly, if the background noise level decreases, the earpiece **100** automatically decreases the volume of the audio content. The processor **206** can track variations on the ambient sound level to adjust the audio content level.

If at step **310**, an acute sound is detected within the ambient sound, the earpiece **100** activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ECR **125**. The processor **206** permits the ambient sound to pass through the ECR **125** to the ear canal **131** directly for example by replicating the ambient sound external to the ear canal within the ear canal. This is important if the acute sound corresponds to an on-set for a warning sound such as a bell, a car, or an object. In such regard, the ambient sound containing the acute sound is presented directly to the ear canal in an original form. Although, the earpiece **100** inherently provides attenuation due to the physical and mechanical aspects of the earpiece and its sealing properties, the processor **206** can reproduce the ambient sound within the ear canal **131** at an original amplitude level and frequency content to provide "transparency". For instance, the processor **206** measures and applies a transfer function of the ear canal to the passed ambient sound signal to provide an accurate reproduction of the ambient sound within the ear canal.

In one embodiment, the earpiece **100** looks for temporal and spectral characteristics in the ambient sound for detecting acute sounds. For instance, as will be explained ahead, the processor **206** looks for an abrupt change in the Sound Pressure Level (SPL) of an ambient sound across a small time period. The processor **206** can also detect abrupt magnitude changes across frequency sub-bands (e.g. filter-bank, FFT, etc.). Notably, the processor **206** can search for on-sets (e.g., fast rising amplitude wave-front) of an acute sound or other abrupt feature characteristics without initially attempting to initially identify or recognize the sound source. That is, the processor **206** is actively listening for a presence of acute sounds before identifying the type of sound source.

Even though the earplug inherently provides a certain attenuation level (e.g., noise reduction rating), the processor **206** in view of the ear canal level (ECL) and ambient sound level (ASL) can reproduce the ambient sound within the ear canal to allow the user to make an informed decision with regard to the acute sound. The ECL corresponds to all sounds within the ear canal and includes the internal ambient sound level (iASL) resulting from residual ambient sounds through the earpiece and the audio content level (ACL) resulting from the audio delivered via the audio interface **212**. Briefly, xASL is the external ambient sound external to the ear canal and the earpiece (e.g., ambient sound outside the ear canal). iASL is the residual ambient sound that remains internal in the ear canal. The following equations describe the relationship among terms:

$$iASL=xASL-NRR \quad \text{(EQ 1)}$$

$$iASL=ECL-ACL \quad \text{(EQ 2)}$$

As EQ 1 shows, the iASL is the difference between the external ambient sound (xASL) and the attenuation of the earpiece (Noise Reduction Rating) due to the physical and sealing properties of the earpiece. The processor **206** can measure an external ambient sound level (xASL) of the ambient sound with the ASM **111** and subtracts an attenuation level of the earpiece (NRR) from the xASL to estimate the internal ambient sound level (iASL) within the ear canal.

EQ 2 is an alternate, or supplemental, method for calculating the iASL as the difference between the ECL and the Audio Content Level (ACL). By way of the ECM **123**, the processor **206** can estimate an internal ambient sound level (iASL) within the ear canal by subtracting the estimated audio content sound level (ACL) from the ECL. The processor **206** measures a voltage level of the audio content sent to the ECR **125**, and applies a transfer function of the ECR **125** to convert the voltage level to the ACL.

The processor **206** evaluates the equations above to pass sound from the ASM **111** directly to the ECR **125** to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal. Further, the processor **206** can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal.

At step **314**, the earpiece **100** can estimate a proximity of the acute sound. For instance, as will be shown ahead, the processor **206** can perform a correlation analysis on at least two microphones to determine whether the sound source is internal (e.g., the user) or external (e.g., an object other than the user). At step **316**, the earpiece **100** determines whether it is the user's voice that generates the acute sound when the user speaks, or whether it is an external sound such as a vehicle approaching the user. If at step **316**, the processor **206** determines that the acute sound is a result of the user speaking, the processor **206** does not activate a pass-through mode, since this is not considered an external warning sound. The pass-through mode permits ambient sound detected at the ASM **111** to be transmitted directly to the ear canal. If however, the acute sound corresponds to an external sound source, such as an on-set of a warning sound, the earpiece at step **318** activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ECR **125**. The earpiece **100** can also present an audible notification to the user indicating that an external sound source generating the acute sound has been detected. The method **300** can proceed back to step **302** to continually monitor for acute sounds in the environment.

FIG. **4** is a detailed approach to the method **400** of FIG. **3** for an Acute-Sound Pass-Through System (ACPTS) in accordance with an exemplary embodiment. The method **400** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **400**, reference will be made to components of FIG. **2**, although it is understood that the method **400** can be implemented in any other manner using other suitable components. The method **400** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

At step **402**, the earpiece **100** captures ambient sound signals from the ASM **111**. At step **404**, the processor **206**

applies analog and discrete time signal processing to condition and compensate the ambient sound signal for the ASM **111** transducer. At step **406**, the processor **206** estimates a background noise level (BNL) as will be discussed ahead. At step **408**, the processor **206** identifies at least one peak in a data buffer storing a portion of the ambient sound signal. The processor **206** at step **410** gets a level of the peak (e.g., dBV). Block **412** presents a method for warning signal detection (e.g. car horns, klaxons). When a warning signal is detected at step **416**, the processor **206** invokes at step **418** a pass-through mode whereby the ASM signal is reproduced with the ECR **125**. Upon activating pass-through mode, the processor **206** can perform a safe level check at step **452**. If a warning signal is not detected, the method **400** proceeds to step **420**.

At step **420**, the processor **206** subtracts the estimated BNL from an SPL of the ambient sound signal to produce signal "A". A high energy level transient signal is indicative of an acute sound. At step **422**, a frequency dependent threshold is retrieved at step **424**, and subtracted from signal "A", as shown in step **422** to produce signal "B". At step **426**, the processor **206** determines if signal "B" is positive. If not, the processor **206** performs a hysterisis to determine if the acute sound has already been detected. If not, the processor at step **428** determines if an SPL of the ambient sound is greater than a signal "C" (e.g. threshold). If the SPL is greater than signal "C", the earpiece generates a user generated sound at step **434**. The signal "C" is used to ensure that the SPL between the signal and background noise is positive and greater than a predetermined amount. For instance, a low SPL threshold (e.g., "C" 40 dB) can be used as shown in step **430**, although it can adapt to different environmental conditions. The low SPL threshold provides an absolute measure to the SPL difference. At step **436**, a proximity of a sound source generating the acute sound can be estimated as will be discussed ahead. The method **400** can continue to step **432**.

Briefly, if a transient, high-level sound (or acute sound) is detected in the ambient sound signal (ASM input signal), then it is converted to a level, and its magnitude compared with the BNL is calculated. The magnitude of this resulting difference (signal "A") is compared with the threshold (see step **422**). If the value is positive, and the level of the transient is greater than a predefined threshold (see step **428**), the processor **206** invokes the optional Source Proximity Detector at step **436**, which determines if the acute sound was created by the User's voice (i.e., a user generated sound). If a user-generated sound is NOT detected, then Pass-through operation at step **438** is invoked, whereby the ambient sound signal is reproduced with the ECR **125**. If the difference signal at step **428** is not positive, or the level of the identified transient is too low, then the hysteresis is invoked at step **432**. The processor **206** decides if the pass-through was recently used at step **440** (e.g. in the last 10 ms). If pass-through mode was recently activated, then processor **206** invokes the pass-through system at step **438**; otherwise there is no pass-through of the ASM signal to the ECR as shown at step **442**. Upon activating pass-through mode, the processor **206** can perform a safe level check at step **452**.

FIG. **5** is a flowchart of a method **500** for acute sound source proximity. The method **500** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **500**, reference will be made to components of FIG. **2**, although it is understood that the method **500** can be implemented in any other manner using other suitable components. The method **500** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **5** describes a method **500** for Source Proximity Detection (SPD) to determine if the Acute sound detected was created by the User's voice operating the earpiece **100**. The SPD method **500** uses as its inputs the external ambient sound signals from left and right electro-acoustic earpiece **100** assemblies (e.g., a headphone). In some embodiments the SPD method **500** employs Ear Canal Microphone (ECM) signals from left and right earpiece **100** assemblies placed on left and right ears respectively. The processor **206** performs an electronic cross-correlation between the external ambient sound signals to determine a Pass-through or Non Pass-through operating mode. In the described embodiment whereby the cross-correlation of both the ASM and ECM signals is involved, a pass-through mode is invoked when the cross-correlation analysis for both the left and right earpiece **100** assemblies return a "Pass-through" operating mode, as determined by a logical AND unit.

For instance, at step **502** a left ASM signal from a left headset incorporating the earpiece **100** assembles is received. Simultaneously, at step **504** a right ASM signal from a right headset is received. At step **510**, the processor **206** performs a binaural cross correlation on the left ASM signal and the right ASM signal to evaluate a pass through mode **516**. At step **506** a left ECM signal from the left headset is received. At step **508**, a right ECM signal from the right headset is received. At step **514**, the processor **206** performs a binaural cross correlation on the left ECM signal and the right ECM signal to evaluate a pass through mode **518**. A pass through mode **524** is invoked if both the ASM and ECM cross correlation analysis are the same as determined in step **520**. A safe level check can be performed by processor **206** at step **522**.

FIG. **6** is a flowchart of a method **600** for binaural analysis. The method **600** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **600**, reference will be made to components of FIG. **2**, although it is understood that the method **600** can be implemented in any other manner using other suitable components. The method **600** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **6** describes a component of the SPD method **500** wherein a cross-correlation of two input audio signals **602** and **604** (e.g., left and right ASM signals) is calculated. The input signals may first be weighted using a frequency-dependant filter (e.g. an FIR-type filter) using filter coefficients **606** and filtering networks **608** and **610**. Alternatively, an interchannel cross-correlation calculated with function **612** can return a frequency-dependant correlation such as a coherence function. The absolute maximum peak of a calculated cross-correlation **614** can be subtracted from a mean (or RMS) **616** correlation, with subtractor **622**, and compared **628** with a predefined threshold **626**, to determine if the peak is significantly greater than the average correlation (i.e. a test for peakedness). Alternatively, the maxima of the peak may simply be compared with the threshold **628** without the subtraction process **622**. If the lag-time of the peak **618** is at approximately lag-sample 0, then the sound source is determined, at step **624**, as being on the interaural axis-indicative of User-generated speech, and a no-pass through mode is returned **630** (a further function described in FIG. **7** may be used to confirm that the sound source originates in the User-head, rather than external to the user-

and further confirming that the acute sound is a User-generated voice sound). The logical AND unit **632** activates the pass-through mode **636** if both criteria in the decision units **628** and **624** confirm that the absolute maxima of the peak is above a predefined threshold **626**, AND the lag of the peak is NOT at approximately lag sample zero. A safe level check may be performed by processor **206** at step **634**.

FIG. **7** is a flowchart of a method **700** for logic control. The method **700** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **700**, reference will be made to components of FIG. **2**, although it is understood that the method **700** can be implemented in any other manner using other suitable components. The method **700** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **7** describes a further component of the SPD method **500**, which is optional to confirm that the acute sound source is from a location indicative of user-generated speech; i.e. inside the head. Method steps **702-712** are similar to Method steps **502-514** of FIG. **5**. The cross-correlations of step **710** and **712** provide a time-lag of the maximum absolute peak for a pair of input signals; the ASM and ECM signals for the same headset (e.g. the ASM and ECM for the left headset). At step **714** a left lag of a peak of the left cross correlation is determined, and simultaneously, a right lag of a peak of the right cross correlation is determined at step **718**. If a lag of a respective peak is greater than zero—this indicates that the sound arrived at the ECM signal before the ASM signal. Decision step **716** determines if the lag is greater than zero for both the left and right headsets- and activates the pass-through mode **722** if so. A safe level check may be performed by processor **206** at step **720**.

FIG. **8** is a flowchart of a method **800** for estimating background sound level. The method **800** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **800**, reference will be made to components of FIG. **2**, although it is understood that the method **800** can be implemented in any other manner using other suitable components. The method **800** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, method **800** receives as its input **802** either or both the ASM signal from ASM **111** and a signal from the ECM **123**. An audio buffer **804** of the input audio signal is accumulated (e.g. 10 ms of data), which is then processed by squaring step **806** to obtain the temporal envelope. The envelope is smoothed (e.g. an FIR-type low-pass digital filter) at step **808** using a smoothing window **810** stored in data memory (e.g. a Hanning or Hamming shaped window). At step **812**, transient peaks in the input buffer can be identified and removed to determine a "steady-state" Background Noise Level (BNL). At step **814** an average BNL **816** can be obtained (similar to, or the same as, the RMS) that is frequency dependant or a single value averaged over all frequencies. If the ECM **123** is used to determine the BNL, then decision step **818** adjusts the ambient BNL estimation to provide an equivalent ear-canal BNL SPL, by deducting an Earpiece Noise Reduction Rating **828** from the BNL estimate **826**. Alternatively, if the ECM **123** is used, then the Audio Content SPL level (ACL) **822** of any reproduced Audio Content **820** is deducted from the ECM level at step **824**. The updated BNL estimate is then converted to a Sound Pressure Level (SPL) equivalent **832** (i.e. substantially equal to the SPL at the ear-drum in which the earphone device is inserted) by taking into account the sensitivity (e.g. measured in V per dB) of either the ASM **111** or ECM **123** at steps **830** and **834** respectively. The resulting BNL SPL is then combined at step **842** with the previous BNL estimate **840**, by averaging **838** a weighted previous BNL (weighted with coefficient **836**), to give a new ear-canal BNL **844**.

FIG. **9** is a flowchart of a method **900** for maintaining constant audio content level (ACL) to internal ambient sound level (iASL). The method **900** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **900**, reference will be made to components of FIG. **2**, although it is understood that the method **900** can be implemented in any other manner using other suitable components. The method **900** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **9** describes a method **900** for Constant Signal-to-Noise Ratio (CSNRS). At step **904** an input signal is captured from the ASM **111** and processed at step **910** (e.g. ADC, EQ, gain). Similarly, at step **906** an input signal from the ECM **123** is captured and processed at step **912**. The method **900** also receives as input an Audio Content signal **902**, e.g. a music audio signal from a portable Media Player or mobile-phone, which is processed with an analog and digital signal processing system as shown in step **908**. An Audio Content Level (ACL) is determined at step **914** based on an earpiece sensitivity from step **916**, and returns a dBV value.

In one exemplary embodiment, method **900** calculates a RMS value over a window (e.g. the last 100 ms). The RMS value can then be first weighted with a first weighting coefficient and then averaged with a weighted previous level estimate. The ACL is converted to an equivalent SPL value (ACL), which may use either a look-up-table or algorithm to calculate the ear-canal SPL of the signal if it was reproduced with the ECR **125**. To calculate the equivalent ear canal SPL, the sensitivity of the ear canal receiver can be factored in during processing.

At step **922** the BNL is estimated using inputs from either or both the ASM signal at step **902**, and/or the ECM signal at step **906**. The BNL may be adjusted by the earpiece noise reduction rating **924**. These signals are selected using the BNL input switch at step **918**, which may be controlled automatically or with a specific user-generated manual operation at step **926**. The Ear-Canal SNR is calculated at step **920** by differencing the ACL from step **914** and the BNL from step **922** and the resulting SNR **930** is passed to the method step **932** for AGC coefficient calculation. The AGC coefficient calculation **932** calculates gains for the Audio Content signal and ASM signal from the Automatic Gain Control steps **928** and **936** (for the Audio Content and ASM signals, respectively). AGC coefficient calculation **932** may use a default preferred SNR **938** or a user-preferred SNR **934** in its calculation. After the ASM signal and Audio content signal have been processed by the AGCs **928** and **936**, the two signals are mixed at step **940**.

At step **942**, a safe-level check determines if the resulting mixed signal is too high, if it were reproduced with the ECR **125** as shown in block **944**. The safe-level check can use information regarding the user's listening history to determine if the user's sound exposure is such that it may cause a temporary or a permanent hearing threshold shift. If such high levels are measured, then the safe-level check reduces the signal level of the mixed signals via a feedback path to step **940**. The resulting audio signal generated after step **942** is then reproduced with the ECR **125**.

FIG. **10** is a flowchart of a method **950** for maintaining a constant signal to noise ratio based on automatic gain control (AGC). The method **950** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **950**, reference will be made to components of FIG. **2**, although it is understood that the method **950** can be implemented in any other manner using other suitable components. The method **950** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Method **950** describes calculation of AGC coefficients. The method **950** receives as its inputs an Ear Canal SNR **952** and a target SNR **960** to provide a SNR mismatch **958**. The target SNR **964** is chosen from a pre-defined SNR **954**, sorted in computer memory or a manually defined SNR **956**. At step **958**, a difference is calculated between the actual ear-canal SNR and the target SNR to produce the mismatch **962**. The mismatch level **962** is smoothed over time at step **968**, which uses a previous mismatch **970** that is weighted using single or multiple weighting coefficients **966**, to give a new time-smoothed SNR mismatch **974**. Depending on the magnitude of this mismatch, various operating modes **972**, **978** can be invoked, for example, as described by the AGC decision module **976** (step **932** in FIG. **9**).

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. 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.

We claim:

1. A system, comprising:
a microphone;
a memory that stores instructions;
a processor that executes the instructions to perform operations, the operations comprising:
estimating a background noise level (BNL) of an environment;
identifying a peak level (SPL) stored in an audio buffer including an external ambient sound, wherein the SPL is of the external ambient sound;
generating a SPL-BNL difference by subtracting the BNL from the SPL identified in the audio buffer;
comparing the SPL-BNL difference to a threshold to determine if a sound has been detected; and
detecting the sound when the SPL-BNL difference is greater than the threshold.

2. The system of claim **1**, wherein the operations further comprise storing the external ambient sound in the audio buffer.

3. The system of claim **1**, wherein the operations further comprise detecting a proximity of the sound to determine whether the sound is generated by a user.

4. The system of claim **3**, wherein the operations further comprise providing the sound to an ear canal receiver if the sound was not generated by the user.

5. The system of claim **4**, wherein the operations further comprise providing the sound to the ear canal receiver by replicating the external ambient sound to an ear canal of the user via the ear canal receiver.

6. The system of claim **5**, wherein the operations further comprise applying a transfer function of the ear canal to the external ambient sound to provide a reproduction of the external ambient sound within the ear canal.

7. The system of claim **3**, wherein the operations further comprise not providing the sound to an ear canal receiver if the sound was generated by the user.

8. The system of claim **1**, wherein the operations further comprise detecting the sound based on a temporal and spectral characteristic detected in the external ambient sound.

9. The system of claim **1**, wherein the operations further comprise performing a correlation analysis on the microphone and at least one other microphone.

10. The system of claim **1**, wherein the operations further comprise determining a time lag in the sound between the microphone and at least one other microphone.

11. The system of claim **10**, wherein the operations further comprise determining a direction of the sound based on the time lag.

12. A method, comprising:
estimating a background noise level (BNL) of an environment;
identifying a peak level (SPL) stored in an audio buffer including an external ambient sound, wherein the SPL is of the external ambient sound;
generating, by utilizing instructions from a memory that are executed by a processor, a SPL-BNL difference by subtracting the BNL from the SPL identified in the audio buffer;
comparing the SPL-BNL difference to a threshold to determine if a sound has been detected; and
detecting the sound when the SPL-BNL difference is greater than the threshold.

13. The method of claim **12**, further comprising determining whether the sound is from a location indicative of user-generated speech.

14. The method of claim **12**, further comprising determining a time lag of a maximum absolute peak for a pair of input signals comprising an ambient sound microphone signal and an ear canal microphone signal associated with a headset.

15. The method of claim **12**, further comprising determining a left lag of a peak of a left cross correlation of a left headset and a right lag of a peak of a right cross correlation of a right headset.

16. The method of claim **15**, further comprising determining if the left lag and the right lag are greater than zero.

17. The method of claim **16**, further comprising activating a pass-through mode if the left lag and the right lag are determined to be greater than zero.

18. The method of claim **12**, further comprising applying analog and discrete time signal processing to condition the external ambient sound.

19. The method of claim **12**, further comprising estimating a proximity of the acute sound.

20. A device, comprising:
a memory that stores instructions;
a processor that executes the instructions to perform operations, the operations comprising:
estimating a background noise level (BNL) of an environment;
identifying a peak level (SPL) stored in an audio buffer including an external ambient sound, wherein the SPL is of the external ambient sound;

generating a SPL-BNL difference by subtracting the BNL from the SPL identified in the audio buffer;

comparing the SPL-BNL difference to a threshold to determine if a sound has been detected;

detecting the sound when the SPL-BNL difference is greater than the threshold; and

passing the sound to an ear canal receiver if the sound is determined to not be user-generated.

* * * * *

Exhibit PA-E

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(54) External ear canal voice detection

(57) Ear-level full duplex audio communication systems each include one or two ear attachment devices, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, that wirelessly communicates 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, 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. Each ear attachment device includes a voice operated exchange (VOX), housed within the device, to preserve energy and hence, maximize the period between battery replacement or recharges. The VOX also gates various sounds detected by the system to control possible echoes and ringing.

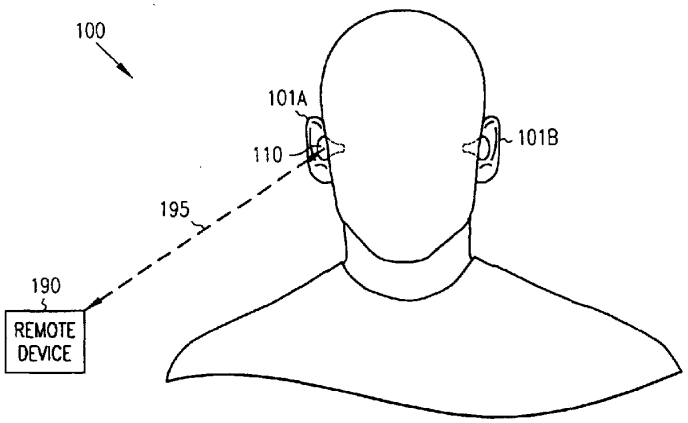


FIG. 1

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Exhibit PA-E

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 receive 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 white 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] Ear-level full duplex audio communication systems each include one or two ear attachment 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.

[0009] In one embodiment of the invention, a system includes an earmold configured for use as an ITE device. The earmold houses a microphone for use in an ear canal, a processor, and a wireless transmitter. The microphone receives an occluded sound from about the ear canal. The processor processes the occluded sound. The wireless transmitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0010] In one 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 about the ear canal. The BTE module includes a processor and a wireless transmitter. The processor processes the occluded sound. The wireless transmitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0011] In one 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 about 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.

[0012] In one embodiment, an occluded sound is detected using a microphone placed in an ear canal. An incoming radio signal representing a remote sound from a remote device is also detected. A voice operated exchange (VOX) housed within an ear-level communication device is started 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 includes one of an ITE device and a BTE device.

When the occluded sound is detected while the incoming radio signal is not detected, the occluded sound is gated on, and remote sound is gated off. When the incoming radio signal is detected while the occluded sound is not detected, the remote sound is gated on, and the occluded sound is gated off.

[0013] Preferred embodiments of the present invention will now be described by way of example only and with reference to the accompanying drawings, in which:

PIG. 1 is an illustration of an embodiment of an ear-level communication system and portions 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 (TTC) device used as the single ear-level device of 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.

[0014] 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.

[0015] 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.

[0016] 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, 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.

[0017] 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.

[0018] 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.

[0019] 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.

[0020] 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.

[0021] 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.

[0022] 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.

[0023] 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.

[0024] 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.

[0025] 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 size is therefore generally smaller than that of ITE deviee 310A.

[0026] 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.

[0027] 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.

[0028] 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.

[0029] 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.

[0030] 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.

[0031] 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 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.

[0032] 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.

[0033] 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.

[0034] 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.

[0035] 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.

[0036] 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.

[0037] 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.

[0038] 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 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.

[0039] 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.

[0040] 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.

[0041] 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.

[0042] 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.

[0043] 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.

[0044] BTE device 610 includes a behind-the-ear-module (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.

[0045] 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.

[0046] 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.

[0047] 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.

[0048] 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.

[0049] 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.

[0050] 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.

[0051] 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. 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.

[0052] 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.

[0053] 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.

[0054] 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.

[0055] 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.

[0056] 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.

[0057] 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.

[0058] 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.

[0059] 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 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.

[0060] 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.

[0061] It is to be understood that the above detailed description is intended to be illustrative, and not restric-

tive. 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 having a first ear canal and a second ear having a second ear canal, comprising:

   a first ear-level device including:

   a first microphone adapted for use in one of the first ear canal and the second ear canal, the first microphone receiving an occluded sound from about the one of the first ear canal and the second ear canal;
   a first processor, connected to the first microphone, to process the occluded sound;
   a wireless transmitter, connected to the first processor, to receive the processed occluded sound from the first processor and transmit a first wireless signal representing the occluded sound; and
   a first earmold housing at least the first microphone, at least a portion of the first earmold configured for placement in the one of the first ear canal and the second ear canal.

2. The system according to claim 1, wherein the first earmold houses the first microphone, the first processor, and the wireless transmitter, and wherein the first earmold is configured for use as an in-the-ear (ITE) device.

3. The system according to claim 1, wherein the first ear level device comprises an ITE module including the first microphone and a behind-the-ear (BTE) module attached to the ITE module, the BTE module including the first processor and the wireless transmitter.

4. The system according to any of the preceding claims, further comprising a voice operated exchange (VOX) circuit connected to the first processor.

5. The system according to claim 4, wherein the VOX circuit comprises an occluded sound gating module, coupled to the first microphone, to blank or substantially attenuate the occluded sound.

6. The system according to claim 1, further comprising:

   a wireless receiver, connected to the first processor, to receive a second wireless signal representing a remote sound; and
   a speaker connected to the first processor, the speaker adapted for use in the one of the first ear canal and the second ear canal,

   wherein:

   the first processor further processes the remote sound, and the speaker is adapted to transmit the processed remote sound to the one of the first ear canal and the second ear canal; and
   the first earmold further houses at least the speaker.

7. The system according to claim 6, wherein the first microphone and the speaker include one common device receiving the occluded sound and transmitting the processed remote sound.

8. The system according to any of claims 6 and 7, further comprising a second microphone connected to the first processor, the second microphone adapted to receive an ambient sound from outside the one of the first ear canal and the second ear canal, and wherein:

   the first processor further processes the ambient sound; and
   the speaker is configured to transmit the processed remote and ambient sounds to the one of the first ear canal and the second ear canal.

9. The system according to claim 8, further comprising a voice operated exchange (VOX) circuit connected to the first processor.

10. The system according to claim 9, wherein the VOX circuit comprises an occluded sound gating module, coupled to the first microphone, to blank or substantially attenuate the occluded sound.

11. The system according to any of claims 9 and 10, wherein the VOX circuit comprises a remote sound gating module, coupled to the wireless transceiver, to blank or substantially attenuate the remote sound.

12. The system according to any of claims 9 -11, wherein the VOX circuit comprises an ambient sound gating module, coupled to the first microphone, to blank or substantially attenuate the ambient sound.

13. The system according to any of the preceding

claims, wherein the first processor comprises a speech recognition module.

14. The system according to any of claims 6 -13, wherein the first earmold houses the first microphone, the first processor, the wireless transmitter, the wireless receiver, and the speaker, and wherein the first carmold is configured for use as an in-the-ear (ITE) device.

15. The system according to claim 14, wherein the first earmold is configured for use as an in-the-canal (ITC) device.

16. The system according to claim 15, wherein the first earmold is configured for use as a completely-in-the-canal (CIC) device.

17. The system according to any of claims 6 - 13, wherein the first ear level device comprises an ITE module including the first microphone and the speaker and a behind-the-ear (BTE) module attached to the ITE module and including the second microphone, the first processor, the wireless transmitter, and the wireless receiver.

18. The system of claim 17, wherein the BTE module comprises a voice operated exchange (VOX) circuit.

19. The system of claim 17, wherein the BTE module comprises a rechargeable battery.

20. The system according to any of claims 6 - 19, further comprising a remote device communicatively coupled to the wireless transmitter and the wireless receiver via a telemetry link providing for simultaneous bi-directional communication between the first ear-level device and the remote device.

21. The system according to claim 20, wherein the remote device comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

22. The system according to claim 1, wherein the first microphone is adapted for use in the first ear canal and receives the occluded sound from about the first ear canal, wherein the at least the portion of the first earmold is configured for placement in the first ear canal, and further comprising a second ear-level device including:

a wireless receiver to receive a second wireless signal representing a remote sound;
a second processor, connected to the wireless receiver, to convert the second wireless signal to the remote sound;
a speaker connected to the second processor, the speaker adapted for use in the second ear canal to transmit the remote sound to the second ear canal; 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.

23. The system according to claim 22, wherein the second ear-level device further comprises a second microphone connected to the second processor, the second microphone adapted to receive an ambient sound from outside the second ear canal, and wherein:

the second processor further processes the ambient sound; and
the speaker is configured to transmit the remote sound and the processed ambient sound to the second ear canal.

24. The system according to any of claims 22 and 23, wherein the first ear-level device further comprises a first voice operated exchange (VOX) circuit connected to the first processor, and the second ear-level device further comprises a second VOX circuit connected to the second processor,

25. The system according to any of claims 22 - 24, wherein the first processor and the second processor each comprise a speech recognition module.

26. The system according to any of claims 22 - 25, wherein the first and second ear-level devices are each an behind-the-ear (BTE) device.

27. The system according to any of claims 22 - 25, wherein the first and second ear-level devices are each an in-the-ear (ITE) device.

28. The system according to claim 27, wherein the first and second ear-level devices are each an in-the-canal (ITC) device.

29. The system according to claim 28, wherein the first and second ear-level devices are each an completely-in-the-canal (CIC) device.

30. The system according to any of claims 22 - 29, further comprising a remote device communicatively coupled to the first and second ear-level devices.

31. The system according to claim 30, wherein the remote device comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

32. 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 the 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 (BTB) 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.

33. The method according to claim 32, 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.

34. The method according to any of claims 32 and 33, 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.

35. The method according to any of claims 32 - 34, 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.

36. The method according to claim 35, 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.

37. The method according to claim 35, 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.

38. The method according to claim 35, 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.

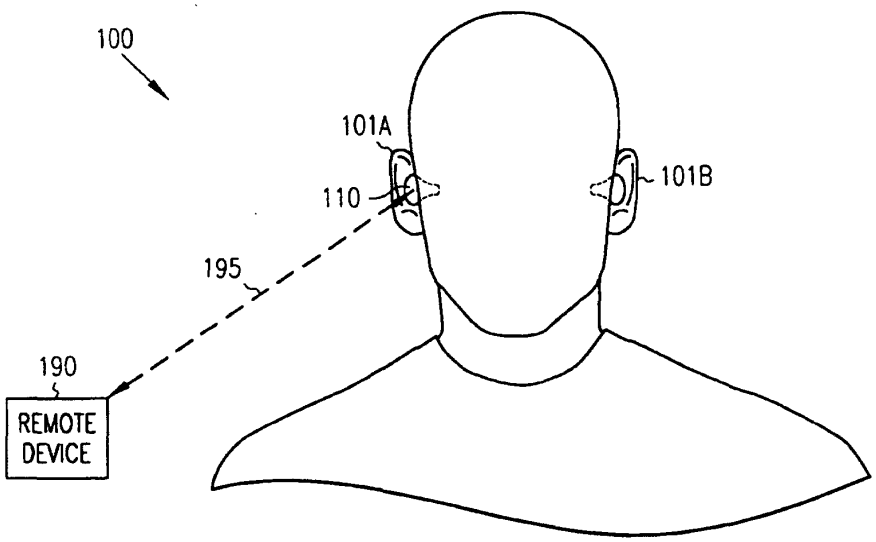

100
101A
101B
110
195
190
REMOTE DEVICE

FIG. 1

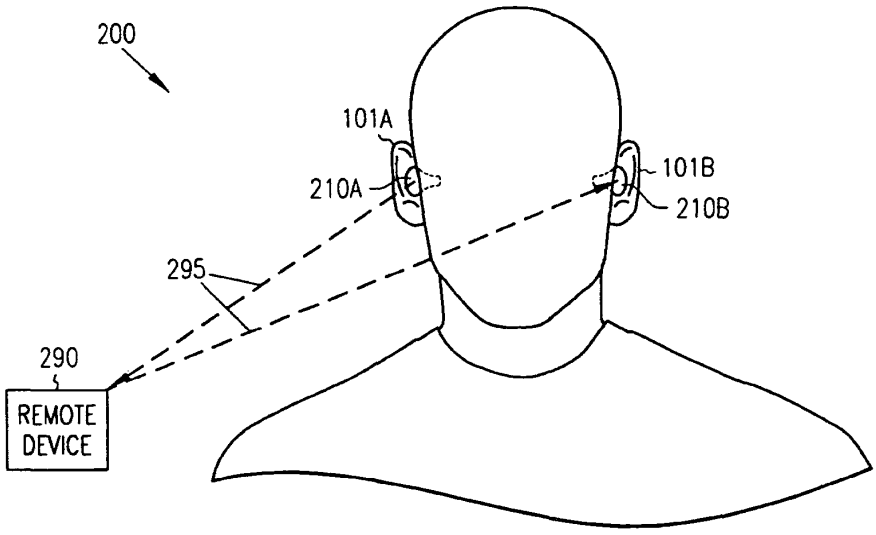

200
101A
101B
210A
210B
295
290
REMOTE DEVICE

FIG. 2

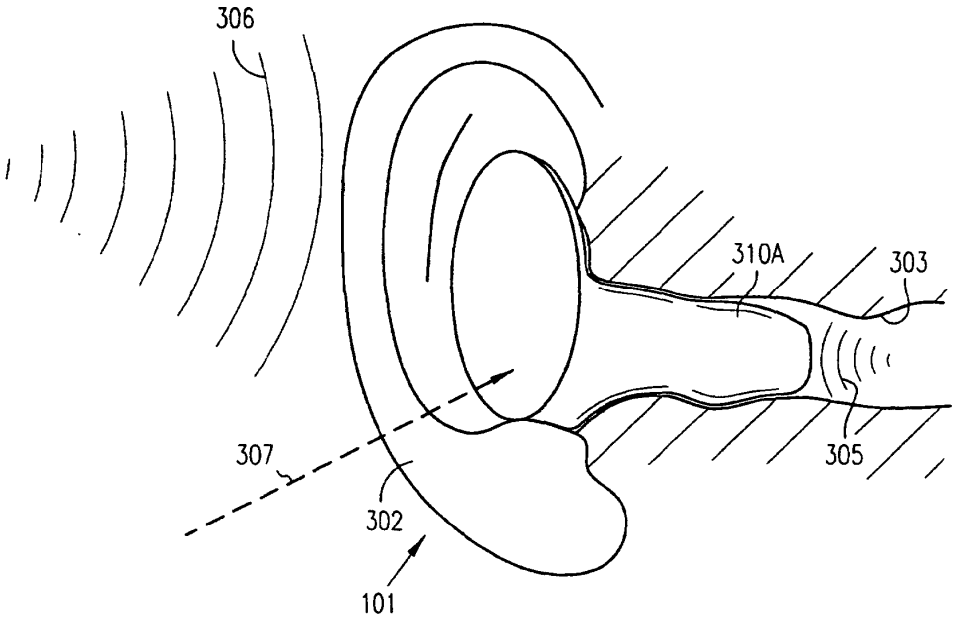


FIG. 3A

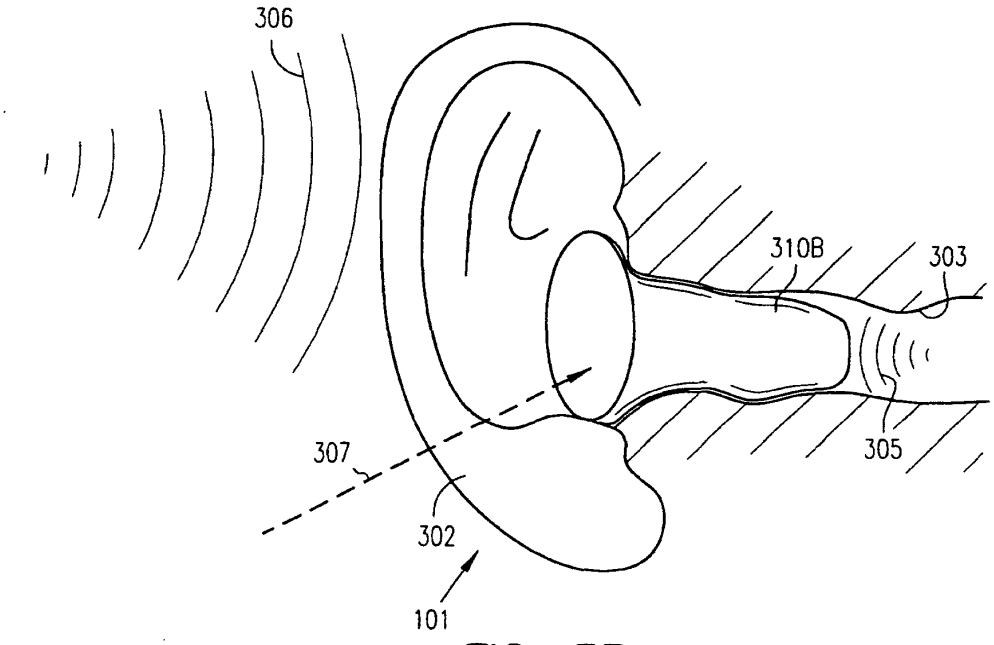


FIG. 3B

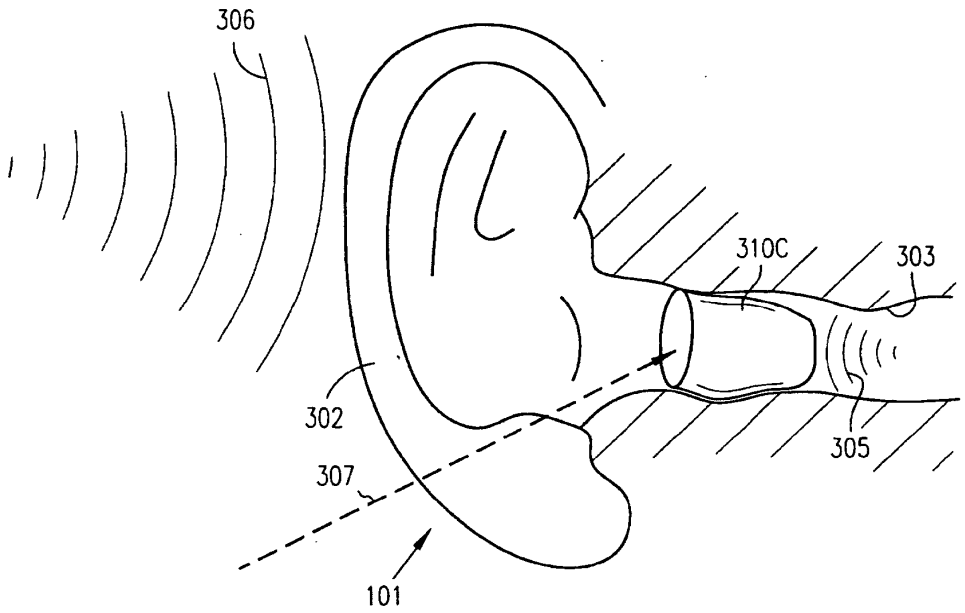


FIG. 3C

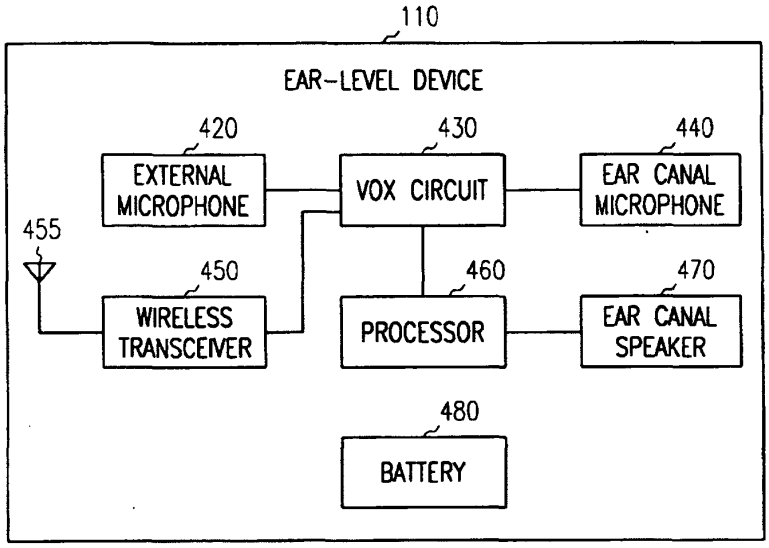

110
EAR-LEVEL DEVICE
420
EXTERNAL MICROPHONE
430
VOX CIRCUIT
440
EAR CANAL MICROPHONE
455
450
WIRELESS TRANSCEIVER
460
PROCESSOR
470
EAR CANAL SPEAKER
480
BATTERY

FIG. 4A

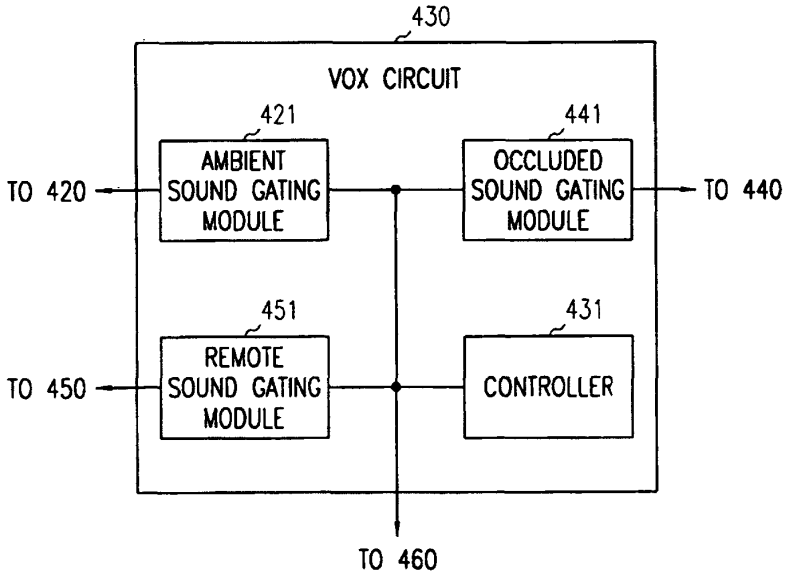

430
VOX CIRCUIT
421
AMBIENT SOUND GATING MODULE
441
OCCLUDED SOUND GATING MODULE
TO 420
TO 440
451
REMOTE SOUND GATING MODULE
431
CONTROLLER
TO 450
TO 460

FIG. 4B

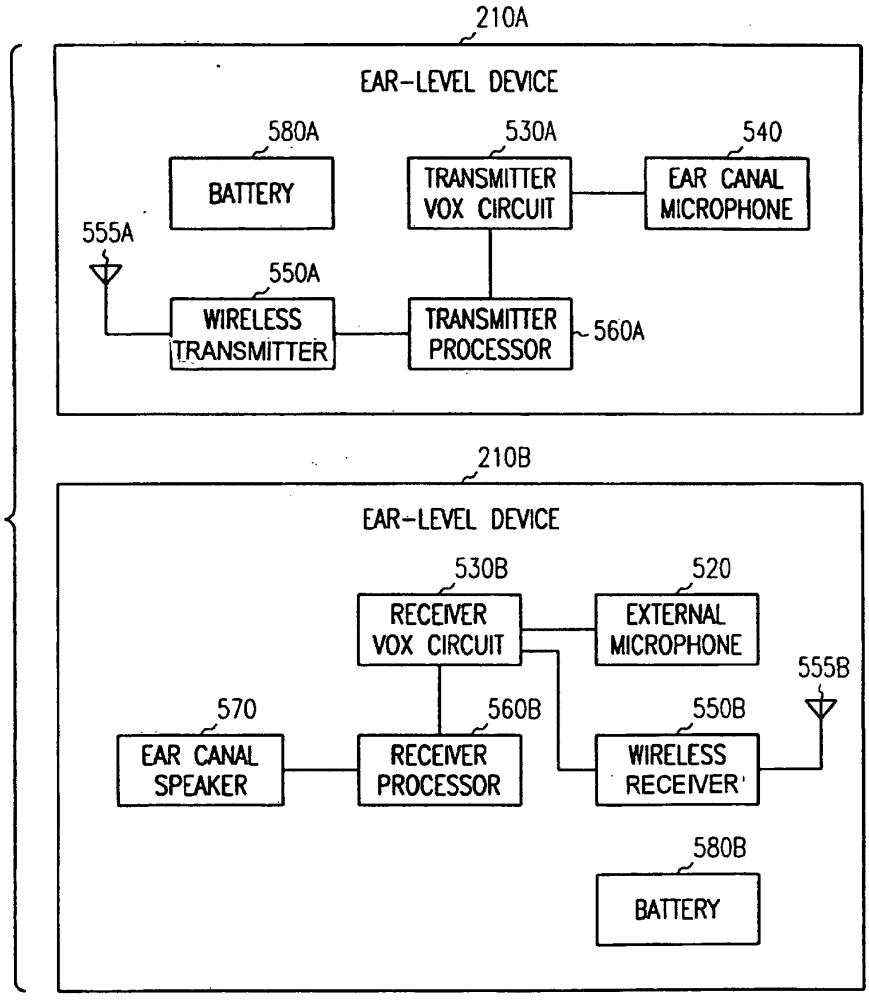

210A
EAR-LEVEL DEVICE
580A
BATTERY
530A
TRANSMITTER VOX CIRCUIT
540
EAR CANAL MICROPHONE
555A
550A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
560A
210B
EAR-LEVEL DEVICE
530B
RECEIVER VOX CIRCUIT
520
EXTERNAL MICROPHONE
555B
570
EAR CANAL SPEAKER
560B
RECEIVER PROCESSOR
550B
WIRELESS RECEIVER
580B
BATTERY

FIG. 5

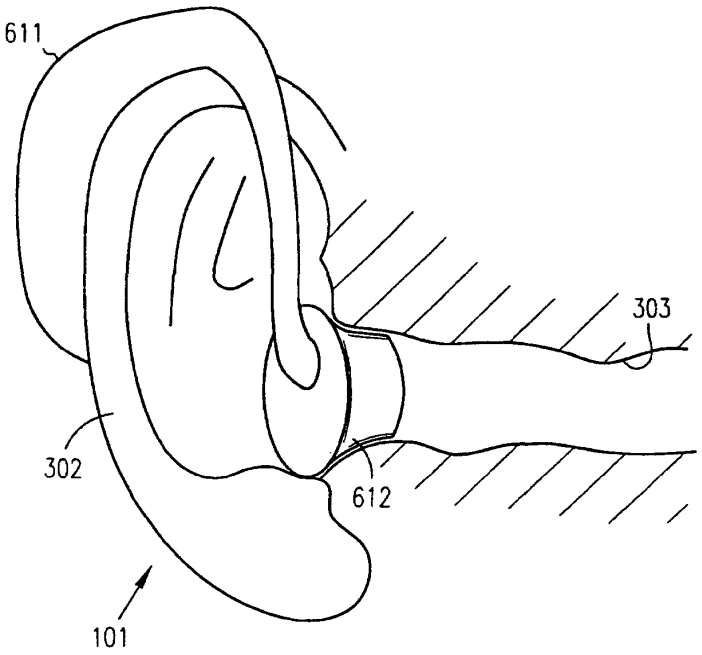

611
303
302
612
101

FIG. 6

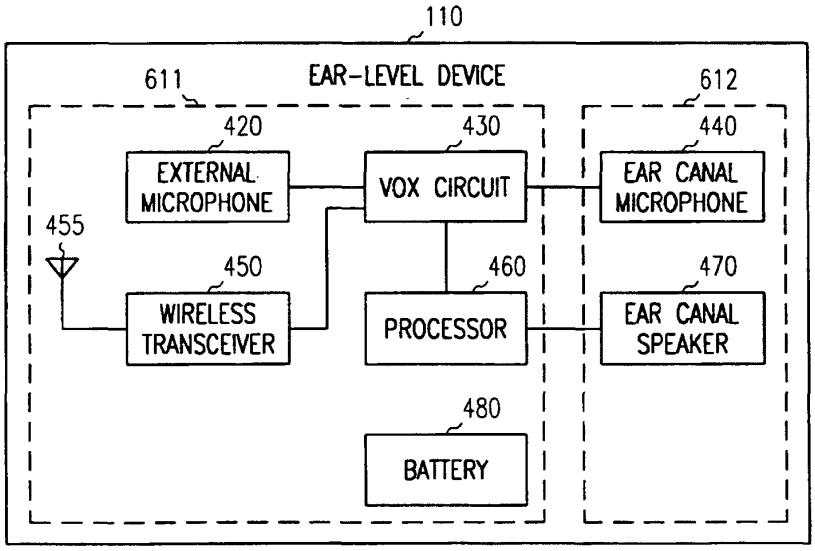

110
611
EAR-LEVEL DEVICE
612
420
430
440
EXTERNAL MICROPHONE
VOX CIRCUIT
EAR CANAL MICROPHONE
455
450
460
470
WIRELESS TRANSCEIVER
PROCESSOR
EAR CANAL SPEAKER
480
BATTERY

FIG. 7

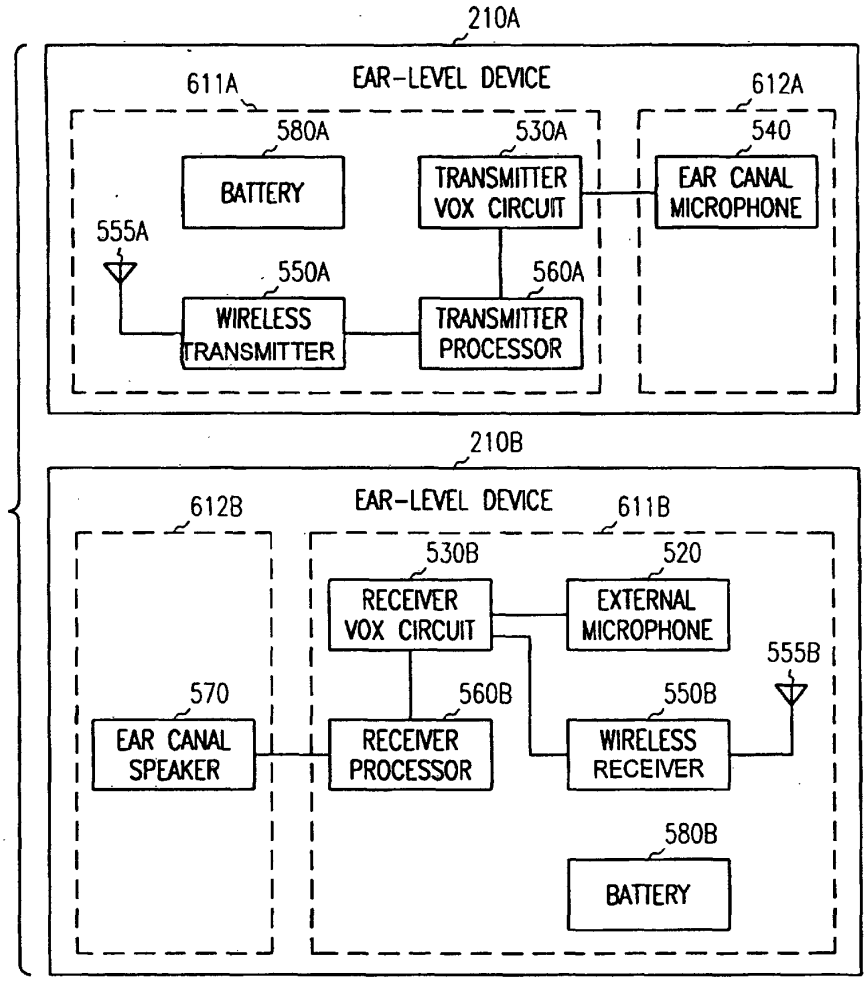

210A
611A
EAR-LEVEL DEVICE
612A
580A
530A
540
BATTERY
TRANSMITTER VOX CIRCUIT
EAR CANAL MICROPHONE
555A
550A
560A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
210B
612B
EAR-LEVEL DEVICE
611B
530B
520
RECEIVER VOX CIRCUIT
EXTERNAL MICROPHONE
555B
570
560B
550B
EAR CANAL SPEAKER
RECEIVER PROCESSOR
WIRELESS RECEIVER
580B
BATTERY

FIG. 8

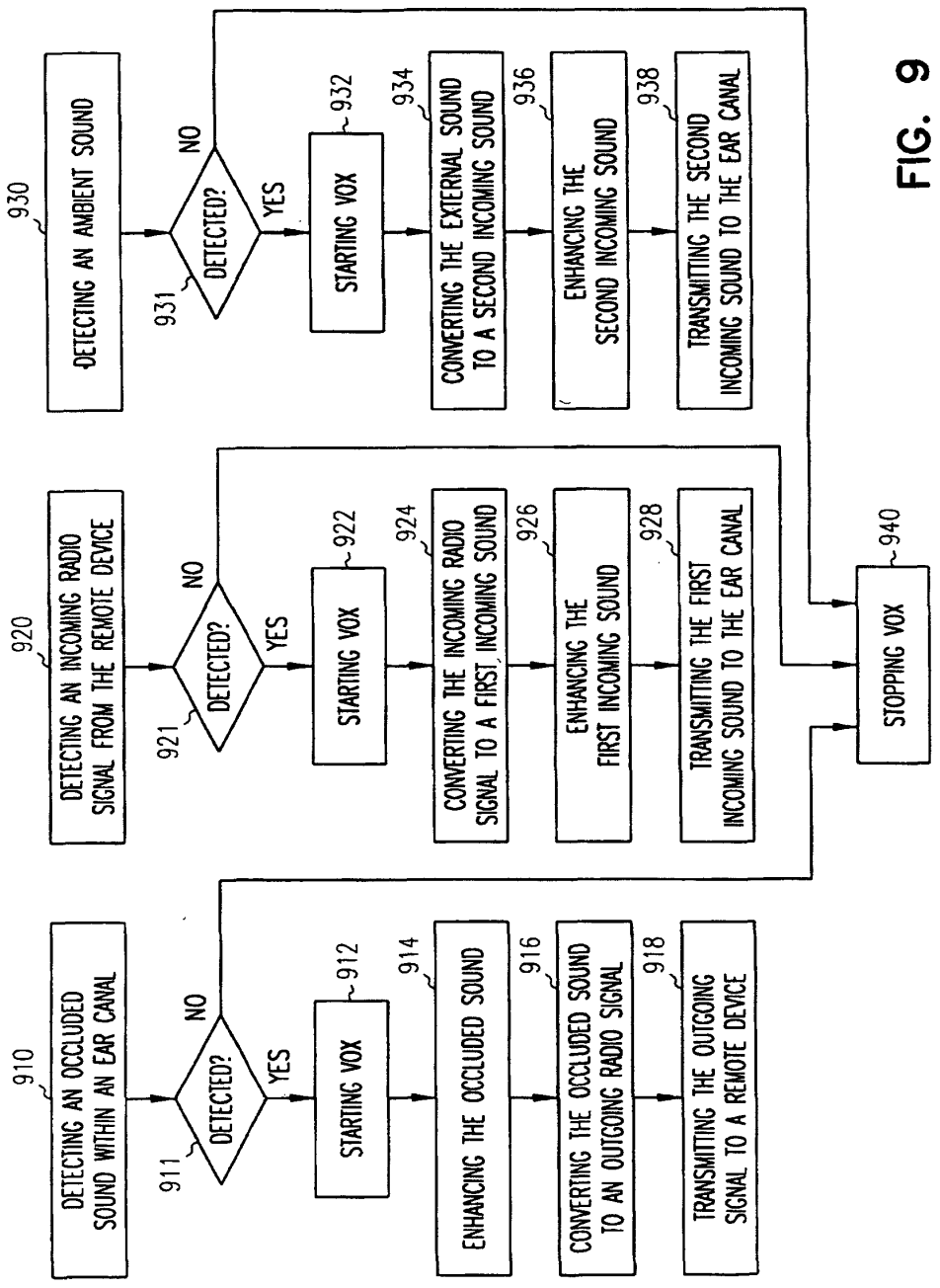

910
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
911
DETECTED?
NO
YES
912
STARTING VOX
914
ENHANCING THE OCCLUDED SOUND
916
CONVERTING THE OCCLUDED SOUND TO AN OUTGOING RADIO SIGNAL
918
TRANSMITTING THE OUTGOING SIGNAL TO A REMOTE DEVICE
920
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
921
DETECTED?
NO
YES
922
STARTING VOX
924
CONVERTING THE INCOMING RADIO SIGNAL TO A FIRST INCOMING SOUND
926
ENHANCING THE FIRST INCOMING SOUND
928
TRANSMITTING THE FIRST INCOMING SOUND TO THE EAR CANAL
930
DETECTING AN AMBIENT SOUND
931
DETECTED?
NO
YES
932
STARTING VOX
934
CONVERTING THE EXTERNAL SOUND TO A SECOND INCOMING SOUND
936
ENHANCING THE SECOND INCOMING SOUND
938
TRANSMITTING THE SECOND INCOMING SOUND TO THE EAR CANAL
940
STOPPING VOX

FIG. 9

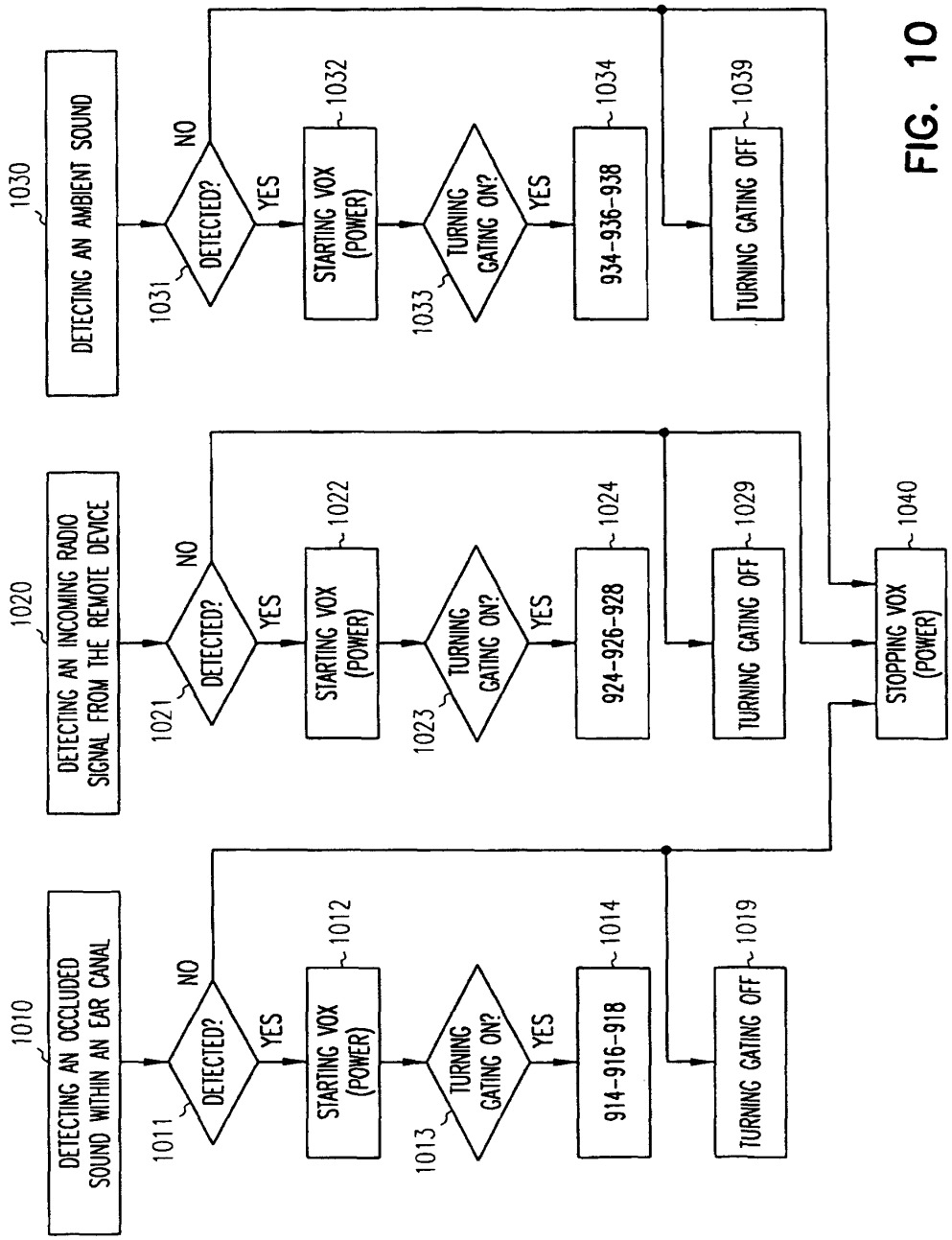

1010
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
1011
DETECTED?
NO
YES
1012
STARTING VOX (POWER)
1013
TURNING GATING ON?
YES
1014
914-916-918
1019
TURNING GATING OFF
1020
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
1021
DETECTED?
NO
YES
1022
STARTING VOX (POWER)
1023
TURNING GATING ON?
YES
1024
924-926-928
1029
TURNING GATING OFF
1030
DETECTING AN AMBIENT SOUND
1031
DETECTED?
NO
YES
1032
STARTING VOX (POWER)
1033
TURNING GATING ON?
YES
1034
934-936-938
1039
TURNING GATING OFF
1040
STOPPING VOX (POWER)

FIG. 10

Exhibit PA-G

US005251263A

United States Patent [19]

Andrea et al.

[11] **Patent Number: 5,251,263**

[45] **Date of Patent: Oct. 5, 1993**

[54] **ADAPTIVE NOISE CANCELLATION AND SPEECH ENHANCEMENT SYSTEM AND APPARATUS THEREFOR**

[75] Inventors: **Douglas Andrea**, Old Brookville; **John Kowalski**, Greenlawn, both of N.Y.

[73] Assignee: **Andrea Electronics Corporation**, Long Island City, N.Y.

[21] Appl. No.: **887,500**

[22] Filed: **May 22, 1992**

[51] **Int. Cl.**[5] **G10K 11/16**

[52] **U.S. Cl.** **381/71**; 381/94; 381/72

[58] **Field of Search** 381/71, 72, 94, 74; 379/430

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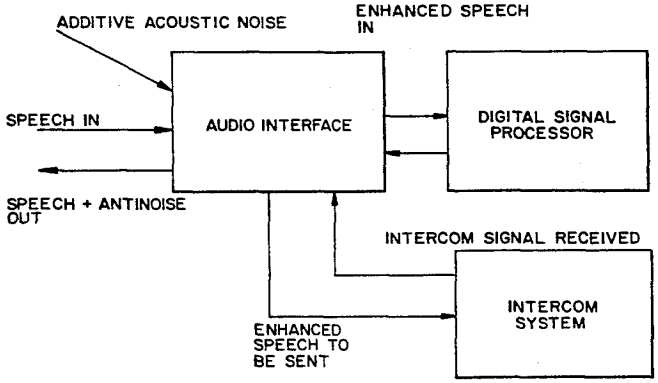
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Primary Examiner—Forester W. Isen
Attorney, Agent, or Firm—Frishauf, Holtz, Goodman & Woodward

[57] **ABSTRACT**

Disclosed is a headset apparatus for use in an intercommunications system, the headset suppressing both noise in the vicinity of a transducer delivering sound to an operator's ear and in outgoing speech from the operator.

19 Claims, 16 Drawing Sheets



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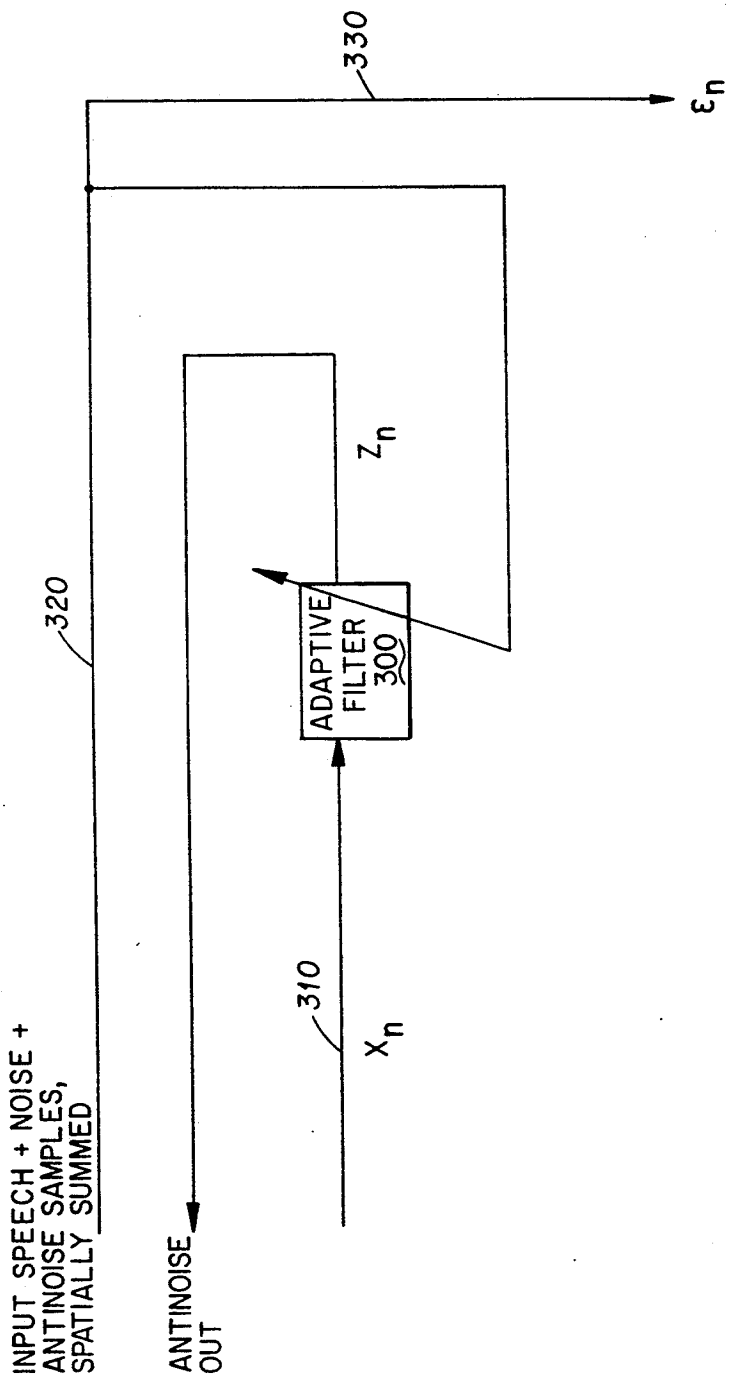

330
ε_n
z_n
320
ADAPTIVE FILTER 300
310
x_n
INPUT SPEECH + NOISE + ANTINOISE SAMPLES, SPATIALLY SUMMED
ANTINOISE OUT

Fig. 1

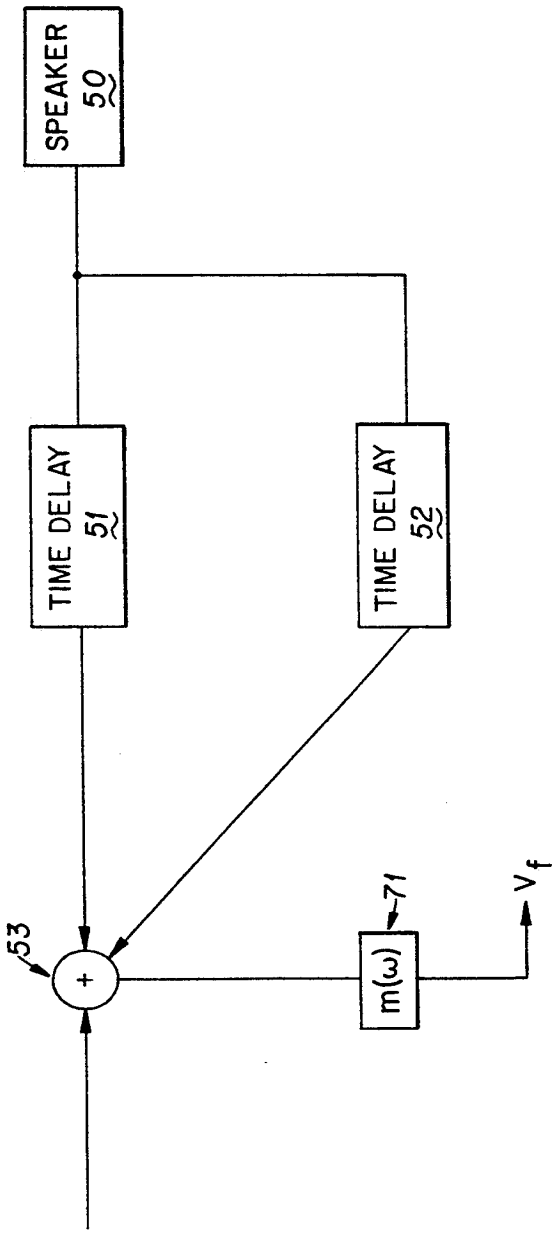


Fig. 2

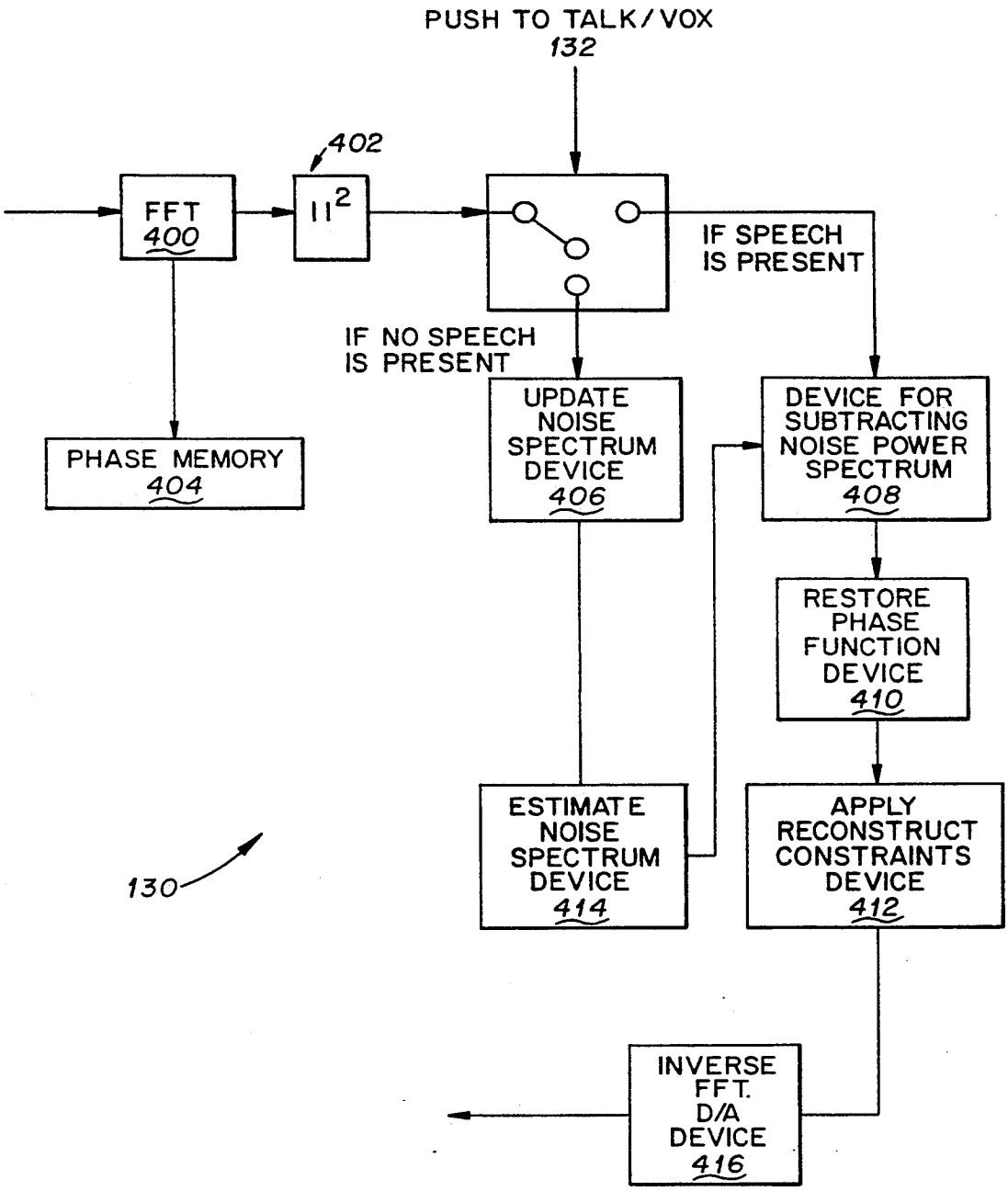

PUSH TO TALK/VOX
132
402
FFT
400
$|I|^2$
IF SPEECH
IS PRESENT
IF NO SPEECH
IS PRESENT
PHASE MEMORY
404
UPDATE
NOISE
SPECTRUM
DEVICE
406
DEVICE FOR
SUBTRACTING
NOISE POWER
SPECTRUM
408
RESTORE
PHASE
FUNCTION
DEVICE
410
ESTIMATE
NOISE
SPECTRUM
DEVICE
414
APPLY
RECONSTRUCT
CONSTRAINTS
DEVICE
412
130
INVERSE
FFT.
D/A
DEVICE
416

Fig. 3

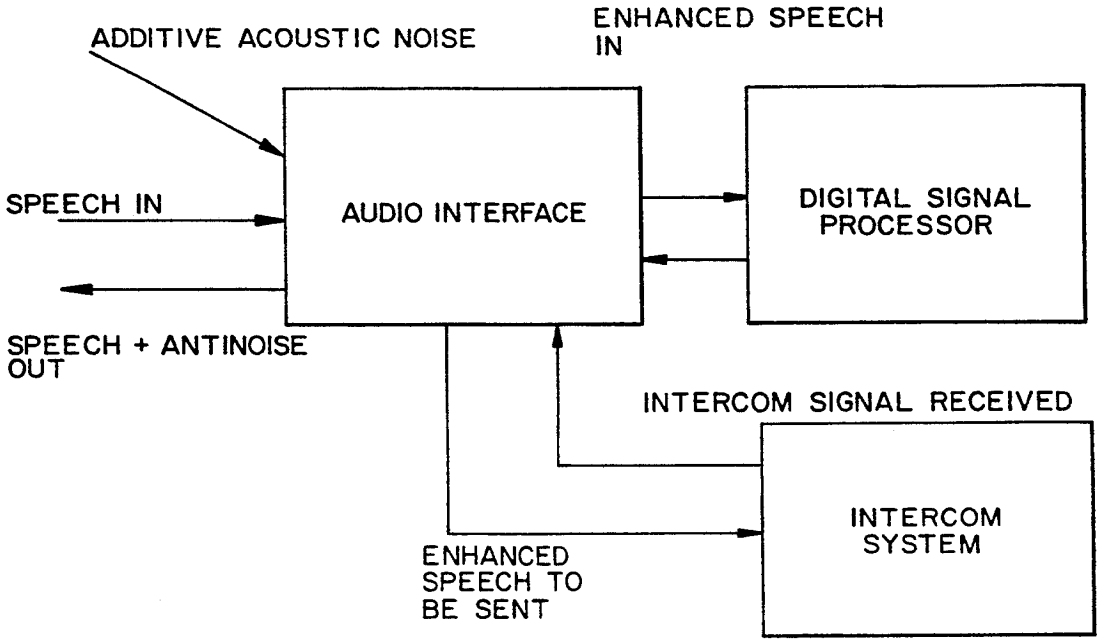

ADDITIVE ACOUSTIC NOISE
ENHANCED SPEECH IN
SPEECH IN
AUDIO INTERFACE
DIGITAL SIGNAL PROCESSOR
SPEECH + ANTINOISE OUT
INTERCOM SIGNAL RECEIVED
INTERCOM SYSTEM
ENHANCED SPEECH TO BE SENT

Fig. 7

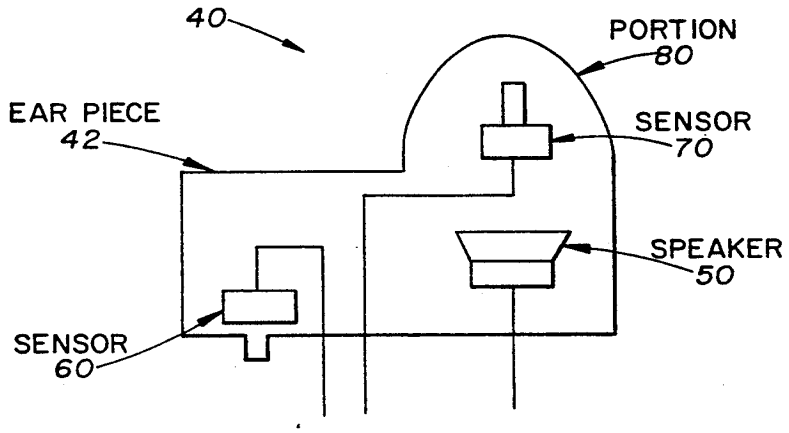

40
PORTION 80
EAR PIECE 42
SENSOR 70
SPEAKER 50
SENSOR 60

Fig. 4

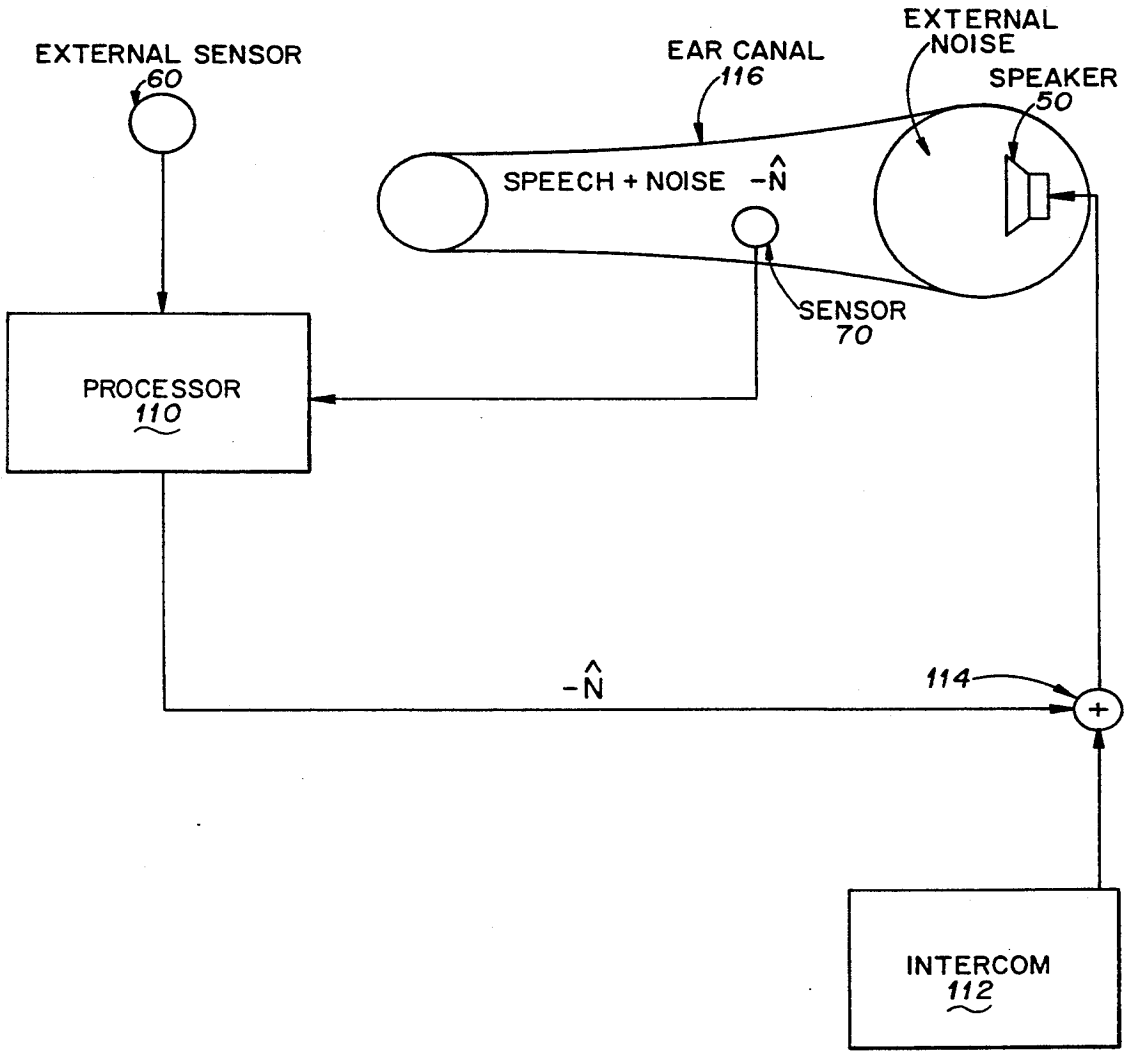

EXTERNAL SENSOR
60
EAR CANAL
116
EXTERNAL
NOISE
SPEAKER
50
SPEECH + NOISE
$-\hat{N}$
SENSOR
70
PROCESSOR
110
$-\hat{N}$
114
+
INTERCOM
112

Fig. 5

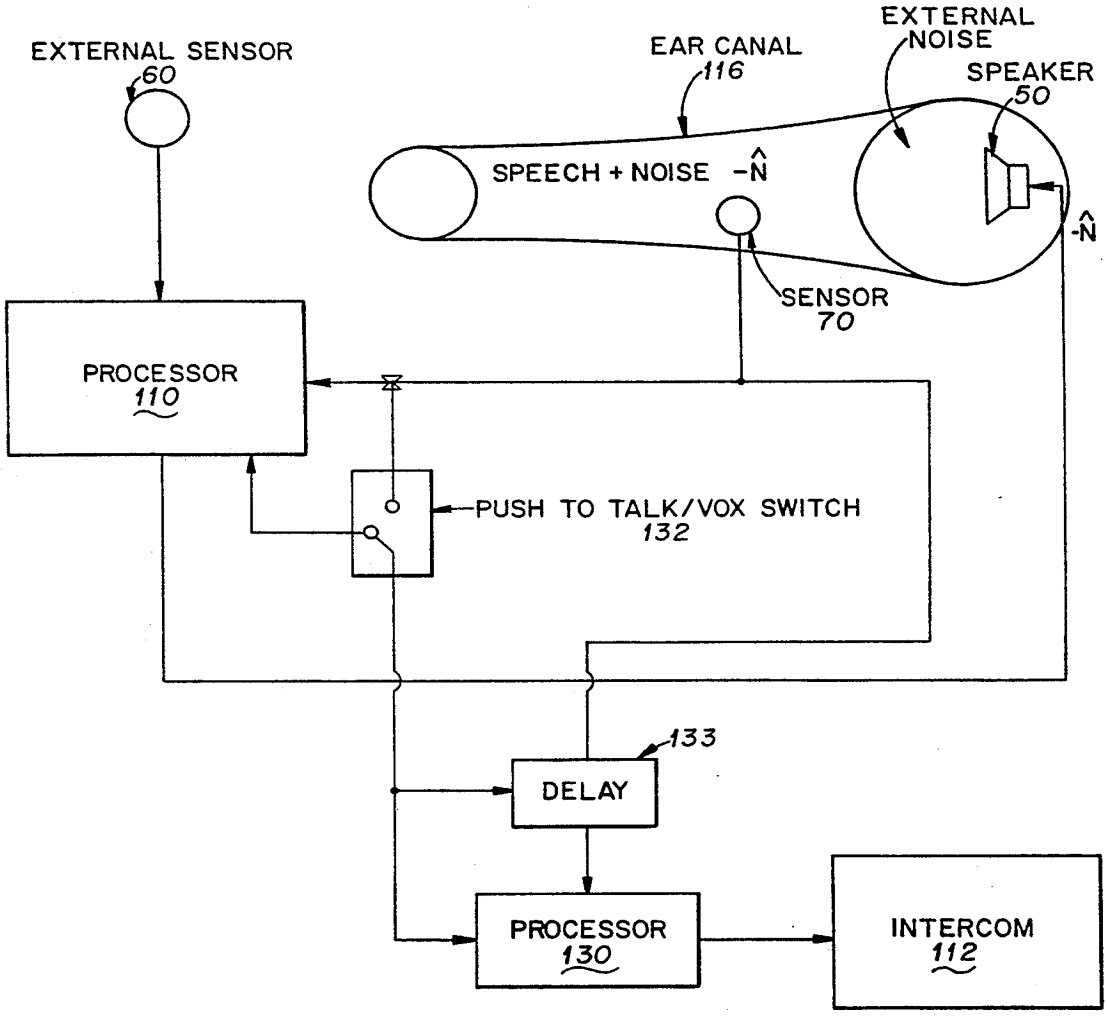

EXTERNAL SENSOR
60
EAR CANAL
116
EXTERNAL
NOISE
SPEAKER
50
SPEECH + NOISE -N̂
-N̂
SENSOR
70
PROCESSOR
110
PUSH TO TALK/VOX SWITCH
132
133
DELAY
PROCESSOR
130
INTERCOM
112

Fig. 6

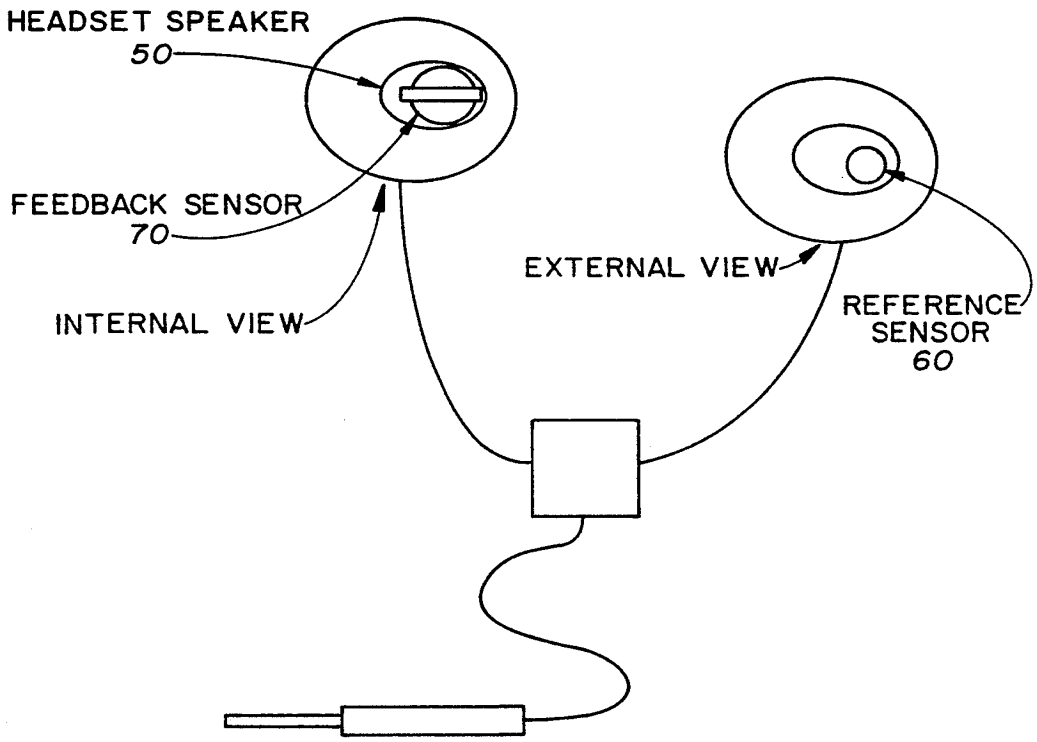

HEADSET SPEAKER
50
FEEDBACK SENSOR
70
INTERNAL VIEW
EXTERNAL VIEW
REFERENCE
SENSOR
60

Fig. 8

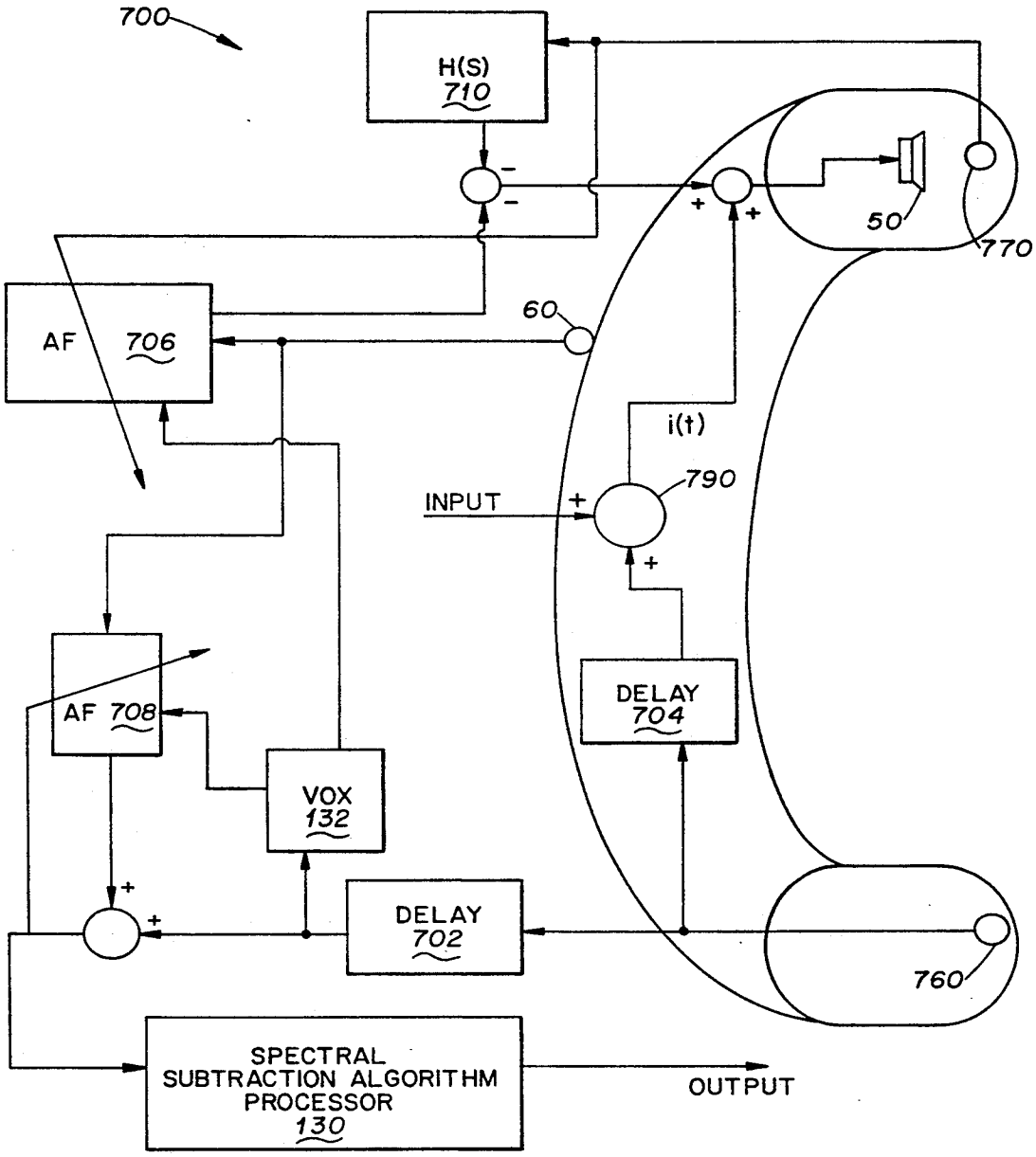

700
H(S)
710
50
770
60
AF 706
i(t)
790
INPUT
AF 708
DELAY
704
VOX
132
DELAY
702
760
SPECTRAL
SUBTRACTION ALGORITHM
PROCESSOR
130
OUTPUT

Fig. 9

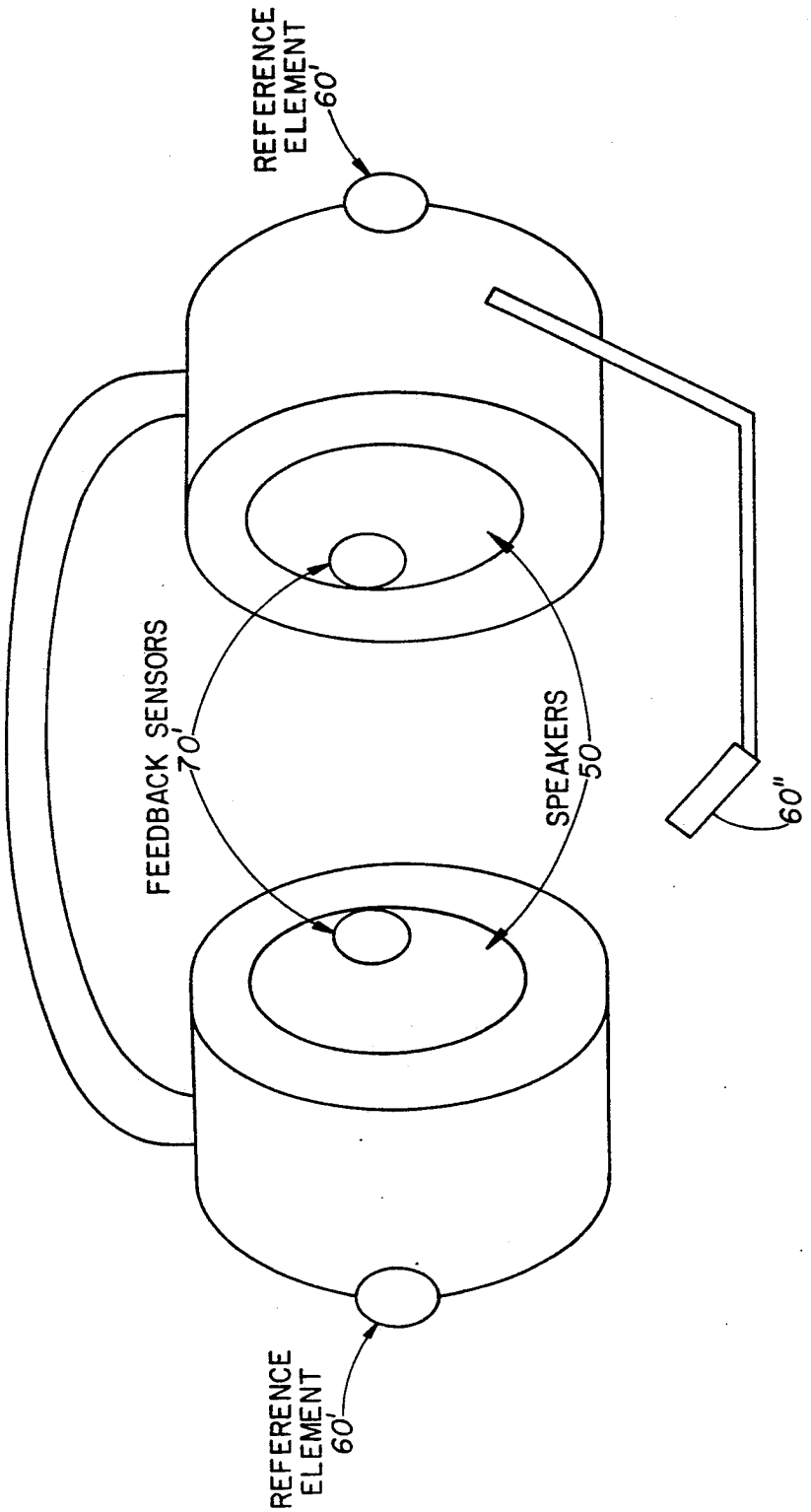

REFERENCE ELEMENT 60'
FEEDBACK SENSORS 70'
SPEAKERS 50
60"
REFERENCE ELEMENT 60'

Fig. 10

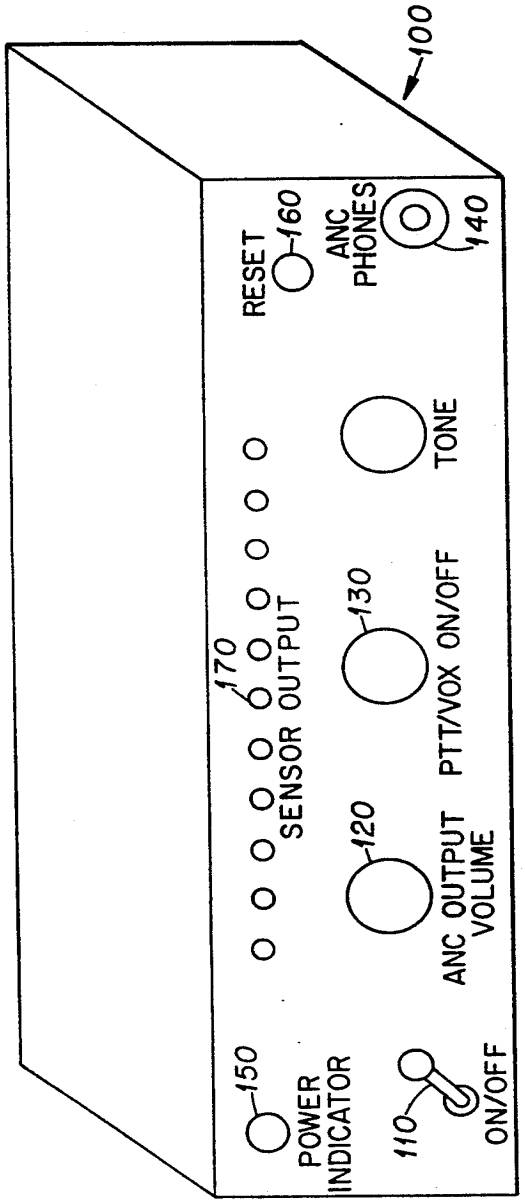

100
RESET
160
ANC PHONES
140
TONE
170
SENSOR OUTPUT
130
PTT/VOX ON/OFF
120
ANC OUTPUT VOLUME
150
POWER INDICATOR
110
ON/OFF

Fig. 11

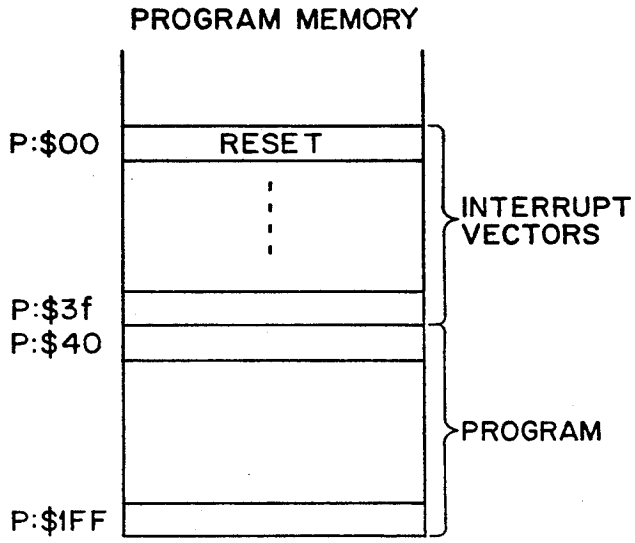


Fig. 12A

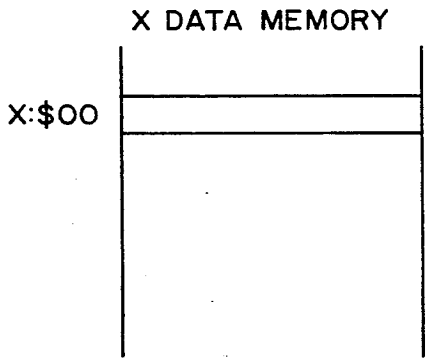


Fig. 12B

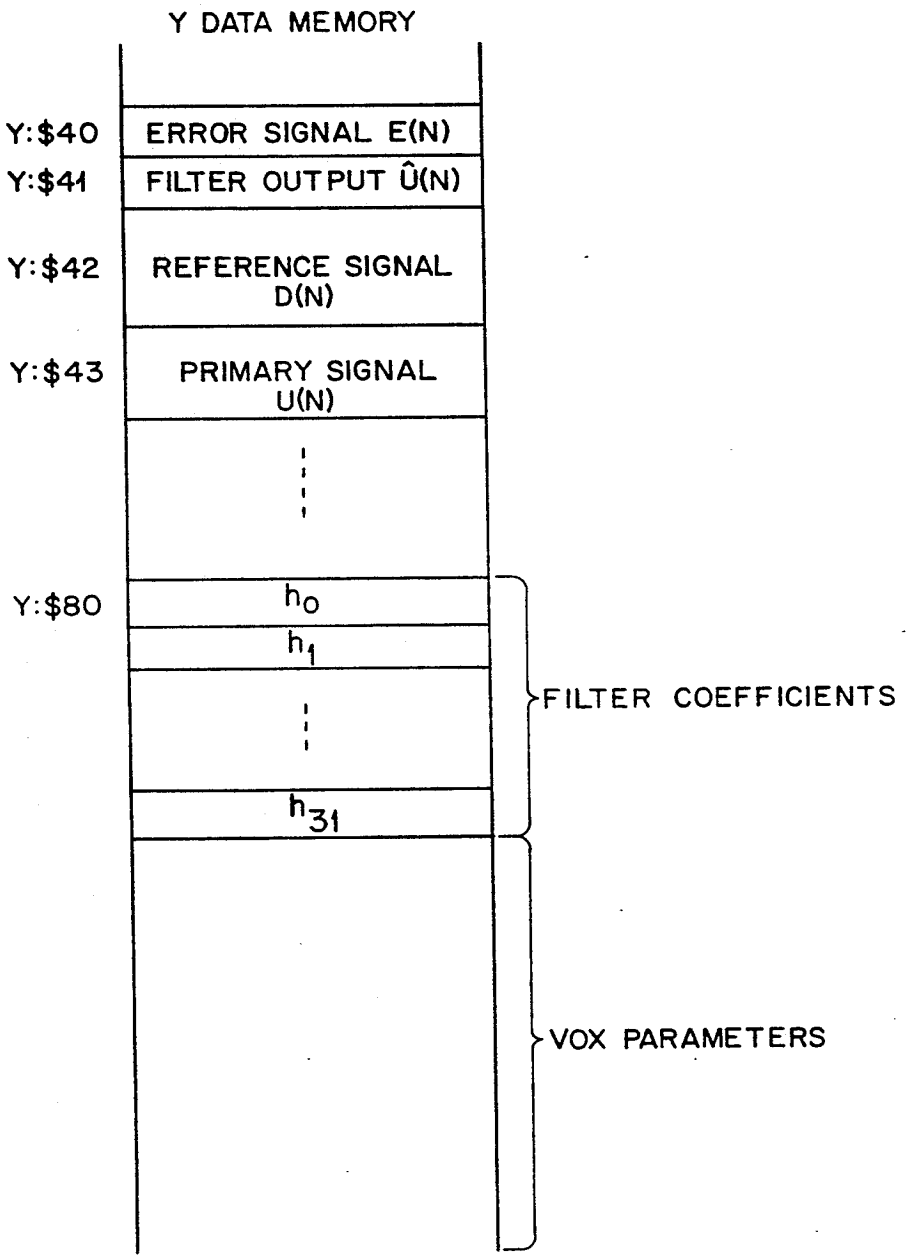

Y DATA MEMORY
Y:$40
ERROR SIGNAL E(N)
Y:$41
FILTER OUTPUT Û(N)
Y:$42
REFERENCE SIGNAL D(N)
Y:$43
PRIMARY SIGNAL U(N)
Y:$80
h0
h1
h31
FILTER COEFFICIENTS
VOX PARAMETERS

Fig. 12C

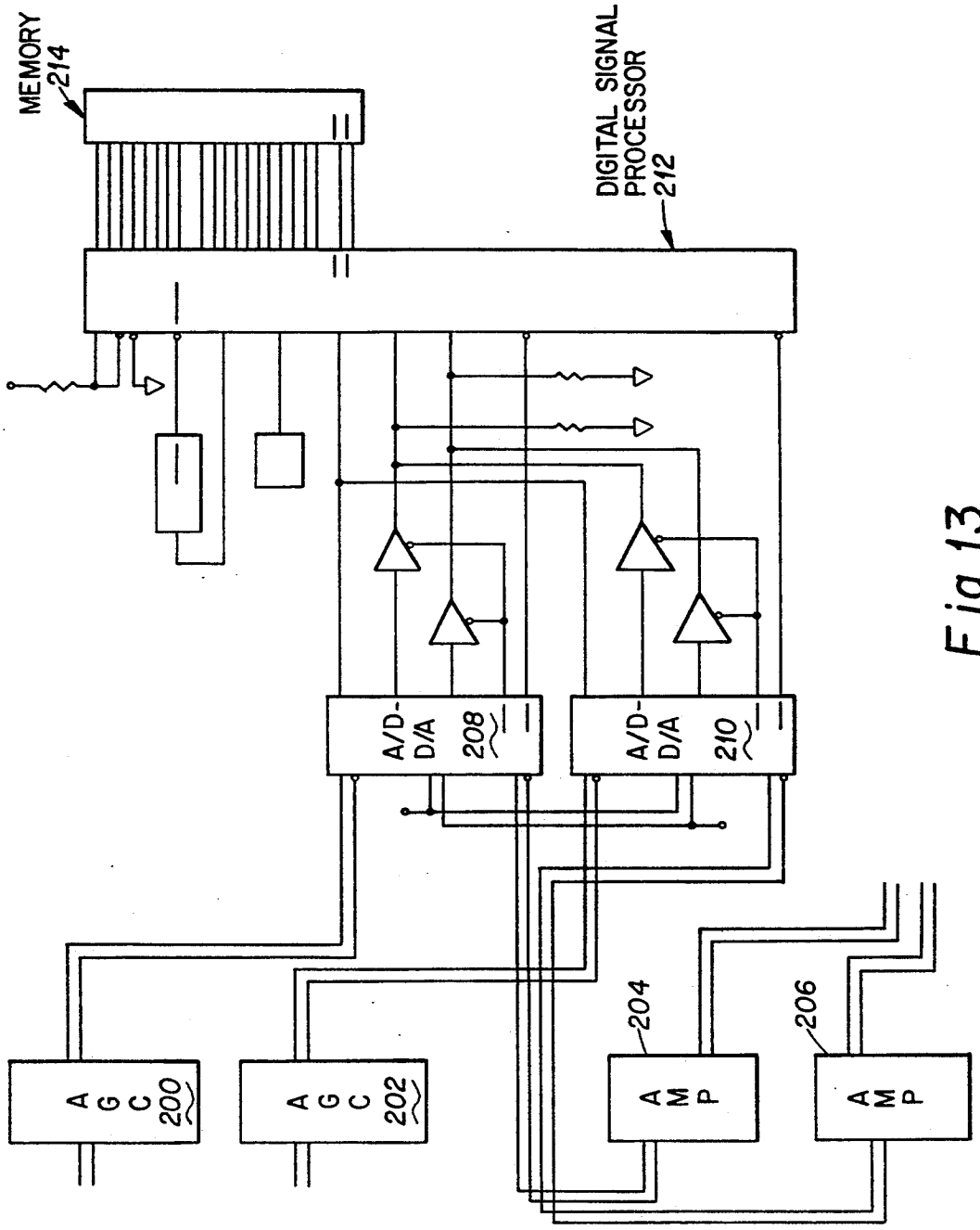

MEMORY
214
DIGITAL SIGNAL
PROCESSOR
212
A/D-
D/A
208
A/D-
D/A
210
AGC
200
AGC
202
AMP
204
AMP
206

Fig. 13

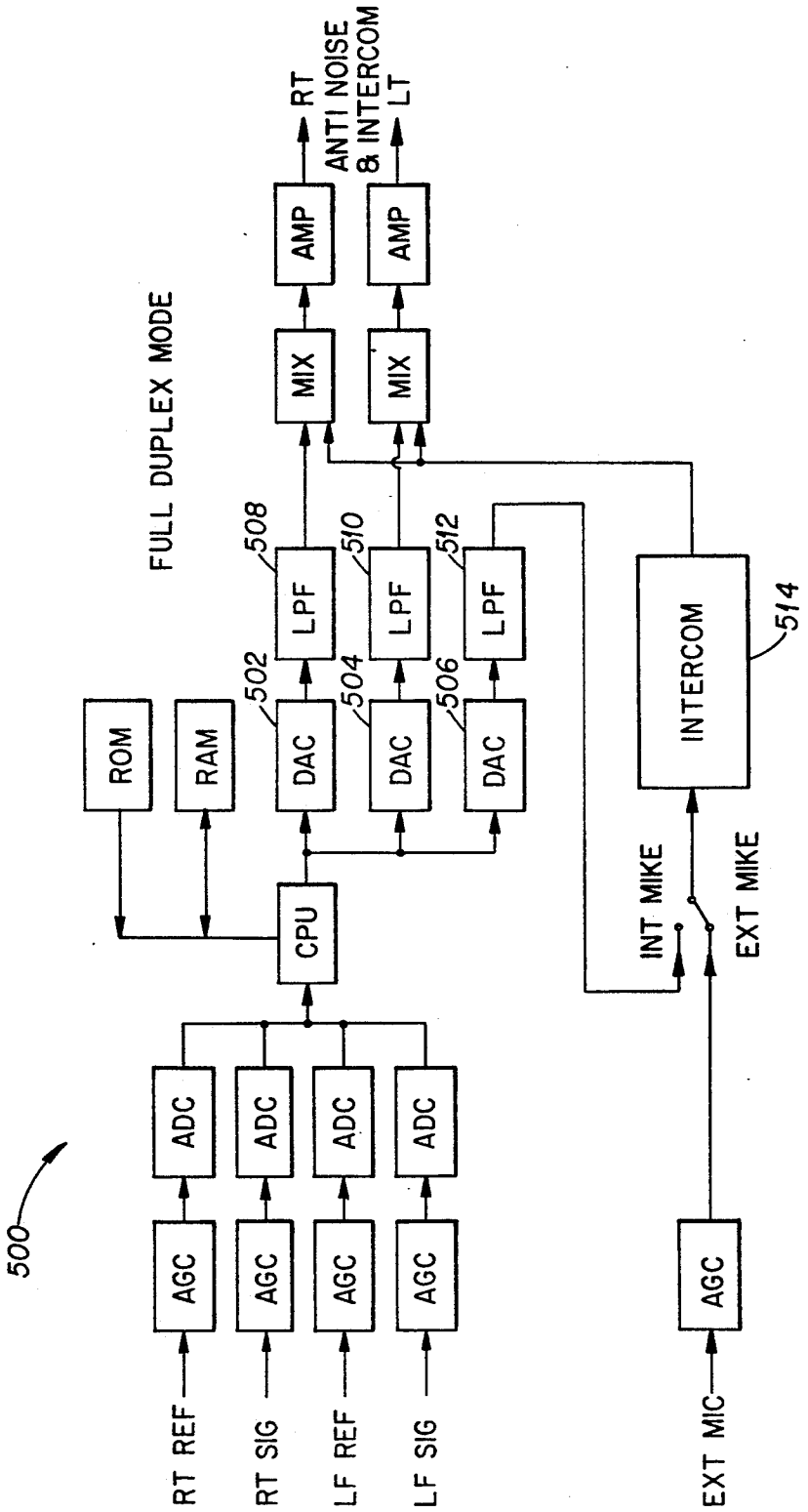

RT ANTI NOISE & INTERCOM LT
AMP
AMP
MIX
MIX
FULL DUPLEX MODE
508
510
512
LPF
LPF
LPF
502
504
506
ROM
RAM
DAC
DAC
DAC
INTERCOM
514
CPU
INT MIKE
EXT MIKE
ADC
ADC
ADC
ADC
AGC
AGC
AGC
AGC
AGC
500
RT REF
RT SIG
LF REF
LF SIG
EXT MIC

Fig. 14

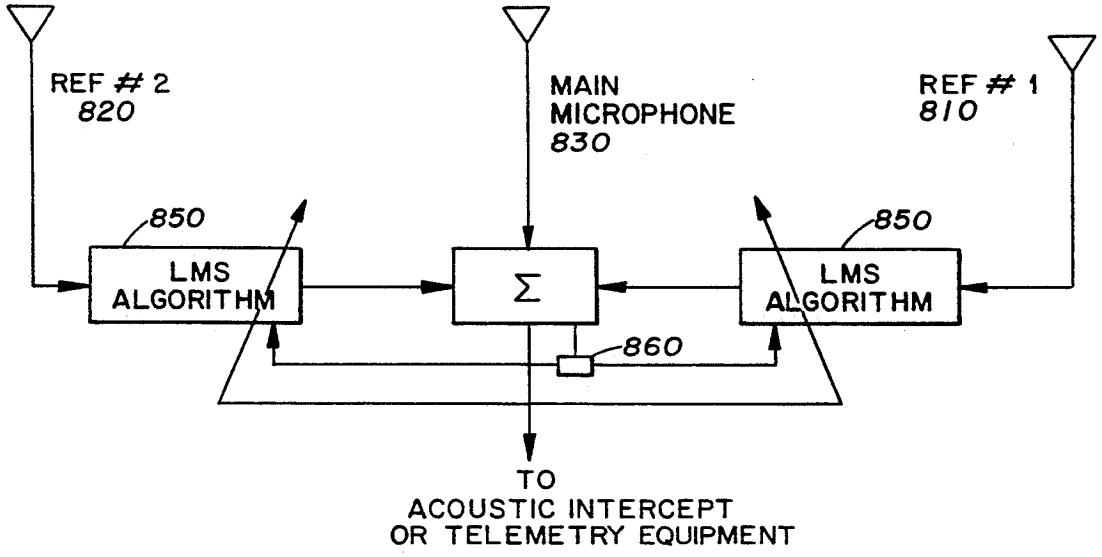

REF # 2
820
MAIN
MICROPHONE
830
REF # 1
810
850
LMS
ALGORITHM
Σ
850
LMS
ALGORITHM
860
TO
ACOUSTIC INTERCEPT
OR TELEMETRY EQUIPMENT

Fig. 15

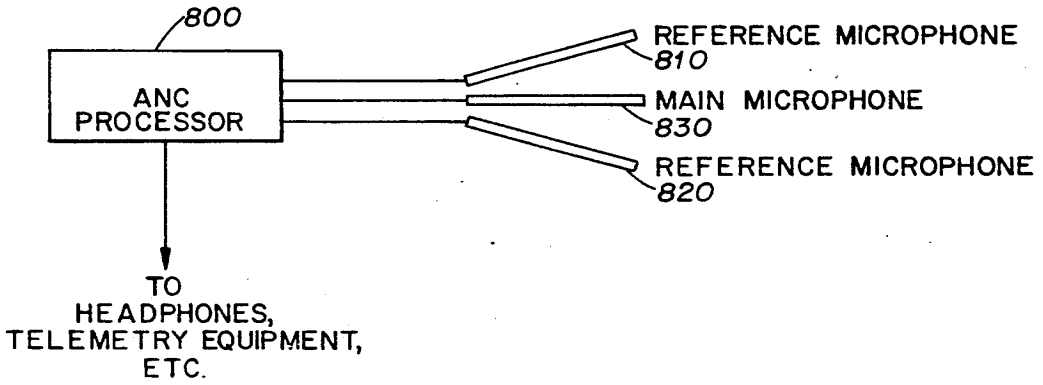

800
ANC
PROCESSOR
REFERENCE MICROPHONE
810
MAIN MICROPHONE
830
REFERENCE MICROPHONE
820
TO
HEADPHONES,
TELEMETRY EQUIPMENT,
ETC.

Fig. 16

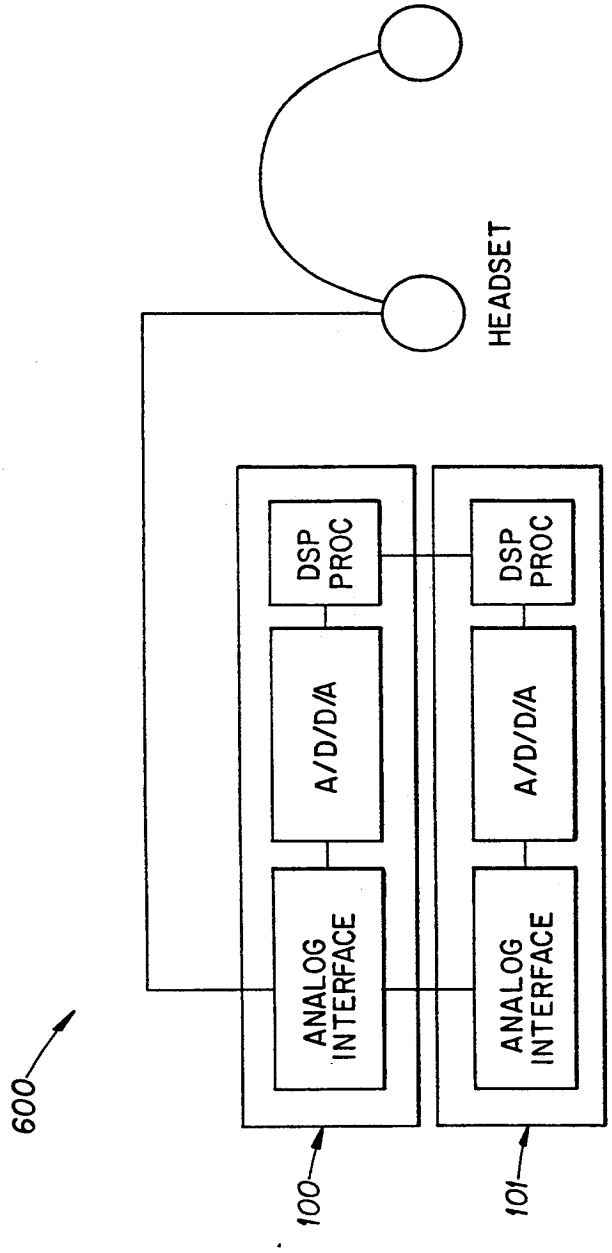

HEADSET
DSP PROC
DSP PROC
A/D/D/A
A/D/D/A
ANALOG INTERFACE
ANALOG INTERFACE
600
100
101

Fig. 17

ADAPTIVE NOISE CANCELLATION AND SPEECH ENHANCEMENT SYSTEM AND APPARATUS THEREFOR

FIELD OF THE INVENTION

The present invention relates to a noise cancellation system and apparatus therefor and, more particularly, to a system and apparatus for improving the quality, intelligibility and reliability of speech and data input into intercommunications systems and intercepts in noisy environments.

BACKGROUND OF THE INVENTION

Typically, noise cancellation systems enhance the quality of speech in an environment having a relatively high level of ambient background noise (a noisy environment). A number of systems attempt to estimate background noise so that it can be subtracted from a signal.

Acoustic noise suppression has been implemented in a wide variety of settings such as basic hearing aids (see, e.g., Langberg); cooling fans, e.g., those found in computers (see Hill); driving devices in a chamber, for instance, a compressor (see Nagayasu); propeller driven aircraft (see Elliot); vehicle seats (see Ziegler '600); and voice transmission in an emergency vehicle (see Cantrell). Dual-input adaptive cancelers are known in the communications areas (see, e.g., Widrow. 1975). Widrow (1975) particularly illustrates the use of least mean square (LMS) gradient control algorithms in such apparatus (see also Zinser and Zeiqler). These publications are cited at the end of this specification.

Although these publications describe numerous acoustic noise suppression techniques, they fail to provide an acoustic noise suppression technique as set forth in the present invention. For example, in Borth, a noise estimation means generates and stores an estimate of background noise based upon a pre-processed input signal; a noise detection means performs speech/noise decision based upon a post-processed signal; and the noise detection means provides the speech/noise decision to noise estimation means so that the background noise estimate is updated only when the detected minima of post-processed signal energy is below a predetermined minima. A key difference between the present invention and Borth is that the present invention employs an improved speech detection means utilizing delays so as to minimize any possibility of lost unvoiced consonants, preferably by use of an adaptive non-parametric detector statistic based upon a Kolmogorov-Smirnov Test, to more evenly discriminate speech from a given user from background noise consisting of several voices.

Langberg relates to an electronic earplug seated in the concha fossa (the hollow external portion adjacent to the opening of the ear canal), which acts as a passive acoustical barrier. The earplug contains a summing microphone which detects noise which has penetrated the occluded ear canal and the output signal from the summing microphone is used to initiate active noise reduction. Langberg does not appear to teach or suggest an in-ear or in-earpiece microphone for detecting speech, nor does Langberg describe or the headset of the present invention.

More specifically, Langberg does not appear to teach or suggest compensation for use in the ear. Langberg also does not appear to teach or suggest filtering to account for density changes in the ear which may otherwise lead to ear canal/middle ear impedance mismatching or instabilities at certain frequencies.

Further, an embodiment of the present invention utilizes an "in-ear" microphone to transmit speech when a push-to-talk or voice-operated-switch (VOX) switch is depressed or activated. The embodiments of the present invention also utilize a "reference sensor" located in an external portion of the "earplug", which is acoustically isolated from the earplug, to measure background noise. Another embodiment of the present invention employs an adaptive filter means, e.g., using a least mean square (LMS) algorithm, to account for variations in the feedback path. These features do not appear to be taught or suggested by Langberg.

Zeigler '188 relates to canceling only harmonic disturbances. Unlike Zeigler, in the present invention, any type of random or harmonic disturbances may be canceled. Further, in the present invention, there are compensators for feedback and reference paths to ensure that the channels are matched in both amplitude and phase over a specified band; this is done non-adaptively. Furthermore, the present invention also provides filtering compensation to achieve broadband as well as narrow band cancellation.

Nagayasu merely eliminates noise without any apparent teaching or suggestion to enhance speech. Landgarten merely relates to monitoring, testing and controlling vibration. Sasaki does not appear to teach or suggest employing an adaptive system that automatically compensates for changes in feedback as in the present invention. For instance, the present invention may use filtering to compensate for the speaker/ear canal transfer function (to match reference and feedback channels). This filtering does not non-adaptively couple the reference and feedback signals. In the present invention, filtering is used to minimize signal decorrelation effects so as to extend the ability of the adaptive processor to cancel noise when the noise statistics and feedback path change.

Stettiner is akin to Borth and likewise fails to teach or suggest a VOX switch with a push-to-talk option for determining speech from noise. In the present invention which is a voice detection means (algorithm) is streamlined and robustized by using inter alia, a nonparametric test such as a Kolmogorov-Smirnov Test. Cantrell addresses the dominant harmonic by essentially using a phase locked loop approach to control a notched filter. While the present invention focuses on speech enhancement, by the use of adaptive filters, the present invention is able to work with a wider variety of modulated signals, as well as with several given signals at a time. Elliot relates to zonal quieting to control the phase of a propeller or fan and does not appear to relate to speech enhancement. Likewise, Hill is concerned with reducing noise in rotating equipment, such as, a fan, and does not appear to relate to speech enhancement as in the present invention. Zinser merely provides a variation on the LMS algorithm.

Thus, the prior art fails to provide a noise cancellation and speech enhancement system and apparatus as described in the present invention. More specifically, the prior art fails to provide a noise cancellation system including a spectral subtractor and a push-to-talk or VOX switch which enables a speech/noise determination by use of test statistics including, for example, sample zero crossings, changes in the number of tonals,

energy and a nonparametric test such as a Kolmogorov-Smirnov Test. Further, utilizing a spectral algorithm with a LMS algorithm in a noise cancellation and speech enhancement system and apparatus, as in the present invention, has not been taught or suggested. Furthermore, a noise cancellation system wherein the spectral algorithm not only employs the above-described test statistics, but also includes a constraint function for minimizing residual musical noise, as well as a tracking device to identify tonal noise components and predict trajectories thereof, as in the present invention, has not been taught or suggested. Further, the in-ear and in headset (or handset, e.g., for telephones) devices of the present invention which utilize the previously mentioned features have not been heretofore taught or suggested.

While all of the above-described prior art systems relate to acoustic noise cancellation, they are limited by the lack of performance in severe noise environments; namely, they fail to perform in severe noise environments wherein the noise is highly impulsive or poorly defined.

OBJECTS AND SUMMARY OF THE INVENTION

It is an object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus which avoids the foregoing problems associated with the prior art.

More specifically, it is an object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus for enabling a more acceptable detection of speech from a signal having a speech signal in the environment of a relatively high level of noise.

It is another object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus for improving the quality of the output speech obtained from subtracting noise from a signal having a speech signal and noise.

It is a further object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus for improving the quality and intelligibility of speech input into intercommunications systems in noisy environments, e.g., automobiles, boats, trucks, work places (such as construction, manufacturing, stock or commodities trading, and the like) and public places (such as bars, taverns, restaurants, and the like) and, particularly, for aircraft and military environments.

It is yet another object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus which can be applied to an existing intercom, radio, acoustic intercept or communications systems, so as to require relatively little or no modifications to the existing system.

It is still another object of the present invention to provide an adaptive noise cancellation and speech enhancement system and apparatus which can be employed in various applications, including, without limitation, telephone and telecommunications applications and adaptive microphone applications.

In accordance with an aspect of this invention, headset apparatus is provided which comprises at least one housing each having first microphone means for receiving a reference signal, second microphone means for receiving a signal from within the respective housing and speaker means for providing an acoustic signal to the operator, and means coupled to each of the housings for holding the housing over a respective ear of the operator.

In accordance with another aspect of this invention, headset apparatus is provided which comprises a housing having speaker means for providing an acoustic signal to an operator using the headset, first microphone means for receiving the signal from the operator, second microphone means for receiving a reference signal, third microphone means for receiving the acoustic signal, delay means for delaying the signal from the operator so as to form a delayed signal, and means for receiving an input signal and for combining the same with the delayed signal for supplying to the speaker means.

In accordance with still another aspect of this invention, a system is provided for communicating between two or more operators which comprises headset means including at least one housing each having first microphone means for receiving a reference signal, second microphone means for receiving an internal signal from within the respective housing and speaker means for providing an input signal to the respective operator using the headset means; first processing means for respectively processing each of the reference signals received from the respective first microphone means so as to form a first processed signal; and means for combining the respective processed reference signal and a signal from another of the operators so as to form the input signal.

In accordance with still a further aspect of this invention, a system is provided for communicating between two or more operators which comprises headset means including a housing having speaker means for providing an acoustic signal to the operator using the headset means, first microphone means for receiving a signal from the operator using the headset means, second microphone means for receiving an external reference signal, third microphone means for receiving the acoustic signal, and means for receiving an input signal from another of the operators and combining the same with the signal from the operator for supply to the speaker means; first processing means receiving respective output signals from the first, second and third microphone means for processing the same in accordance with a first predetermined processing so as to obtain a first processed signal; and second processing means for processing the first processed signal in accordance with the second predetermined processing so as to obtain a second processed signal for supply to the other operator.

Other objects, features and advantages of the present invention will become apparent from the following detailed description of the illustrated embodiments when read in conjunction with the accompanied drawings in which corresponding components are identified by the same reference numerals.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 illustrates a device utilizing a LMS algorithm according to an embodiment of the invention;

FIG. 2 illustrates a speaker, ear canal and microphone;

FIG. 3 illustrates a device utilizing a Spectral Subtraction Algorithm device according to an embodiment of the invention;

FIG. 4 illustrates an in-ear headset according to an embodiment of the invention;

FIG. 5 illustrates a receive mode of the system according to an embodiment of the invention;

FIG. 6 illustrates a transmit mode of the system according to an embodiment of the invention;

FIG. 7 illustrates an adaptive noise cancellation and speech enhancement system according to an embodiment of the invention;

FIG. 8 illustrates a headset unit according to an embodiment of the invention;

FIG. 9 illustrates a telephone headset system according to an embodiment of the invention;

FIG. 10 illustrates a headset unit according to an embodiment of the invention having a boom mike;

FIG. 11 illustrates an ANC unit according to an embodiment of the invention;

FIGS. 12A, 12B an 12C illustrates a memory map;

FIG. 13 illustrates a schematic diagram for incorporating the algorithms;

FIG. 14 illustrates the schematic diagram of a multiple channel embodiment of the processor of FIG. 13 in block form;

FIG. 15 illustrates an adaptive microphone in block diagram form for use with the system of the invention;

FIG. 16 further illustrates the microphone of FIG. 15; and

FIG. 17 illustrates an expansion unit in accordance with another embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The adaptive noise cancellation (ANC) and speech enhancement system and apparatus of the present invention provide acoustical and electrical "in wire" speech enhancement for intercom, telephone, and radio application and the like in a compact, reliable lightweight low power consumption unit.

The system uses advanced digital signal processing technology to implement adaptive noise canceling and a digital voice operated switch (VOX).

The system includes an adaptive noise canceling headset, adaptive processor, and an analog-to-digital (A/D) and digital-to-analog (D/A) conversation interface. The headset includes a reference microphone, feedback microphone, speaker, housing (e.g., a plastic housing) and wires coupled to the processor. The A/D interface converts the voltage levels and impedances from those of the microphone to those required by an A/D chip for optimal (minimum mean square error) quantization. A/D and D/A conversion is accomplished by use of any suitable chip or chips, preferably two Texas Instruments TLC32046 A/D/D/A chips.

The digital outputs of the A/D and D/A chips are fed to a serial port on a digital processor, which is preferably a Motorola DSP56001. The DSP56001 processor performs all digital signal processing functions, and supplies the outputs to the above-mentioned interface chips. The output signals are (1) used to drive the speaker for acoustic, e.g., in-ear cancellation and (2) an output speech enhanced signal that is used for intercom and voice radio transmission. This signal optionally can be encoded to easily interface with digital radios. That is, the processor can provide speech that is continuously variable slope delta (CVSD) modulated or adaptive predictive coded (APC) or otherwise encoded or encrypted.

The system provides the capability to cancel one channel acoustically and to provide one channel of "in wire" enhanced speech. The system also has the capability to be adapted for growth to two acoustic channels and one channel of in wire cancellation by use of an additional ANC unit, designated as an expansion unit in the following discussion. The expansion unit contains the additional A/D/D/A's conversion and processing. The system of the invention can also be further expanded, for instance to four channels, as discussed below.

The system is preferably battery operated with DC to DC conversion circuitry to enable the system to be used in a variety of environments.

The system provides at least 10 dB signal to noise ratio improvement at a +3 dB signal to noise ratio (or less) over a 60 dB dynamic range. This number is measurable by using colored Gaussian noise in a 0 to 4 kHz band and using additive speech.

All basic system circuitry including the analog and digital signal processing circuitry is preferably contained on one or more relatively small PC boards. As a result, a processor **100**, as shown in FIG. **11**, has a volume of approximately 20 cubic inches. The basic processor unit **100** will be hereinafter referred to as the ANC unit.

With one channel of acoustic cancellation being performed using the basic unit, the headset has one ear with acoustically damping padding. When the basic ANC unit is employed with an Expansion Unit, a two-channel headset provides acoustic cancellation in each ear. The ANC unit may also contain a larger number of channels, for example, 4 channels, which may be configured so as to lie on a single board. (This unit is hereinafter referred to as "Four Channel Unit").

The system controls of the ANC unit **100** (FIG. **11**) include an ON/OFF switch **110**, an ANC output volume control **120**, a VOX threshold control and a push-to-talk (PTT) switch **130**. The headset plugs into the ANC processor unit at an interface **140**. The ANC processor unit **100** also contains jacks for interfacing with intercoms and radios and for expansion to four channels using the Expansion Unit. The ANC unit **100** may also have a power indicator **150**, a reset **160**, and a sensor output **170**.

An in-ear headset unit **40**, and its operation in conjunction with the system are generally shown in FIGS. **4**, **5**, and **6**.

The in-ear headset **40** (FIG. **4**) generally includes a speaker **50**, a reference sensor or microphone **60** which is adapted to detect external background signals, and a feedback/voice transmission sensor or microphone **70** which are mounted in a single ear piece **42**. A portion **80** of the in-ear headset **40** is located closest to or in the ear canal opening of the operator using the headset **40**. As is to be appreciated, the respective positions of the speaker **50** and sensors **60** and **70** relative to the portion **80** may vary. For the operator's other ear, a conventional passive attenuator headset (not shown) can be employed to reduce the noise. As a result, an enhanced performance headset, that is, a headset which includes the in-ear headset **40** and the conventional passive attenuator headset, provides active adaptive speech enhancement in both ears. This headset is preferably used when the Expansion Unit or multiple channel unit is present.

The microphones or sensors preferably provide linearity up to approximately 125 dB spl, and each has a dynamic range of at least 75 dB. The distortion of the microphone is preferably less than 0.5% total harmonic distortion (THD). The total group delay for the microphone is preferably less than 25 microseconds (1/40 kHz).

The speaker **70** in the headset **40** and in the passive attenuator headset (not shown) are preferably those typically used in portable, lightweight, in-ear headsets. These speaker(s) preferably have a THD less than 1% and generate 125 dB spl at a distance of approximately 0.5 inches away from the speaker. The speaker(s) preferably have a group delay <25 microseconds.

FIGS. **5** and **6** illustrate the operation of the headset **40** in the receive and transmit modes, respectively. More specifically, in the receive mode, as shown in FIG. **5**, external background or reference acoustic sounds (which may be referred to as noise), for example, sirens, engine noises and crowd noises, are picked-up by the sensor **60** and supplied to a processor **110** along with a feedback or error signal from the sensor **70**. The processor **110** processes the received signal in accordance with a predetermined algorithm, which is preferably an LMS type algorithm. That is, the LMS processor **110** supplies the reference signals through a tap delay finite impulse response (FIR) filter, in which the coefficients of the filter (filter taps) are controlled by correlating the feedback signal with the respective signal at each tap. The output of the processor **10**, which represents antinoise, is supplied to a combining circuit **114** along with a signal from an intercom **112**. The combined signal from the combining circuit **114** is supplied to the speaker **50**, contained within the headset **40** which is located in the ear canal **116** of the operator, so as to be heard by the operator.

In the receive mode (FIG. **5**), the LMS algorithm alone is used to cancel the ambient noise. The headphone speaker transduces the summed antinoise and intercom output.

In the transmit mode, as shown in FIG. **6**, background or reference noise is pick-up by the sensor **60** and supplied along with the error signal from the sensor **70** to the processor **110** so as to output an anti-noise signal for supply to the speaker **50**, as hereinbefore described with reference to FIG. **5**. When the operator wearing the headset **40** wishes to speak with another person via the present system, the operator speaks aloud so as to form a signal which is translated through the operator's eustachian tube (which extends approximately from the back portion of the throat to the middle ear portion of the ear) to the inside of the ear. As a result, this signal is detected by the sensor **70** and thereafter supplied to a processor **130** through a delay circuit **133** which imposes a predetermined delay so as to capture speech onsets and a switch **132**, which may be either a push-to-talk switch or a VOX switch. If the push-to-talk switch is being utilized, then when the operator wishes to talk to the other person(s), the operator merely activates the switch **132** so as to supply the signal from the sensor to the processor **130**. If, on the other hand, the VOX switch is used, a VOX algorithm, as hereinafter more fully described, determines that the signal from the sensor **70** is activating a speech or voice signal from the operator and not just an error or feedback signal whereupon the VOX switch enables the voice signal to be transmitted to the processor **130**. The processor **130** processes the received signals in accordance with a predetermined algorithm, such as a spectral subtraction algorithm as hereinafter more fully described. The processed signal from the processor **130** is thereafter supplied to the intercom **112** so as to be heard by the other person. Further, upon activating the switch **132**, the signal from the sensor **70** is not supplied to the processor **110**, since this signal is currently a voice signal and not an error signal.

In the transmit mode (FIG. **6**), the headset speaker **50** is used to play the antinoise output from the Least Mean Square (LMS) algorithm of the Adaptive Processor **110**. The error signal is produced by the spatial summation of the speech, the estimated antinoise, and the ambient acoustic noise. This error signal is correlated with an externally received reference signal from the sensor **60** to derive the optimum Finite Impulse Response (FIR) filter parameters, using the below equations:

$$Z_n = W_n^T X_n \tag{A}$$

$$X_n = (X_n, X_{n-1}, \ldots X_{n-N})^T \tag{B}$$

$$W_n = (W_0 W_1, \ldots W_N)^T \tag{C}$$

$$W_n = (1-a)W_{n-1} + Ga\epsilon_n X_n \tag{D}$$

The inputs to the LMS algorithm are compensated in software to account for the transfer function of the speaker element.

The error signal is also sent to the spectral subtraction processor **30**. When no speech is transmitted as determined by the Push-To-Talk (PTT) or Voice Operated (VOX) switch **132**, the spectral subtraction processor updates the noise spectrum estimate. However, when the PTT is depressed, or when the VOX detects a speech signal, the spectral subtraction processor disables the noise spectrum updating, and uses the noise spectrum to subtract from the speech plus noise spectrum.

The spectral subtraction algorithm or processor **130** is able to remove residual broadband components present in the signal to be transmitted which the LMS algorithm cannot, because the LMS algorithm's residual noise signal is approximately uncorrelated with the reference data in steady state.

The signal processing for the present system is preferably performed in circuits contained within the processor unit **100** (FIG. **11**). An example of such circuits is illustrated in FIGS. **13** and **14**. More specifically, as shown in FIG. **13**, input analog signals are respectively received and supplied through automatic gain control (AGC) circuits **200** and **202** to A/D-D/A converters **208** and **210**, acting herein as analog interface chips, so as to be converted into digital form. The digital output signals therefrom are supplied to a digital signal processor **212** which receives the algorithm from a memory device **214**, for example, a ROM, during initialization of the system. The processor **212** preforms the processing in accordance with the stored algorithms and respectively supplies processed output digital signals through A/D-D/A converters **208** and **210**, so as to be converted into analog form, to amplifiers **204** and **206** so as to be amplified and supplied therefrom.

In the preferred embodiment, the circuits as shown in FIG. **13** are utilized. That is, in one circuit, the VOX and the LMS algorithms are stored in the memory **214**, and, in the second circuit, the spectral subtraction algorithm is stored therein.

The components without reference numeral designations will not be specifically discussed, since these connections are readily apparent to those skilled in the art.

FIG. **14** illustrates a multiple channel adaptive noise cancellation unit **500** which is substantially similar to that previously in reference to FIG. **13**. However, the unit **500** is capable of providing acoustic cancellation in

two ears and in-wire cancellation, as hereafter described. Further, in multiple adaptive noise cancellation unit **500** there are three digital-to-analog converters **502, 504** and **506** and low-pass filters **508, 510** and **512** so as to provide the necessary signals for each ear and an intercom **514** connected thereto. Further description of unit **500** will not be presented herein as it is otherwise substantially similar to the circuit of FIG. **13**.

The outputs from the processor are converted to analog, and sent to the intercom. Depending on whether the intercom operates in half or full duplex mode there are two or three outputs from the processor which are output time division multiplexed.

The system uses a relatively low amount of power, that is, approximately 5 watts or less.

The ANC unit **100** cancels noise without distorting speech in a predetermined frequency bandwidth, which is preferably 0 to 4 kHz. This bandwidth is maintained from the sensor inputs to the A/D chips. The signal levels at the microphone outputs are preferably in the range of 5–20 mV rms. The analog interface circuitry preferably provides gain sufficient to be within the limiting region of the A/D 99.9% of the time when Gaussian noise is digitized. That is, the levels are set in such a way as when a Gaussian signal with average power given by the average power rating for the microphone is used, the gain is such that clipping occurs 0.1% of the time. The signals in the reference and feedback preferably track each other to within 1 dB in amplitude and within 0.5 degree of phase over the zero to 4 kHz band.

The ANC analog interface circuitry preferably provides outputs compatible with standard intercoms and radios. Optionally, the ANC can provide a digital interface to be used for digital radio communication.

Each automatic gain control (AGC) device preferably provides a dynamic range of 70 dB. The attack time and decay times, defined as the settling times for the AGC, are preferably 1.5 ms and 125 ms, respectively. Each AGC provides exponential gain control by feeding back the magnitude or magnitude squared of the input signal, subtracting a bias and low pass filtering the result. This signal is then supplied to an attenuator included in the AGC's **200** and **202** (FIG. **13**).

As stated earlier, the analog and data conversion hardware are expandable through the use of an optional expansion unit. The analog interface circuitry provides a low impedance output (approximately 8 Ohms) with variable levels to provide a match for levels to most intercom and radio interfaces.

The digital signal processing for two channels is performed by any suitable processor, for example, a Motorola DSP 56001 programmable digital signal processor, which receives input data from two suitable analog interface circuits, for example, two Texas Instruments analog interface circuits (AIC) TLC32046. The system provides an expansion port that allows multiple numbers of processor boards or systems to be linked together. This building block feature enables the construction of multichannel multiprocessor adaptive systems. In addition, there is a system ROM, glue logic, and clock circuitry. The Four Channel Unit uses four AIC type chips and one DSP56001 type processor.

The system software is preferably contained in a 2716 2K×8 ROM. When the system is powered up and a reset is executed, the system software will be loaded into the processors' internal RAM The loading sequence is immediately followed by the execution of system software.

The system software preferably includes suitable internal house keeping programs that provide functions such as operator switch controls, switch debounce, system interface configurations, and system initializations. The system preferably provides a watch-dog feature that detects a processor crash such as those typically caused by power supply interruptions. Upon the detection of a processor crash, the watch-dog circuit resets the system and resumes operation without operator intervention. The interface for multiple processors can be any suitable interface to appropriately link the multiple processor.

The software is implemented using any suitable digital signal processor that preferably has at least 512 words of separate program and data memory, such as the Motorola DSP56001 digital signal processor. That is, the DSP56001 processor has three types of internal memory: program memory, X data memory, and Y data memory. The internal program memory may contain 512 words, and each of the internal X and Y memories may contain 256 words, wherein each word has 24 bits.

The present invention cancels noise and enhances speech by utilizing a program which uses approximately 1K of program memory and 2K of each of the X and Y data memories. External memory can be added to the DSP56001 to provide additional memory which may be desired. Three 32K×8 EPROMs may be used to expand the program memory of the DSP56001 in the preferred embodiment from 512 words to 32K. Similarly, two 8K×24 RAM may be used to expand the X and Y memories from 256 words each to 8K words.

The present program resides in program memory locations P:$0000 to P:$03FF. A program memory map is depicted in FIGS. **12A, 12B** and **12C**. On Power Up ("On/Off" switched from "Off" to "On"), the DSP56001 bootstrap program is invoked, which loads the program from the external ROM into the processor. The processor then initializes all relevant data memory to its required values. When this is complete, the LMS program and VOX program are executed. The spectral subtraction algorithm may be executed on a processor like processor **212** (FIG. **13**) in the expansion unit **600** (FIG. **17**).

Therefore, the present apparatus for obtaining the desired noise cancellation and speech enhancement preferably includes at least three devices, which are coupled together in a cascaded manner, that is an LMS device, a spectral subtraction device, and a PTT/VOX device ("VOX device"). As previously discussed, these devices operate in accordance with stored algorithms. The LMS and spectral subtraction devices generally obtain the desired noise cancellation and speech enhancement with the VOX device providing proper performance of the ANC.

The LMS or algorithm is used due to its simplicity and robustness in adaptive filtering applications. Adaptive filters have the ability to adjust their own coefficients automatically, and their design requires little or no a priori knowledge of signal or noise characteristics. As is to be appreciated, while the preferred embodiment uses the LMS device to perform coefficient adaptation, the present invention is not so limited and any other means may be used to perform coefficient adaptation.

FIG. **1** shows a block diagram of an adaptive filter **300** used as an adaptive noise canceler (ANC). The ANC has two inputs: a primary input **310** and a refer-

ence input **320**. The primary input **310** consists of a speech signal corrupted with noise. The reference input **320** consists of noise that is uncorrelated with the speech in the primary input, but is correlated to the noise in that input. The adaptive filter **300** uses the noise in the reference input **320** to estimate the noise in the primary input **310**. The estimated noise is then subtracted from the primary input to produce a noise-free speech signal **330** which is also the input to the spectral subtraction device. The algorithm that is used to update the filter coefficients is preferably the normalized LMS algorithm and may be expressed by the following two equations:

$$e(n)=d(n)-\hat{d}(n) \quad (1)$$

$$wk(n+1)=(1-alpha)wk(n)+Muo*e(n)*u(n-k+1)/\|u(n)\| \quad (2)$$

wherein u(n) is the signal received by the primary input **310**, d(n) is the signal received by the reference input **320**, d (n) is the filter output, wk(n) is the kth filter coefficient at time n, Muo is the step-size parameter, alpha is a predetermined coefficient for enabling the filter coefficient to adapt more quickly in non-stationary noise environments, and $\|u(n)\|$ is the norm of the input signal u(n). The norm is preferably defined as the maximum magnitude input sample of a sample history of the most recent N input samples, where N is the filter order. The adaptive transversal filter that is used preferably has an order of 32.

In the normalized LMS algorithm, instead of using a constant step-size parameter to update the filter coefficients as is done in the LMS algorithm, the step-size parameter is divided by the norm squared of the input signal. One drawback of the LMS algorithm is that the convergence rate of the algorithm decreases as the input signal level decreases. However, by dividing the step-size parameter by the norm of the input signal, the effective step-size is increased when the input signal level decreases. Although the effective larger step-size leads to a faster convergence rate, it also causes relatively larger steady-state error.

The voice operated switch (VOX) device (algorithm) is designed to determine the presence or absence of speech in a high noise environment such as a poorly specified high noise environment.

As a result, the VOX device is able to stop the coefficient adaptation when it detects speech in the reference input. This is especially useful since when a large speech signal is present in the error signal it can cause the filter coefficients to diverge from their optimal values. The digital VOX device **132** is preferably implemented in the software to detect speech when present in the reference input **310**. When there is no speech the device adapts to the input noise and the filter coefficients are updated after each new received sample. When the digital VOX device detects a speech signal, the device (algorithm) stops updating the filter coefficients and uses the current filter coefficients to compute an estimate of the noise present in the reference. Especially when the noise environment does not vary significantly during the period when speech is present in the reference input **310**, the performance of the device is not degraded significantly. A step-size parameter of preferably 0.095 is used with a forgetting factor of 0.999. The forgetting factor allows the VOX device to have a finite memory which is desirable when operating in a non-stationary environment.

The output of the VOX device is a binary valued variable, VUV. If VUV=1, then speech is present; otherwise if VUV=0 speech is considered absent.

The VOX device preferably buffers 196 samples of output speech from the LMS device to the spectral subtraction algorithm. Upon detecting speech, each of the samples are output, but delayed by 196 samples. This buffer value is a preferred baseline, but may be increased or decreased. The purpose of this buffer is to accurately reproduce onsets of the speech waveform.

The VOX device provides a detection probability of 0.99 down to 2 dB signal-to-noise ratio (SNR) of either speech or silence. This is preferably measured with 300 Hz sine waves in Gaussian noise. The VOX uses at least two detectors; a robust distribution-free detector to test magnitudes and a magnitude detector for the first linear prediction coefficient. The magnitude detector maps any distribution of the hypothesis into the uniform distribution and tests for deviations from the uniform distribution. This distribution detector tests for changes in the marginal distribution. The magnitude detector for the first predictor coefficient effectively tests the hypothesis of (local) stationarity against the alternative. Detection decisions are made on the basis of changes in the predictor coefficient over at least approximately 256 samples. As a baseline, the system operates on high-pass filtered speech. Parameters of the VOX device may be varied depending on the application of the adaptive noise cancellation and speech enhancement system.

FIG. **2** depicts a model for the feedback microphone sensor **70**, ear canal **116** and speaker **50**. The model illustrated in FIG. **2** assumes that, as a result of the adaptation process, a signal is produced that is out of phase with the input. The assumption of a slight contraction of the ear volume of the ear canal will give rise to an impedance mismatch between the ear canal and the middle ear; this, combined with a closed off ear canal will cause acoustic reflections to occur. As a first approximation, this is approximated by a weighted sum of two time delay elements **51** and **52** as shown in FIG. **2**. The transfer function of the system may be expressed as in equation 3:

$$\frac{V_o}{V_i}=\frac{m(\omega)}{1+S(\omega)[e^{-j\omega Td_1}+\alpha\, e^{-j\omega Td_2}]} \quad (3)$$

wherein:

V_i=input

V_o=output at speaker

$m(\omega)$=microphone transfer function

$S(\omega)$=speaker transfer function

T_{d2}=time delay

T_{d2}=time delay, which is assumed to be approximately 0

Typically,

$$\frac{\omega T_{d2}}{2}\Big| << \pi \text{ so that}$$

$$\frac{V_o}{V_i}\approx\frac{m(\omega)}{1+e^{-jTd_2/2}\,S(\omega)\left[1+\frac{\omega^2 T_{d2}^2}{4}\right]}$$

The speaker **50** and the sensor **70** can be shown to have a transfer function as represented at block **71** having a zero at the origin and two poles determined by the mechanical and electrical characterizations of the

speaker (see J. White, "*Considerations in High Fidelity Moving Coil Earphone Design*", IEEE Trans. Audio, pp. 188°194 Nov.–Dec. 1963).

Thus, the compensator may include a filter with four poles and three zeroes to fully compensate for all effects. In addition, if the filter transfer function is $H(\omega)$, then $|H(\omega)\ S(\omega)| >> 1$ may be obtained from 0 to 10 kHz so as to insure stability of the compensator. The actual filter parameters can be set by one skilled in the art without undue experimentation.

The spectral subtraction algorithm is preferably implemented by the processor **212** in the expansion unit **101**, as previously described. FIG. **3** depicts the spectral subtraction algorithm device. Input speech samples are input into a buffer (not shown), preferably a **128** sample buffer. These speech samples are Fast Fourier Transformed (FFT'd) by FFT **400** and converted to polar form (magnitude and phase). The phase position is stored in a phase memory **404** and the magnitude portion is supplied to a magnitude squared device **402**, whereupon the magnitude portion is squared and supplied to the PTT/VOX 132. When the push to talk or VOX indicates the absence of speech, the magnitude of the FFT is used to update a noise spectrum model in the device **406**. The update equation is given by equation 4 below:

$$|\hat{\underline{N}}(\omega)| = |\hat{\underline{N}}(\omega)(1-\xi)+\xi G|(\underline{x(\omega)}| - |\hat{\omega}(\omega)|) \quad (4)$$

wherein

$1-\xi$ is the "forgetting factor";

$|\underline{X(\omega)}|$ is the input FFT magnitude;

$|\underline{N(\omega)}|$ is the noise spectrum; and G controls averaging time. The resultant is stored as an estimated noise signal in a estimate noise spectrum device **414**.

Upon the detection of speech, the noise spectrum is subtracted form the speech spectrum in a device for subtracting estimated noise **408**. The resultant is limited so that values more than 23 dB down are clipped to be 23 dB down. In addition, the resulting spectrum magnitude is filtered with a 5 point moving average. When this is completed, the phase function is restored in a restore phase function device **410**, the signal is inverse FFT'd and converted to an analog signal in a device **416**.

The coefficients for the moving average filter may be expressed as in equation 5:

$$\underline{Y}^n = \sum_{h=-2}^{2} \dot{\omega}_h \underline{x}_{n-h} \quad (5)$$

wherein:

$\underline{Y}n$=output of the moving average filter

$\underline{x}$=FFT magnitude and,

$\dot{\omega}_{-2}=\dot{\omega}_{-1}=\dot{\omega}_2=\dot{\omega}_1=0.05$

$\dot{\omega}_0=0.8$

Additional reconstruction constraints can be applied by detecting the number and trajectory of tonals in the noise and removing them based upon predicted trajectories.

In other words, the FFT device **400** or algorithm is used to compute the sample spectra of the noise and speech signals. A push to talk or digital VOX **132** is used to determine speech from noise. The digital VOX or algorithm (code and programming provided below) preferably uses the following test statistics to determine the speech from the noise: sample zero crossings, changes in the number of tonals, energy, and a nonparametric test such as the Kolmogorov-Smirnov Test on the sample data. The Spectral Subtraction processor **130**, in the absence of noise, updates a sample power spectrum of the noise. When speech is detected, the phase is computed from the spectrum, the noise spectrum estimate is subtracted from the speech spectrum, the phase function is restored, and additional reconstruction constraints based upon the number of tonals detected in the noise and the shape of the noise spectrum are used to reconstruct the speech signal with attenuated noise.

FIG. **7** depicts an overall system of the present invention. As noted previously, the present invention is especially well suited for use in a high noise environment, e.g., as part of a communications system such as an intercom, telephone, etc. in any typical high noise environment. The audio interface (see, for example, FIGS. **4, 8, 9, 10**) preferably includes a sensor to sense the ambient noise, a sensor to sense the signal in the ear canal, and a speaker to produce the local anti-noise and received speech, all resident in a headset, and the necessary analog signal conditioning circuitry to insure optimum quantization of the signal in the Digital Signal Processor (DSP). The Digital Signal Processor estimates the noise, and removes an estimate from the speech plus noise. The system configuration in the transmit and receive modes are illustrated in the previously described FIGS. **6** and **5**, respectively.

The Adaptive Noise Canceler (ANC) unit is shown in FIG. **11**. The ANC Output Volume control **120** allows the user to vary the amount of noise cancellation by varying the LMS algorithm Loop gain and by varying the Reconstruction constraints in the Spectra Subtraction Algorithm. The ANC headphones (FIGS. **4** and **8**) contain an in ear sensor **70** acoustically isolated with the use of plastic tubing, the speaker **50** to play the antinoise and speech, and an external sensor **60** to sense the ambient acoustic noise. The in ear sensor **70** also acts as a microphone in the transmit mode. The entire headset is designed with a plastic cover that conforms to the ear canal, providing the acoustic conduction necessary for in ear transmission.

An alternative headset configuration is shown in FIG. **10**. In this diagram the feedback microphone **70'** is not used as a transmit microphone; instead, a boom mike **60''** is used. In this configuration reference sensors **60'** are positioned externally, and boom microphone **60''** can also be used as an additional reference sensor by using standard dual diaphragm noise canceling elements. This configuration allows the user to have an "over the ear" fit instead of "in the ear" if desired.

A third headset configuration **700**, to be used in telephone applications is presented in FIG. **9**. The assumptions of the noise field insure that the signal i(t) will be statistically independent of the noise signal N(t). This fact allows the adaptive filter AF **706** to cancel noise at the input to a linear filter H(s) **710**, which is highly correlated with the signal from the microphone **60**. Microphone **60** can be configured to incorporate noise canceling 2-element microphones wired to subtract common mode signals that might arise from vibration, speech transduced from the headset casing, etc.

The linear filter H(s) **710** acts as a compensator for the responses of the speaker **50** transfer function S(s). Ideally, it would have a transfer function C/S(s); C is a constant chosen so that $|S(s)M(s)H(s)| >> 1$. An adaptive filter AF **708** is provided to account for the fact that

the noise field at the output of a microphone or sensor **760** may be decorrelated from the noise present at the microphone **770** and to account for the fact AF **706** cancels a signal that will be sent through speaker **50**. The delays caused by delays **702** and **704** are added to account for the fact that the signal present at microphone **706** will, in general, be decorrelated from the signal at microphone **60**. As before, the output of AF **708** is sent to a Spectral Subtraction algorithm processor **130** for further speech enhancement; and, the VOX **132** is employed in this embodiment as described above.

FIG. **17** illustrates the adaptive noise cancellation unit **100** and the expansion unit **101**. As previously mentioned, the architecture of the expansion unit is substantially similar to that shown in FIG. **13**.

An additional application of the ANC unit is in the area of ambient noise cancellation for microphones in acoustic surveillance or telemetry. Because of the finite acoustic aperture, even directional microphones will have sidelobes in their spatial response patterns. Interference coming into the sidelobe of a conventional microphone can degrade the signal to a noise ratio and/or reception range of acoustic receivers. An ANC unit **800** (FIG. **16**) can be applied to the problem by using two additional reference microphones **810** and **820** with peak spatial responses at the 3 dB points of a main microphone **830**. In FIG. **16**, the main microphone is used to retrieve the desired signal; reference microphones **810** and **820** are used to adaptively filter any unwanted signals in the sidelobes. FIG. **15** depicts a block diagram of how the microphones are connected to adaptive filters **850** and summed.

As before, the VOX algorithm or signal detection means **860** freezes weights when speech is detected in the main microphone. For telemetry applications, the signal detection means **860** can be a match filter/envelope detector/thresholding device or modem output may be used to produce a decision to freeze the weights.

This microphone configuration as in FIG. **16** is useful for acoustic surveillance, acoustic data transmission and motion sensing.

Having herein described in detail preferred embodiments of the present invention, it is to be understood that the invention defined by the appended claims is not to be limited to particular details set forth in this description as many apparent variations thereof are possible without departing from the spirit or scope of the present invention.

The following Appendix lists the VOX algorithm.

APPENDIX

Pseudocode

Table of Variables for VOX Algorithm

Variable	Initial (Power Up) Value	Definition
VUV	0	=1 if Speech present =0 otherwise
stchng	1	=1 if voicing state "just changed" =0 otherwise
count	0	sample counter
x	-	current "desired signal" input
z	-	high pass filtered, magnitude of speech

Exhibit PA-G

quant(4)	0	sample quantiles of z under hypothesis
qup(4)	0	recursively updated quantiles if no voicing state change
array(4)	0	storage of 4 speech samples for initial quantile estimate
gamma	0.00001	1-Forgetting factor for exponential averager
G	100	Gain Factor for Exponential Averager
u(t)	-	unit step function
test_count	0	counter
tstat	0	test statistic of marginal d.f.
Tspeech	0.05	threshold for speech detection
Tnoise	10.00	Threshold for "non speech" detection

wf0	0	predictor coeff. (p.c.) est.
wf1...wf3	0..0	previous p.c. estimates
ef	0	error for predictor estimator
tpc	0	variable for p.c. test
Thresh_pc	tbd	p.c. test threshold
count_pc	0	p.c. sample counter
xm1	0	previous x value

Vox Algorithm

1. High Pass filter speech w/ 1rst Order IIR filter, fp=800Hz.
2. Take magnitude of filter output (call "z")

; Note "return" implies resume normal operation of program
; without any additional state change.

3.

```
if(stchng=1) then
    stchng=0
    test_count=0
    count=0
    test=0
    tstat=0
    count_pc=0
end if
```

```
count_pc=count_pc+1
ef-x-wf0*xm1
xm1=x
wf0=(1-ap)wf0+g(ap)*ef*x
if(count_pc=128) then
count_pc=0
wf3=wf2
wf2=wf1
wf1=wf0
wf0=0
if(test=1) goto prep_test
 count=count+1
 if(count>132) count=133
 if(count=<4) then
     array(count)=z
     if(count<4) return
     else
     Rank array & store result in quant
end if
end if

      do q=1,4
      quant(q)=(1-gamma)*quant(q)
                +(G*gamma)*u(z-quant(q)) -q*.2)
      qup(q)=quant(q)
      end do

      if(count=132) then
      test=1
      return
      end if
```

```
; Have enough samples to test at this point
prep_test     test_count=test_count+1
              do q=1,4
              tstat=tstat+u(quant(q)-z)
              qup(q)=(1-gamma)*qup(q)
                        +(G*gamma)*u(z-qup(q)) -q*.2)
              end do
              if(test_count=128) then
                   test_count=0
;   with 128 samples we test
              if(VUV=0) then
                   if(tstat<Tspeech) then
                        goto pc_test
                   else
; update quantile estimate
                        do q=1,4
                        quant(q)=.75*quant(q)+.25*qup(q)
                        end do
                        return
                   end if
              else
; check for speech end
              if(tstat>Tnoise) then
                   goto pc_test
                   else
              if(8 or more seconds elapsed since speech onset
                   set to noise only)
               else
; update quantile estimate
                        do q=1,4
```

```
quant(q)=.75*quant(q)+.25*qup(q)
end do
return
end if
end if
end if

pc_test
tpc=abs(wf1-wf3)
wf0=0

if(tpc> Thresh_pc) then
stchng=1
VUV=.not.(VUV)  ; logical "NOT" function
return
end
```

(END OF VOX ALGORITHM)

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What is claimed is:

1. Headset apparatus to be worn by an operator for use in a adaptive signal processing system having first processing means for adaptively processing reference signals supplied thereto so as to form a first processed signal, means for combining the first processed signals and an exterior signal supplied from a microphone used by another operator so as to form an input signal, determining means for determining if a received signal represents speech originating from the operator wearing said headset apparatus and including switch means for supplying the speech signal therefrom if the determination indicates that said received signal represents said speech, and second processing means receiving said speech signal from said switch means for adaptively processing the same so as to remove residual broadband components in said speech signal for supply to another of said operators, said apparatus comprising at least one housing each having first microphone means receiving a reference acoustic signal for producing a reference signal for supply to said first processing means, speaker means receiving said input signal for producing audible information to the operator wearing said headset apparatus, and second microphone means receiving an acoustic sound from within the housing for producing a signal for supply to said determining means.

2. An audio communication system for communication between two or more operators, said system comprising:

headset means worn by one of the operators including at least one housing each having first microphone means for producing a reference signal, second microphone means receiving an acoustic sound from within the housing for producing an internal signal therefrom and speaker means for receiving an input signal to provide audible information to the operator using said headset means;

first processing means for adaptively processing the reference signal received from the first microphone means so as to form a first processed signal;

means for combining the first processed signals from the headset means worn by the operator and an exterior signal supplied from a microphone used by another of said operators so as to form said input signal for said speaker means;

determining means receiving said internal signal from the second microphone means for determining if said internal signal represents speech originating from the operator wearing said headset means, said determining means including switch means for supplying the internal signal therefrom if the determination indicates that said internal signal represents said speech; and

second processing means receiving said speech signal from said switch means for adaptively processing the same so as to remove residual broadband components in said speech signal for supply to another of said operators.

3. An audio communication system for communicating between two or more operators according to claim 2, in which said switch means includes a voice operated switch.

4. An audio communication system for communicating between two or more operators according to claim 2, in which said determining means determines if said internal signal represents said speech by utilizing statistics relating to at least one predetermined factor.

5. An audio communication system for communicating between two or more operators according to claim 4, in which said statistics relate to at least one of number of sample zero crossings, changes in number of tonals, energy level and a predetermined nonparametric factor.

6. An audio communication system for communicating between two or more operators according to claim 2, in which said speech which is received by said second microphone means is an acoustic signal supplied through an eustachian tube contained within said operator using said headset means which is generated when the same operator speaks.

7. An audio communication system for communicating between two or more operators according to claim 2, further comprising means for supplying said internal signal from the second microphone means to said first processing means.

8. Headset apparatus to be worn by a first operator for use in an audio communication system for communicating between two or more operators having determining means for determining if a received signal represents speech from said first operator and for supplying a signal so indicating therefrom, first processing means for adaptively processing said received signal so as to obtain a first processed signal, and second processing means for adaptively processing said first processed signal so as to remove residual broadband components in the speech signal from said first operator for supply to another of said operators, said apparatus comprising a housing having speaker means for providing audible information to said first operator using said headset apparatus, first microphone means receiving an acoustic signal from said first operator using said headset apparatus for producing an output signal for supply to said determining means and said first processing means, second microphone means for producing a reference signal for supply to said first processing means, third microphone means for receiving said audible information for supply to said first processing means, delay means for delaying said output signal from said first microphone means so as to form a delayed signal, and means for receiving an input signal supplied from a microphone used by another operator and for combining the same with said delayed signal for supply to said speaker means.

9. An audio communication system for communicating between two or more operators, said system comprising:

headset means used by one of the operators including a housing having speaker means for providing audible information to said one operator using said headset means, first microphone means for receiving an acoustic signal from said one operator using said headset means, second microphone means for producing an external reference signal, third microphone means for receiving said audible information, and means for receiving an input acoustic signal from another of said operators and for combining the same with said acoustic signal from said one operator for supply to said speaker means;

determining means receiving an output signal from said first microphone means for determining if said output signal represents speech from said one operator and for supplying a signal so indicating therefrom;

first processing means receiving respective output signals from said determining means and said first, second and third microphone means for adaptively

processing the same so as to obtain a first processed signal; and

second processing means for adaptively processing said first processed signal so as to remove residual broadband components in the speech signal from said one operator for supply to another of said operators.

10. An audio communication system for communicating between two or more operators according to claim **9**, further comprising first delay means for delaying the output signal from said first microphone means by a first predetermined amount prior to supply to said determining means.

11. An audio communication system for communicating between two or more operators according to claim **10**, in which said headset means further includes second delay means for delaying said output signal from said first microphone means by a second predetermined amount prior to supply to said means for receiving said input acoustic signal.

12. An audio communication system for communicating between two or more operators according to claim **9**, in which said first processing means includes an adaptive filter.

13. An audio communication system for communicating between two or more operators according to claim **12**, in which said first processing means further includes linear filter means for receiving the output signal from said third microphone means.

14. An audio communication system for communicating between two or more operators, said system comprising:

a plurality of headset means each to be worn by a respective operator, each headset means including a housing having speaker means for providing audible information to the operator using the headset means, first microphone means for receiving an acoustic signal from said operator using the headset means, second microphone means for producing an external reference signal, third microphone means for receiving said audible information, and means for receiving an input acoustic signal from another of said operators and for combining the same with said acoustic signal from said operator using the respective headset means for supply to said speaker means;

determining means receiving an output signal from said first microphone means for determining if said output signal represents speech from said operator using the headset means and for supplying a signal so indicating therefrom;

first processing means receiving each respective output signals from said determining means and said first, second and third microphone means for adaptively processing the same so as to obtain a first processed signal; and

second processing means for adaptively processing said first processed signal so as to remove residual broadband components in the speech signal from said operator using the headset means for supply to the other said operator or operators.

15. An audio communication system for communicating between two or more operators according to claim **14**, further comprising first delay means for delaying the output signal from said first microphone means of each of said headset means by a first predetermined amount prior to supply to said determining means.

16. An audio communication system for communicating between two or more operators according to claim **15**, in which each of said headset means further includes second delay means for delaying said output signal from the respective first microphone means by a second predetermined amount prior to supply to the respective means for receiving said input acoustic signal.

17. An audio communication system for communicating between two or more operators according to claim **14**, in which said first processing means includes an adaptive filter.

18. An audio communication system for communicating between two or more operators according to claim **17**, in which said first processing means further includes linear filter means for receiving the output signal from said third microphone means.

19. An adaptive acoustic receiver apparatus comprising:

first microphone means for receiving an acoustic signal and for supplying a first signal therefrom;

second microphone means for receiving a first reference acoustic signal and for supplying a first reference signal therefrom;

third microphone means for receiving a second reference acoustic signal and for supplying a second reference signal therefrom;

first processing means for adaptively processing said first reference signal so as to form a first processed signal;

second processing means for adaptively processing said second reference signal so as to form a second processed signal;

means for determining a noise signal in accordance with said first and second processed signal and for subtracting said noise signal from said first signal;

means for determining if an output signal from said means for determining and subtracting represents desired data and for supplying a signal so indicating therefrom.

* * * * *

Exhibit PA-J



(12) **United States Patent**
Nemirovski

(10) **Patent No.: US 7,983,433 B2**
(45) **Date of Patent: Jul. 19, 2011**

(54) **EARSET ASSEMBLY**

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(21) Appl. No.: **11/557,775**

(22) Filed: **Nov. 8, 2006**

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(51) **Int. Cl.**
H04R 25/00 (2006.01)

(52) **U.S. Cl.** **381/318**; 381/312; 381/328; 381/151; 381/380; 381/317

(58) **Field of Classification Search** 381/312, 381/318, 326, 328, 325
See application file for complete search history.

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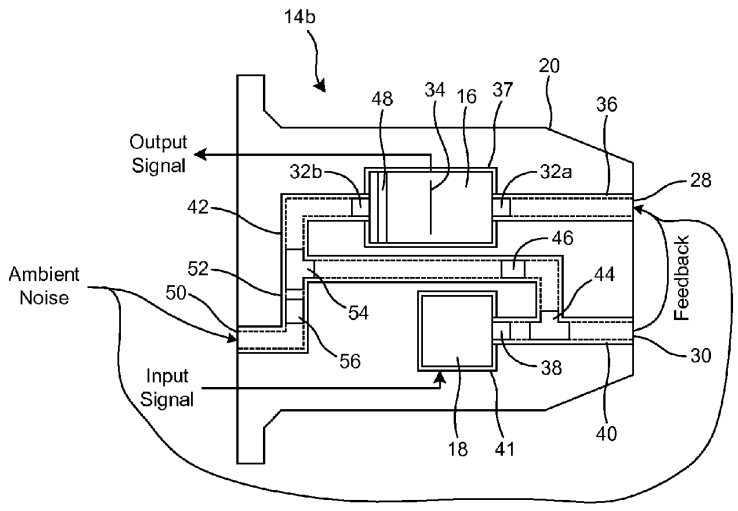
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Primary Examiner — Curtis Kuntz
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(57) **ABSTRACT**

Disclosed is an earset assembly that has a housing having a microphone port and a speaker port. A microphone is enclosed by the housing and has first and second input ports. The first input port is acoustically coupled to the microphone port to detect air pressure changes of the ear of a user. A speaker is enclosed by the housing and has an output port acoustically coupled to the speaker port to broadcast sounds to the user. The output port is acoustically coupled to the second input port of the microphone so that the microphone cancels at least a portion of feedback from the sounds broadcast by the speaker and detected at the first input port of the microphone.

21 Claims, 3 Drawing Sheets



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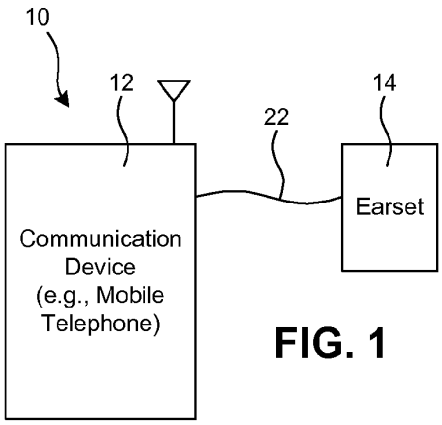


FIG. 1

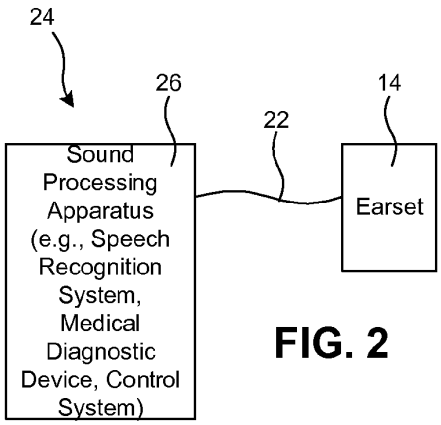


FIG. 2

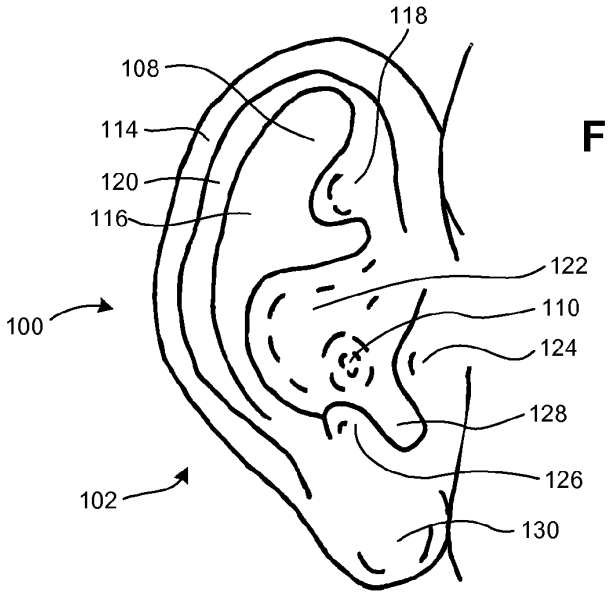


FIG. 3

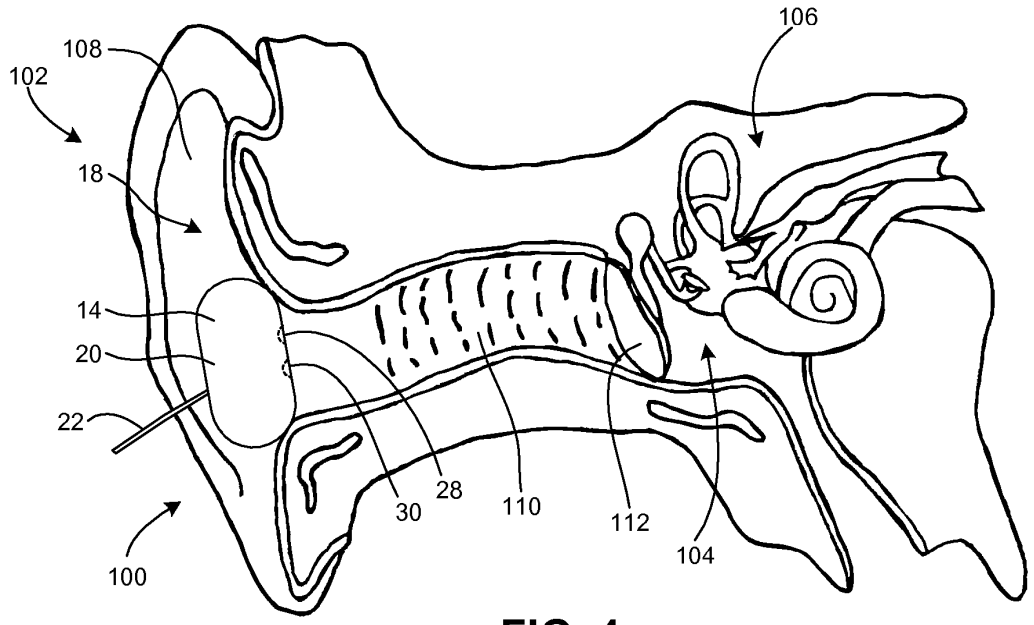

106
108
102
18
14
20
22
30
28
110
112
104
100

FIG. 4

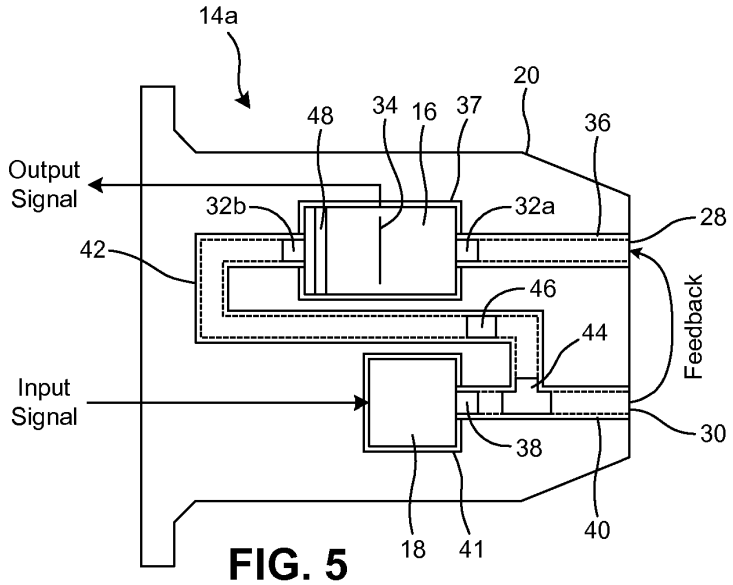

14a
20
48
34
16
37
36
Output Signal
32b
32a
28
42
46
44
Feedback
Input Signal
38
30
18
41
40

FIG. 5

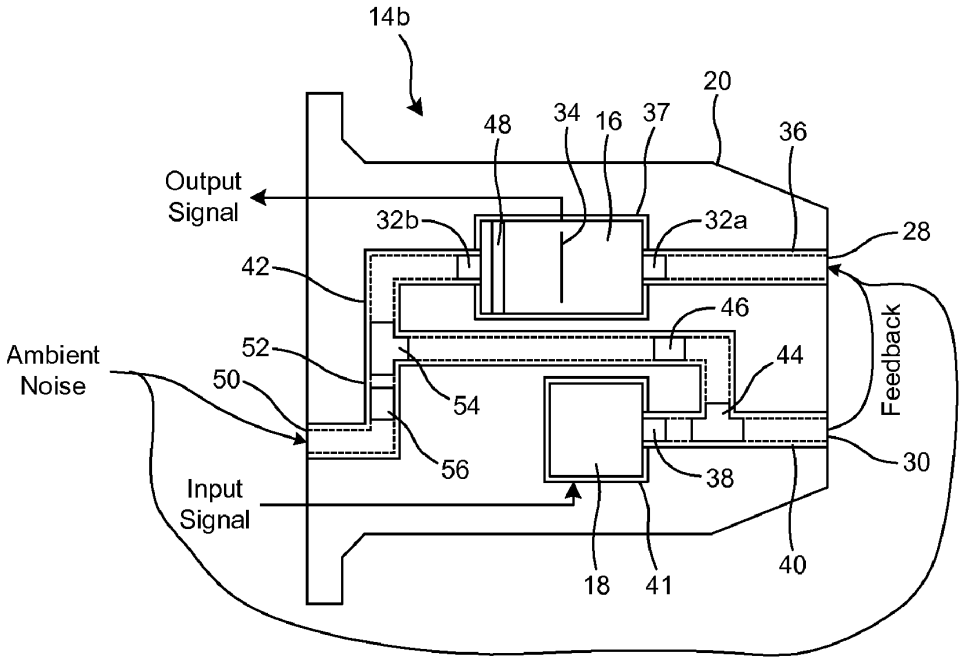

14b
20
48
34
16
37
36
Output Signal
32b
32a
28
42
46
Ambient Noise
52
44
50
54
Feedback
56
38
30
Input Signal
40
18
41

FIG. 6

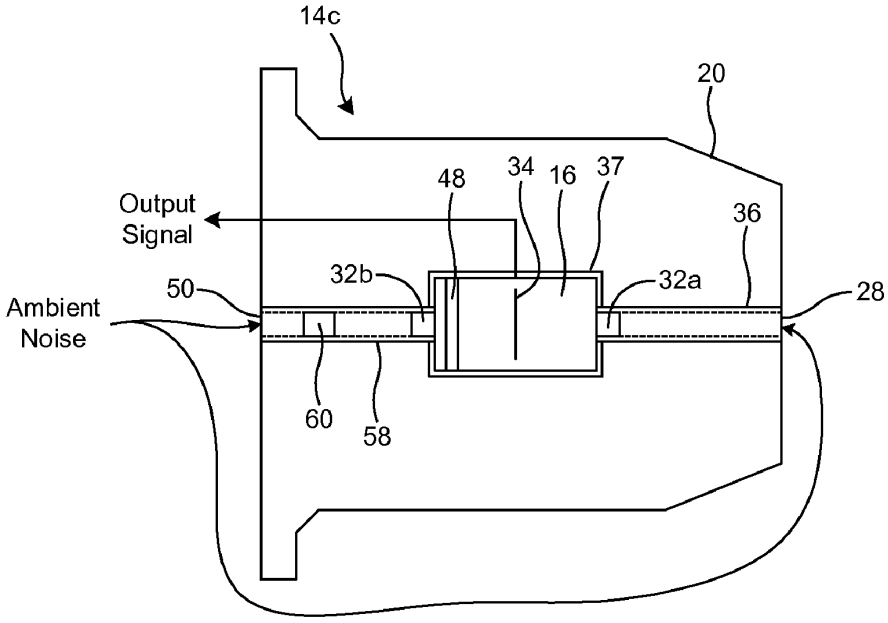

14c
20
48
34
16
37
Output Signal
36
32b
32a
50
28
Ambient Noise
60
58

FIG. 7

EARSET ASSEMBLY

RELATED APPLICATION DATA

This application claims the benefit of U.S. Provisional Patent Application No. 60/734,598 filed Nov. 8, 2005, which is incorporated by reference in its entirety.

TECHNICAL FIELD

The present invention generally relates to a earset assembly having a microphone and a speaker that can be placed with respect to an ear.

BACKGROUND

Wireless mobile telephones, also referred to as a cellular telephones, have become exceedingly popular communication devices. The vast majority of mobile telephones contain a transceiver (e.g., a radio frequency, or RF, transceiver) for establishing a communication link with a remote location, such as a cell phone tower. In order to carry out a conversation with another person using the mobile telephone, the user must hold the telephone adjacent the ear and mouth of the user. This presents the disadvantage of occupying the use of at least one of the user's hands. In many situations, hands free use of the mobile telephone is desirable. The same is true for receivers, or handsets, found in hard-wired telephone systems. For example, the user of a hard-wired telephone system may wish to type on a computer while speaking on the phone. In addition, medical professionals and others have expressed concerns relating to the health of mobile telephone users who engage in prolonged use of an RF transceiver adjacent their head.

There are many commercially available "handsfree" headsets available to users of wireless and/or hard-wired telephone systems. These headsets are intended to assist the user in carrying out a conversation without the use of the user's hands and to locate the telephone away from the user's head. These headsets typically include an ear piece containing a speaker. The ear piece can be removably placed with respect to the user's ear and broadcasts sounds to the user's ear. The headsets also typically include a microphone disposed on a support member that positions the microphone with respect to the user's mouth. The microphone is used to detect speech and other vocalizations emanating from the mouth of the user. The detected sounds are converted into an electrical signal and transmitted by the telephone to a backbone telecommunications network and onto the telephone of another person. In this manner, the user can carry out a fully duplexed conversation with the other person.

However, the headsets can be cumbersome to use. More particularly, care must be taken to ensure that the microphone is properly positioned and that the microphone maintains that position. The need to adjust the headset during a conversation can be distracting to the user. In addition, improper positioning of the microphone may lead to poor and/or unreliable detection of the user's speech. This problem is compounded by the common occurrence of the microphone detecting environmental noise, such as the sound of a passing vehicle, conversations taking place near the user, and the like. The detected environmental noise is ultimately transmitted by the telephone.

Some headsets have suffered from unacceptable levels of feedback from the speaker to the microphone. In addition, many headsets continue to detect a relatively high amount environmental noise. Electrical circuitry has been employed

to address the feedback and/or suppress environmental noise to improve headset performance, but the electrical circuitry often requires a power source and adds complexity and cost to the headset.

Accordingly, there exists a need in the art for an easy to use earset assembly that adequately detects the user's speech and/or other sounds from the user's ear while minimizing the effects of environmental noise and/or feedback from a speaker.

SUMMARY OF THE INVENTION

According to one aspect of the invention, an earset assembly includes a housing having a microphone port and a speaker port; a microphone enclosed by the housing and having first and second input ports, the first input port acoustically coupled to the microphone port to detect air pressure changes of the ear of a user; and a speaker enclosed by the housing and having an output port acoustically coupled to the speaker port to broadcast sounds to the user and the output port acoustically coupled to the second input port of the microphone so that the microphone cancels at least a portion of feedback from the sounds broadcast by the speaker and detected at the first input port of the microphone.

According to another aspect of the invention, an earset assembly includes a housing having first and second microphone ports; and a microphone enclosed by the housing and having a first input port acoustically coupled to the first microphone port to detect air pressure changes of the ear of a user and a second input port acoustically coupled to the second microphone port so that the microphone cancels at least a portion of ambient noise detected at the first input port of the microphone.

BRIEF DESCRIPTION OF DRAWINGS

These and further features of the present invention will be apparent with reference to the following description and drawings, wherein:

FIG. **1** is a schematic block diagram of telecommunications system that includes an earset in accordance with the present invention;

FIG. **2** is a schematic block diagram of a sound processing system that includes an earset in accordance with the present invention;

FIG. **3** is a schematic view of an ear;

FIG. **4** is a cross-sectional view of the ear having an earset disposed with respect thereto;

FIG. **5** is a schematic diagram of an example embodiment of an earset;

FIG. **6** is a schematic diagram of another example embodiment of an earset; and

FIG. **7** is a schematic diagram of yet another example embodiment of an earset.

DESCRIPTION

In the description that follows, like components have been given the same reference numerals, regardless of whether they are shown in different embodiments. To illustrate an embodiment(s) of the present invention in a clear and concise manner, the drawings may not necessarily be to scale and certain features may be shown in somewhat schematic form. Features that are described and/or illustrated with respect to one embodiment may be used in the same way or in a similar way in one or more other embodiments and/or in combination with or instead of the features of the other embodiments.

The present invention is directed to an earset assembly that can be used with, for example, a communication system that allows a user to speak with a remotely located person. As will be discussed below, other possible uses for the earset assembly exist. In one embodiment, the earset assembly includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is arranged to detect sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The speaker is arranged to broadcast sounds to the ear of the user and is arranged with the microphone to reduce the influence of feedback from speaker to microphone. In another embodiment, the earset assembly is arranged to suppress the influence of external sounds, referred to herein as environmental noise or ambient noise. For purposes of the description herein, ambient noise includes sounds generated external to the ear and sounds emanating from the mouth of the user. When used as part of a communication system, the earset assembly allows for separation of a speech input device for a mobile telephone from an RF transceiver of the telephone.

Without intending to be bound by theory, the earset assembly of the present invention allows a user to speak more quietly (e.g., such as at a whisper or near whisper) than is found with conventional headsets. This allows for more private conversations and less disruption to others. There is also a body of evidence indicating that the softer one speaks, the less concentration is needed to maintain the conversation, thereby allowing the individual at least to partially engage in other activities while speaking.

The earset assembly of the present invention does not rely on the detection of sound that has emanated directly from the user's mouth. Therefore, there is a reduced need to repeatedly adjust the position of the earset that would otherwise distract the user and require the use of the user's hands. Also, the size and arrangement of the earset is small, resulting in a more cosmetically appealing device. Such a device can be used unobtrusively. For example, the device would not be noticed as much by others when used in public, or by a person being observed by others, such as, for example, a television news anchor or a secret service agent.

It is noted that the term air pressure changes is used in its broadest sense and includes, for example, sound waves (whether audible to the user or not), pressure fluctuations, vibrations, resonations and the like. In addition, the term air pressure changes as used herein includes vibrations conducted by the bones and tissue of the user that are carried to ear. These conducted vibrations can vibrate the anatomical parts of the ear and/or the housing and lead to sound detection by the microphone. The air pressure changes may be caused by one or more factors, including vibrations of the ear drum, vibrations of bone within the ear, vibrations of other anatomical structures within the ear and vibrations conducted by the bones and/or tissue of the user to the ear and which invoke an air pressure change in and/or emanating from the ear.

As a result, the sensor can be used to detect a person's speech. It is also noted that the term speech is used in its broadest sense and includes spoken words and utterances, as well as other vocalizations produced by the user, including, for example, grunts, whistles, singing, coughs, "clicking" sounds made by movement of the lips or tongue, and the like. To facilitate the description herein, the event of sensing or detecting by the microphone will be referred to as detecting and that which is detected will be referred to as a change within the ear, or simply an air pressure change. The present invention monitors changes within and/or emanating from the human ear which occur instantaneously, or nearly instantaneously, in response to the speech of the person to provide a substantially real-time speech detection system. Other uses for the earset assembly include, for example, a thought detection system, a movement and/or voluntary physical action detection system, a voice recognition system, a medical diagnostic system and so forth. Collectively, these systems will be referred to as sound processing systems. Examples of various communication systems and/or sound processing systems in which the earset assembly described herein can be used are found in co-owned U.S. Pat. Nos. 6,024,700, 6,503,197, 6,47, 368 and 6,671,379, the disclosures of which are herein incorporated by reference in their entireties.

Turning now to the figures, FIG. **1** is a block diagram that illustrates a portion of a communications system **10** for establishing duplexed (two-way) audio communication between two or more individuals. The system **10** includes a communication device **12**, such as a telephone. In the illustrated embodiment, the communication device **12** is a wireless telephone, such as mobile or cellular telephone. The device can establish communication with a communications network (not shown), or backbone network, that enables a user of the device **12** to carry on a conversation with a remotely located person using a remotely located communication device (e.g., another telephone) as is known in the art. It will be appreciated that the communication device **12** and/or any remote communication device (not shown) can be other types of devices, including hardwired (land line) telephones, radios, personal digital assistants (PDAs), portable or stationary computers, voice over internet protocol (VOIP) devices, etc. Also, the communications network can be a network of any type, such as telephone systems, the Internet, a WAN, or a LAN.

The communications system **10** includes an earset assembly, generally referred to by reference numeral **14**. With additional reference to FIGS. **5-7**, the earset **14** can include a microphone **16** and a speaker **18** that are supported by a housing **20**. The physical arrangement and detailed operation of the earset **14** will be described more fully below. The microphone **16** is used to detect sounds in, near and/or emanating from the ear of the user (collectively referred to as air pressure changes of the ear) that result from, for example, speech of the user. The microphone **16** converts those detections into an electrical signal that is input to the communication device **12**. The speaker **18** is used to transmit (i.e., broadcast) sounds to the user. These sounds can include sounds generated in response to signals received by the communication device **12** over the communications network. In this way, the earset **14** and communication device **12** can be used as a bidirectional communication apparatus.

In the illustrated, the earset **14** is coupled to the communication device using an appropriate set of conductors **22**. The conductors can include a wire or wires coupling the microphone **16** to the communication device **12** and the conductors can include a wire or wires coupling the speaker **18** to the communication device **12**. In some configurations, one conductor can be used as a common ground for the microphone **16** and the speaker **18**. In another arrangement, the earset **14** can have a wireless interface with the communication device **12**. For example, a Bluetooth or other appropriate transmitter/receiver arrangement can be used to relay an output signal of the microphone **16** to the communication device **12** and to relay an input signal of the speaker **18** to the earset **14**.

As indicated, the earset **14** can be used with other systems and devices other than the communication device **12**. The general configuration of such a system **24** is shown in FIG. **2**.

For example, the system **24** can include a sound processing apparatus **26** that receives an input signal corresponding to sounds detected by the earset **14** and/or transmits an output signal corresponding to sounds to be broadcast to the user by the earset **14**. The signals can be transmitted over conductor(s) **22** or a wireless link. The sound processing apparatus **26** can include, for example, a logic executing system (e.g., a computer or programmable device) for carrying out a logic routine that processes and/or analyzes the output signal from the earset assembly **14**.

In one example, the sound processing apparatus **26** can be a speech recognition system that converts detected sounds into text. In another example, the sound processing apparatus **26** can be a medical diagnostic system where detected sounds corresponding to the user's heart beat, breathing and/or gastrointestinal system are converted into visual and/or data forms for use by medical professionals. In another example, the sound processing apparatus **26** can be a control system where sounds corresponding to voluntary actions of the user are converted into control instructions for a device, such as a computer, wheelchair, item of machinery, etc. In this embodiment, the sounds can correspond to thoughts of the user as set forth in co-owned U.S. Pat. No. 6,024,700, movements of the user as set forth in co-owned U.S. Pat. No. 6,503,197, or spoken or other vocally generated sounds.

Referring to FIGS. **3** and **4**, an external view and a cross-sectional view of an ear **100** are respectively illustrated. FIG. **4** also schematically shows the earset **14** disposed with respect to the ear. According to Henry Gray's famous text "Anatomy", the human ear is divided into three parts, including the external ear **102**, the middle ear (or tympanum) **104** and the internal ear (or labyrinth) **106**. The middle ear **104** and the internal ear **106** will not be described in great detail herein. The external ear **102** includes an expanded portion, or a pinna **108** (also referred to as an auricle), and an ear canal **110** (also referred to as a meatus or auditory canal). The pinna **108** serves to collect vibrations of the air surrounding the person's head. The ear canal **110** conducts those vibrations to the tympanum, or ear drum **112**.

The pinna **108** has a generally ovoid form with a larger end directed upward and having an outer surface that is irregularly concave and directed slightly forward. The pinna **108** has a number of eminences and depressions. Typically, the ear **100** has a prominent and curved rim, or helix **114**. Generally parallel to the helix **114** is another curved prominence, or antihelix **116**. The antihelix **116** bifurcates to form a triangular depression, or a fossa of the antihelix **118** (also referred to as a fossa triangularis). A narrow, curved depression located between the helix **114** and antihelix **116** is referred to as fossa of the helix, or scapha **120**. The antihelix **116** also curves around a deep, capacious cavity, or the concha **122** (the concha **122** being divided by the commencement of the helix **114**, or crus helicis, into an upper part, termed the cymba conchae, and a lower part, termed the cavum conchae). The concha **122** leads inward to an opening of the ear canal **110**. In front of the concha **122** and projecting backward (usually over the opening of the ear canal **110**) is a pointed eminence, or tragus **124**. Opposite the tragus **124** is a tubercle, or antitragus **126**. A notch-like depression, or incisura intertragica **128**, is disposed between the tragus **124** and antitragus **126**. A lobule **130** is present under the tragus **124** and antitragus **126**.

The ear canal **110** is an oval cylindrical passage extending from a bottom of the concha **122** to the ear drum **112**. The ear canal **110** is about an inch and a half in length when measured from the tragus **124** to the ear drum **112**. When measured from the bottom of the concha **122** to the ear drum **112**, the ear canal is about an inch long. The ear canal **110** forms a gradual "S-shaped" curve and is directed, at first, inward, forward and slightly upward (i.e., pars externa). The ear canal **110** then passes inward and backward (i.e., pars media) and then passes inward, forward and slightly downward (i.e., pars interna).

It is not certain what physical, chemical or neural mechanism causes or generates the changes in air pressure in or near the ear or sounds to some from the ear in response to various actions of the user. However, due to the connection of the oral cavity to the ear via the eustachian tube, speech and movements of the mouth may cause a change in air pressure or an airflow to or from the ear leading to a detectable air pressure change that can be detected by the microphone **16**. Regardless of the exact physical, chemical or neural mechanism, empirical testing has confirmed that the user's speech generates pressure changes in, near or from the ear of the person. Consequently, the air pressure changes can be monitored in or near the ear and used to detect the speech of a user.

The present invention uses various forms of the terms "changes in air pressure", "changes within the ear" and sounds "emanating" or "coming from" the ear in their broadest sense to characterize the parameter being measured. Changes in air pressure may alternatively be characterized as sound waves. These sound waves (or vibrations) may propagate through mediums other than air, such as bone and tissue. As is well known by those skilled in the art, as a sound wave spreads out from its source its intensity falls off (the energy per unit area decreases with the inverse square of the distance), but the total energy is constant.

FIG. **4** illustrates the earset **14** inserted at least partially into the ear **100** of a person (i.e., at least within the cavity defined by the pinna **108**, if not deeper within the ear **100** such as within the concha **122**, at the opening of the ear canal **110** or slightly into the ear canal **110**).

With additional reference to FIG. **5**, the components of an embodiment of the earset **14***a* are schematically illustrated. The earset **14***a* includes the housing **20**. Enclosed by the housing **20** is the microphone **16** and the speaker **18**. The housing **20** can take on a number of different physical configurations. For example, the housing **20** can resemble the housing design of a hearing aid, and particularly a digital hearing aid, for similar insertion, or partial insertion, into the ear **100**. Alternatively, the housing **20** can resemble a miniature earphone as found in conventional wireless telephone headsets or as used with personal audio/music players. The earset **14***a* can be retained by insertion into the ear **100**, by a member disposed over or hanging from the ear and/or by a headset assembly.

The housing **20** can be made from any suitable material, such as plastic, rubber or a gel-like material. In a preferred embodiment, the housing **20**, or portions thereof, is made of relatively rigid plastic, but alternative embodiments can includes making the housing from pliable material, sound absorbing (or sound proofing) material and/or include sound insulating material such as foam. The housing **20** defines a hollow cavity in which the operative components of the earset **14***a* are placed. Voids in the cavity can be unfilled or filled with foam or other material. In another arrangement, the inside surfaces of the housing **20** can be shaped to conform to the components contained therein so that the volume of any unoccupied cavities surrounding the various components is minimized.

The housing **20** is wider than an opening of the ear canal **110** and engages the pinna **108**. In one embodiment, the housing **20** fits within the concha **122** and is retained, at least in part, by the tragus **124** and/or the antitragus **126**. Such arrangement at least partially insulates the portions of the housing **20** that faces the ear canal **110** from externally gen-

erated noise and air pressure changes. However, as discussed in greater detail below, an operative feature of the earset **14***a* can be to allow sound waves originating from locations other than the ear to travel at least in part around the housing **20**.

The housing **20** can be custom designed for the individual to form a close and comfortable fit with the ear of the individual. Alternatively, the housing can have a standard, or "stock", design for all individuals which is fabricated in a number of sizes. As one skilled in the art will appreciate, many alternative configurations for the housing **20** are possible and each are considered to fall within the scope of the present invention.

The earset **14***a* includes a microphone port **28** and a speaker port **30**. The microphone port **28** and the speaker port **30** can be, for example, openings in the housing **20** that are arranged to be placed communicatively with the ear canal **110**, such as adjacent the opening of the ear canal **110**.

The microphone **16**, which can be a unidirectional microphone, includes two input ports **32***a* and **32***b*. For example, the input ports **32***a* and **32***b* can include vibration receptor knobs that capture sound waves for a transducer element, such as a diaphragm **34**, that functions as an operative component of the microphone **16**. The diaphragm **34** converts sound energy into a voltage that serves as the output signal of the microphone **16**. In the illustrated embodiment, the output signal is based on the ratio of the pressure changes in front of the diaphragm to the pressure changes in back of the diaphragm. Although not illustrated, the earset **14***a* can include a preamplifier to amplify the output signal before the output signal is input to the communication device **12** (FIG. **1**) or the sound processing apparatus **26** (FIG. **2**).

Although the ports **32***a* and **32***b* are illustrated as being on opposite sides of the microphone **16**, other configurations are possible. For example, some suitable commercially available microphones have a cube-like configuration with input ports disposed on adjacent side surfaces.

A first of the input ports **32***a* is operatively coupled to the microphone port **28**. In the illustrated embodiment, the coupling is accomplished by a tube **36** made from a suitable polymer material that acoustically and fluidically couples the microphone port **28** with the input port **32***a*. The tube **36** has an inside diameter that, when urged over the knob of the input port **32***a*, forms a secure fit therewith.

The foregoing arrangement allows detection of air pressure changes of the ear, such as sounds eminating from the ear. In particular, sound waves present at the microphone port **28** are communicated to the input port **32***a* via the tube **36**. This arrangement reduces the detection of sound waves other than those present at the microphone port **28** by minimizing a conveyance path to the microphone **16** for such sound waves. Additional isolation of the microphone **16** can be accomplished by encapsulating the microphone **16** in a suitable polymer that conforms to the exterior surfaces of the body of the microphone **16**, referred to herein as coating **37**.

The speaker **18** includes an output port **38** that can include a vibration transmission knob that emits sound waves. The output port **38** is operatively coupled to the speaker port **30**. In the illustrated embodiment, the coupling is accomplished by a tube **40** made from a suitable polymer material that acoustically and fluidically couples the speaker port **30** with the output port **38**. The tube **40** has an inside diameter that, when urged over the knob of the output port **38**, forms a secure fit therewith.

The foregoing arrangement allows transmission of sound waves from the speaker **18** to the ear. For instance, the sounds output at the speaker port **30** can be communicated to the ear canal **110** for reception by the user via the ear drum **112**. In particular, sound waves generated at output port **38** are communicated to the speaker port **30** via the tube **40**. This arrangement reduces the direct communication of sound waves from the speaker **18** to the first input port **32***a* of the microphone **16**. Additional isolation of the speaker **18** can be accomplished by encapsulating the speaker **18** in a suitable polymer that conforms to the exterior surfaces of the body of the speaker **18**, referred to herein as coating **41**.

Although there is no direct communication path for sound waves from the speaker **18** to the first input port **32***a* of the microphone **32***a*, sound waves emanating from speaker port **30** may become present at the microphone port **28** and detected at input port **32***a*. The sound waves from the speaker **18** and detected by the microphone **16** at input port **32***a* will be referred to herein as feedback. Such feedback may be the result of sound waves from the speaker port **30** traveling through the air to the microphone port **28**, inclusive of sound waves reflected by the ear and traveling through any structural members, such as the earset **14***a* and/or the user.

To minimize the presence of feedback in the output signal generated by the microphone **16**, the second input port **32***b* of the microphone **16** is coupled to receive sound waves emitted by the output port of the **38** of the speaker **18**. In the illustrated embodiment, the coupling is accomplished by a tube **42** made from a suitable polymer material that acoustically and fluidically couples the input port **32***b* with the tube **40**. The tube **40** has an inside diameter that, when urged over the knob of the input port **32***b*, forms a secure fit therewith. The tubes **42** and **40** can be joined by fusing or adhering the tubes together or by mechanical fitting, such as a "Y" or "T" connector **44**. The amplitude of the sound waves conveyed by tube **42** can be reduced by an acoustic resistance **46** inserted into the tube **42**. The acoustic resistance **46** can be a metal sleeve (e.g., a tube) filled with appropriate sound dampening material. The acoustic resistance is selected to substantially equalize the pressure at the input ports **32***a* and **32***b* resulting from sound waves generated by the speaker **18**, but not to introduce a propagation delay in the sound waves.

The microphone **16** is configured as a differential device. That is, opposing sound waves of the same magnitude that are respectively detected by the input ports **32***a* and **32***b* will be substantially or fully canceled at the diaphragm **34**. Therefore, it will be appreciated that feedback detected at input port **32***a* can be at least partially canceled by the sound detected at input **32***b*. To improve the degree of feedback cancellation, the magnitude of the sound waves at the respective input ports **32***a* and **32***b* can be equalized. For instance, the value of the acoustic resistance **46** (e.g., in ohms) can be selected to account for a reduction in the amplitude of the feedback component occurring in the feedback path from speaker port **20** to input port **32***a*. If desired, additional acoustic resistance can be used. For example, an acoustic resistance member can be placed in the tube **40**, more than one acoustic resistance member can be placed in the tube **42**, and/or an acoustic resistance member **48** can be placed in the microphone **16** between the input port **32***b* and the diaphragm **34**.

Another technique for improving the degree of feedback cancellation is to account for the propagation delay of the feedback component detected by input port **32***a* relative to the sound waves from the speaker **18** detected at input port **32***b*. For instance, the length of the various tubes **36**, **40** and **42** can be adjusted to maximize cancellation. Both the amount of acoustic resistance and pathway lengths can be adjusted using theoretical modeling of earset **14***a* performance and/or experimental results.

In the schematic representation of the earset **14***a*, the tubes are shown as having square shape bends. It will be appreci-

ated that the actual construction of the earset **14***a* may have tubes with curved bends. For example, the tubes can be made from flexible tubing. In one embodiment, the tubing can have an inner bore diameter of about 0.5 mm to about 3.0 mm.

With additional reference to FIG. **6**, shown is the earset assembly **14***b* configured to cancel feedback in the manner described with respect to the earset **14***a* of FIG. **5** and configured to reduce the amount of ambient noise present in the output signal of the microphone **16**. For the sake of brevity, features in common between the earset **14***a* and the earset **14***b* will not be described in detail.

When the earset **14***b* is placed with respect to the ear **100**, the microphone port **28** is a least partially shielded from ambient noise. For example, the housing **20** and the head of the user at least partially block externally generated sound waves before reaching the microphone port **28**. In a preferred embodiment, the housing **20** does not seal the opening of the ear canal **110** and, as such, some ambient noise propagates around the housing **20**. Therefore, there can be some ambient noise present at the microphone port **28** that is detected by the microphone **16** via input port **32***a*.

To reduce the amount of ambient noise in the output signal of the microphone **16**, the housing can include a second microphone port **50** configured to communicate ambient noise to the second input port **32***b* of the microphone **16**. In the illustrated embodiment, the communication of ambient noises to the second input port **32***b* is accomplished by a tube **52** made from a suitable polymer material that acoustically and fluidically couples the second microphone port **50** with the tube **42**. As a result, an acoustic pathway is formed from microphone port **50** to input port **32***b*. The tubes **42** and **52** can be joined by fusing or adhering the tubes together or by mechanical fitting, such as a "Y" or "T" connector **54**. The amplitude of the sound waves conveyed by tube **52** can be reduced by an acoustic resistance **56** inserted into the tube **52**. The acoustic resistance **56** can be a metal sleeve (e.g., a tube) filled with appropriate sound dampening material.

The second microphone port **50**, which can be an opening in the housing **20**, can be located on an outwardly facing surface of the housing **20** that points generally away from the ear. For example, the second microphone port **50** can be in a generally opposite position on the housing with respect to the first microphone port **28**.

Similar to the way feedback is canceled by the earsets **14***a* and **14***b*, ambient noise can be canceled by the earset **14***b* using the differential qualities of the microphone **16**. For example, ambient noise detected at input port **32***a* can be at least partially canceled by the sound detected at input **32***b*. To improve the degree of ambient noise cancellation, the magnitude of the sound waves at the respective input ports **32***a* and **32***b* can be equalized. For instance, the value of the acoustic resistance **56** (e.g., in ohms) can be selected to account for a reduction in the amplitude of the ambient noise at the first microphone port **28** relative to that at the second microphone port **50**.

Another technique for improving the degree of ambient noise cancellation is to account for the propagation delay of the ambient noise detected at input port **32***a* relative to ambient noise detected at input port **32***b*. For instance, the length of the various tubes **36**, **40**, **42** and **52** can be adjusted to maximize cancellation. Both the amount of acoustic resistance and pathway lengths can be adjusted using theoretical modeling of earset **14***b* performance and/or experimental results.

As indicated, the ambient noise can be considered to include sounds emanating from the mouth of the user, such as a speech. Such sound can travel through the air toward the ear of the user where some of the sound will be present at the second microphone port **50** and will become detected by the microphone **16** via the second input port **32***b*. Also, some of the sound from the user's mouth may pass around the housing **20** and be present at the first microphone port **28**. This sound can be detected by the microphone **16** via the first input port **32***a*. In the manner described above, the sound from the mouth of the user can become at least partially canceled. As will be appreciated, air pressure changes of the ear (e.g., including sounds corresponding to those emanating from the user's mouth but emanating from the ear of the user) will be primarily present at the first microphone port **28** with little or no presence at the second microphone port **50**. As a result, sounds from the ear of the user will be detected by the microphone **16** and represented in the output signal generated by the microphone **16**.

In some systems, it may be desirable to reduce the presence of ambient noise in the signal processed by the system, but there is no need for a speaker to broadcast sounds to the user. For example, a speech recognition system may not have a need for a speaker. As a result, the configuration of the earset **14** can be modified from those shown in FIGS. **5** and **6** for use with systems where a speaker is not needed.

With additional reference to FIG. **7** shown is an earset **14***c* configured with the microphone **16**, but without the speaker **18**. For the sake of brevity, features in common among the earsets **14***a*, **14***b* and **14***c* will not be described in detail. The earset **14***c* includes the microphone port **28** coupled to the first input port **32***a* with the tube **36** as described above. The earset **14***c* includes the second microphone port **50** coupled to the second input port **32***b* with a tube **58**. The tube **58** acoustically and fluidically couples the input port **32***b* with the second microphone port **58** so as to convey sound waves present at the second microphone port **50** to the input port **32***b* of the microphone **16**. The tube **58** has an inside diameter that, when urged over the knob of the input port **32***b*, forms a secure fit therewith.

Similar to the earset **14***b* of FIG. **6**, the earset **14***c* of FIG. **7** can be reduce the amount of ambient noise present in the output signal of the microphone **16** by cancellation of opposing sound waves at the diaphragm **34** of the microphone **16**. The amplitude of the sound waves conveyed by tube **58** can be reduced by an acoustic resistance **60** inserted into the tube **58**. The acoustic resistance **60** can be a metal sleeve (e.g., a tube) filled with appropriate sound dampening material.

Similar to the way ambient noise is canceled by the earset **14***c*, ambient noise can be canceled by the earset **14***c* using the differential qualities of the microphone **16**. For example, ambient noise detected at input port **32***a* can be at least partially canceled by the sound detected at input **32***b*. To improve the degree of ambient noise cancellation, the magnitude of the sound waves at the respective input ports **32***a* and **32***b* can be equalized. For instance, the value of the acoustic resistance **60** (e.g., in ohms) can be selected to account for a reduction in the amplitude of the ambient noise at the first microphone port **28** relative to that at the second microphone port **50**.

Another technique for improving the degree of ambient noise cancellation is to account for the propagation delay of the ambient noise detected at input port **32***a* relative to ambient noise detected at input port **32***b*. For instance, the length of the tubes **36** and **58** can be adjusted to maximize cancellation. Both the amount of acoustic resistance and pathway lengths can be adjusted using theoretical modeling of earset **14***c* performance and/or experimental results.

For each of the earset **14** embodiments, it will be appreciated that the microphone port **28** can be moved closer to or further away from various anatomical structures within the ear **100** as desired for factors such as comfort and to optimize

detection of the user's speech. For most applications, one earset **14** can be sufficient to detect speech and/or other sounds generated by the user. However, two earsets **14** can be used by positioning an earset with respect to each ear of the user.

In an alternative arrangement to the earsets **14** shown in FIGS. **5** to **7**, the microphone **16** can be replaced with two matched microphones. A first of the microphones can be arranged to detect air pressure changes of the ear and a second of the microphones can be arranged to detect sounds external and adjacent the ear. The output of the second microphone can be delayed, such as with an all pass filter. The outputs of one or both of the microphones can be attenuated or amplified, if appropriate, and then combined by effectively subtracting the output of the second microphone from the output of the first microphone, for example. The resulting signal can be used by a communications device or other sound processing system.

Although particular embodiments of the invention have been described in detail, it is understood that the invention is not limited correspondingly in scope, but includes all changes, modifications and equivalents coming within the spirit and terms of the claims appended hereto.

What is claimed is:

1. An earset assembly, comprising:

a housing having a microphone port and a speaker port that are located in physical communication with an ear canal of a user when the earset assembly is retained by an ear of the user;

a microphone enclosed by the housing and having first and second input ports, the first input port acoustically coupled to the microphone port by a first tube so that sound waves present at the microphone port of the housing are communicated to the first input port of the microphone via the first tube so that the microphone detects air pressure changes occurring in the ear canal of the user, wherein the first tube is enclosed by the housing and wherein the coupling of the first input port of the microphone to the microphone port of the housing is closed to an environment surrounding the user; and

a speaker enclosed by the housing and having an output port acoustically coupled to the speaker port by a second tube that is enclosed by the housing to broadcast sounds to the ear canal of the user and the output port acoustically coupled to the second input port of the microphone by a third tube that is enclosed by the housing, is separate from the first tube, and is acoustically joined with the second tube so that the microphone cancels at least a portion of feedback from the sounds broadcast by the speaker and detected at the first input port of the microphone by travel through the second tube, the ear canal and the first tube.

2. The earset assembly according to claim **1**, wherein the microphone is a unidirectional microphone.

3. The earset assembly according to claim **1**, wherein the input ports of the microphone are vibration receptor knobs and the output port of the speaker is a vibration transmission knob.

4. The earset assembly according to claim **1**, wherein the air pressure changes occurring in the ear canal of the user include sound waves emanating from the ear that correspond to speech of the user.

5. The earset assembly according to claim **1**, wherein the acoustic coupling from the output port of the speaker to the second input port of the microphone includes an acoustic resistance.

6. The earset assembly according to claim **1**, wherein the microphone includes an acoustic resistance between the second input port and a transducer element.

7. The earset assembly according to claim **1**, wherein the housing has a second microphone port acoustically coupled to the second input port of the microphone so that the microphone cancels at least a portion of ambient noise detected at the first input port of the microphone.

8. The earset assembly according to claim **7**, wherein the ambient noise includes sounds emanating from the mouth of the user.

9. The earset assembly according to claim **7**, wherein a fourth tube that is enclosed by the housing is used to establish the acoustic coupling of the second microphone port with the second input port of the microphone.

10. The earset assembly according to claim **9**, wherein the fourth tube is acoustically joined to the third tube.

11. A communication system comprising the earset assembly of claim **1** and a telephone having a connection to the earset assembly to receive an output signal from the microphone and to transmit an output signal to the speaker.

12. A sound processing system comprising the earset assembly of claim **1** and a sound processing apparatus having a connection to the earset assembly to receive an output signal of the microphone.

13. The sound processing system according to claim **12**, wherein the sound processing apparatus executes a logic routine to process the output signal.

14. The sound processing system according to claim **12**, wherein the sound processing apparatus is a speech recognition system.

15. The sound processing system according to claim **12**, wherein the sound processing apparatus is a medical diagnostic system.

16. The sound processing system according to claim **12**, wherein the sound processing apparatus is a control system for a controllable device.

17. An earset assembly, comprising:

a housing having a first microphone port that is located in physical communication with an ear canal of a user when the earset assembly is retained by an ear of the user and a second microphone port that is open to an environment surrounding the user; and

a microphone enclosed by the housing and having:

a first input port acoustically coupled to the first microphone port by a first tube so that sound waves present at the microphone port of the housing are communicated to the first input port of the microphone via the first tube so that the microphone detects air pressure changes occurring in the ear canal of the user, wherein the first tube is enclosed by the housing and wherein the coupling of the first input port of the microphone to the microphone port of the housing is closed to the environment surrounding the user; and

a second input port acoustically coupled to the second microphone port by a second tube that is enclosed by the housing so that the microphone cancels at least a

portion of ambient noise from the environment that is detected at the first input port of the microphone.

18. The earset assembly according to claim **17**, wherein the acoustic coupling from the second microphone port to the second input port of the microphone includes an acoustic resistance.

19. The earset assembly according to claim **17**, wherein the air pressure changes occurring in the ear canal of the user include sound waves emanating from the ear that correspond to speech of the user.

20. The earset assembly according to claim **17**, wherein the ambient noise includes sounds emanating from the mouth of a user.

21. A sound processing system comprising the earset assembly of claim **17** and a sound processing apparatus having a connection to the earset assembly to receive an output signal of the microphone.

* * * * *

Exhibit PA-1449

SUBSTITUTE 1449 INFORMATION DISCLOSURE STATEMENT	ATTY. DOCKET NO.	APPLICATION NO.
	0027071.00133	To Be Assigned
	APPLICANT	
	Staton Techiya, LLC	
	FILING DATE	GROUP
	Herewith	To Be Assigned

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EXAMINER	DATE CONSIDERED

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

*Copies of the listed references are either submitted herewith or were previously cited by or submitted to, the Office in a prior application. Pursuant to 37 C.F.R. § 1.97(d) and MPEP §609, the indicated reference may have been previously cited by or submitted to, the Office in a prior application, where the prior application is identified by its U.S. Application Number in this Information Disclosure Statement.

Exhibit PA-1449

SUBSTITUTE 1449 INFORMATION DISCLOSURE STATEMENT	ATTY. DOCKET NO.	APPLICATION NO.
	0027071.00133	To Be Assigned
	APPLICANT	
	Staton Techiya, LLC	
	FILING DATE	GROUP
	Herewith	To Be Assigned

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EXAMINER	DATE CONSIDERED

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

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Exhibit PA-D

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(54) **MICROPHONE WITH IMPROVED SOUND INLET PORT**

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(57) **ABSTRACT**

A microphone comprises a housing defining an inner volume and including a first exterior surface with an aperture leading to the inner volume. The microphone includes a transducing assembly within the housing for converting sound into an electrical signal. A sound inlet plate defines, typically in combination with the first exterior surface, a passageway for transmitting sound to the aperture The passageway receives the sound from an opening in the sound inlet plate. The opening is offset from the location at which the aperture is positioned on the exterior surface. The sound inlet plate is made very thin so that it does not extend substantially away from the housing.

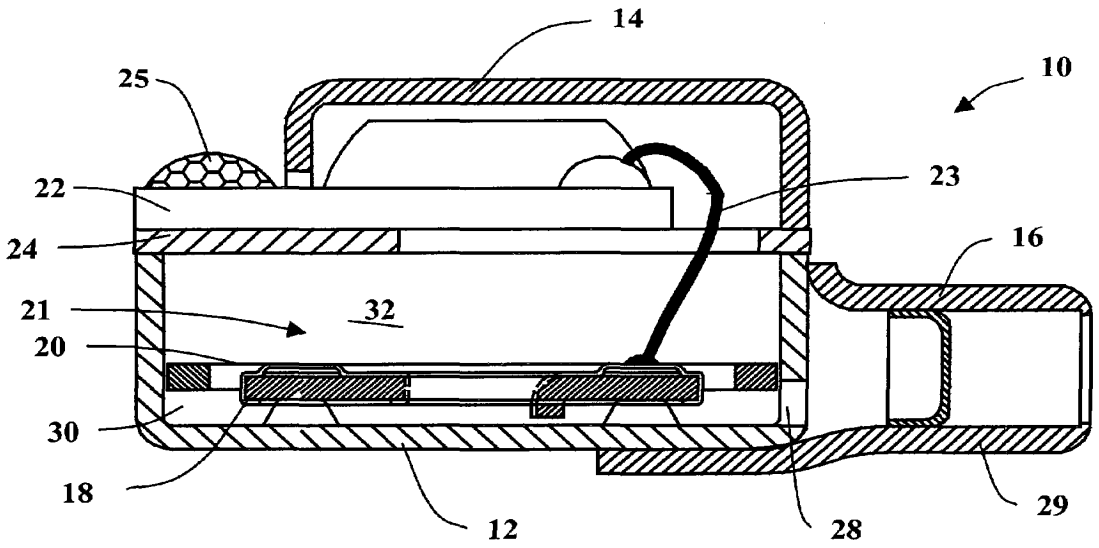


Exhibit PA-D

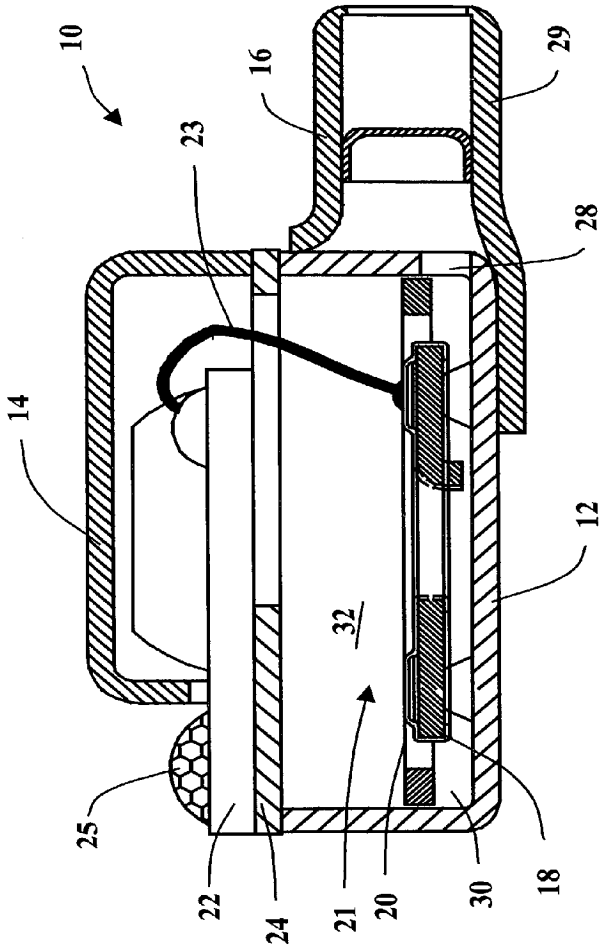


FIG. 1A

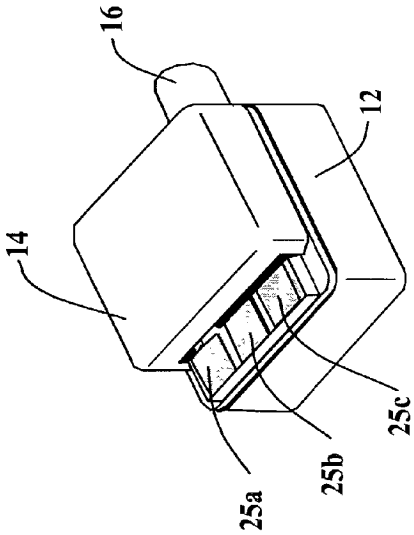


FIG. 1B

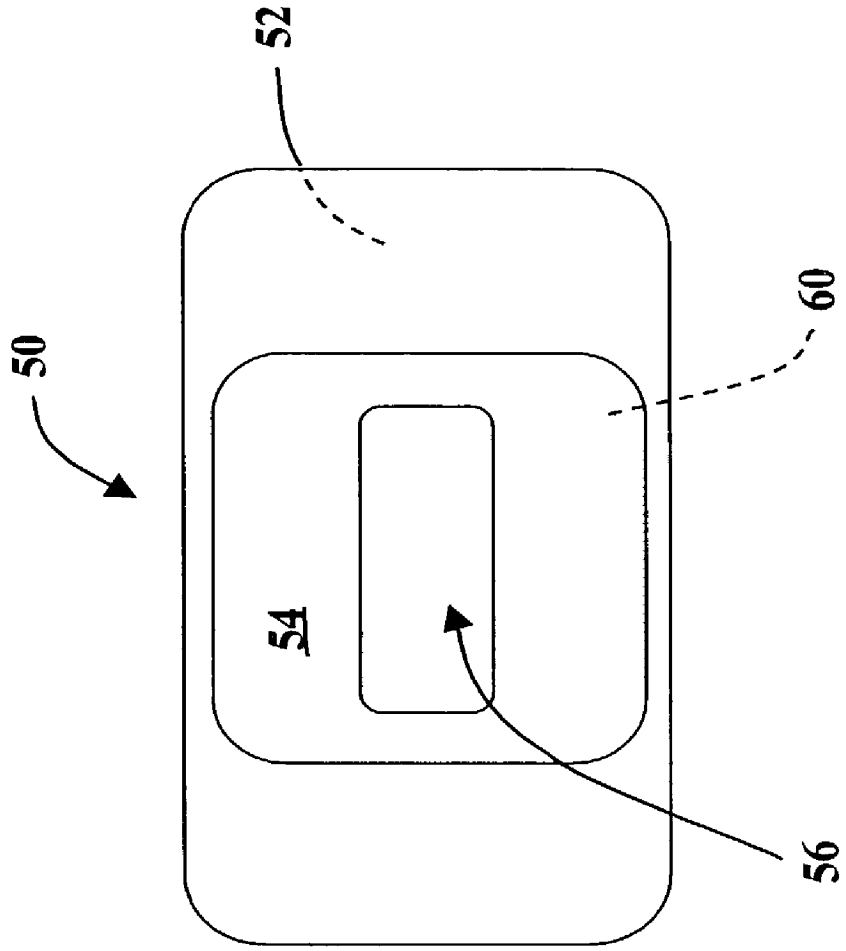


FIG. 2A

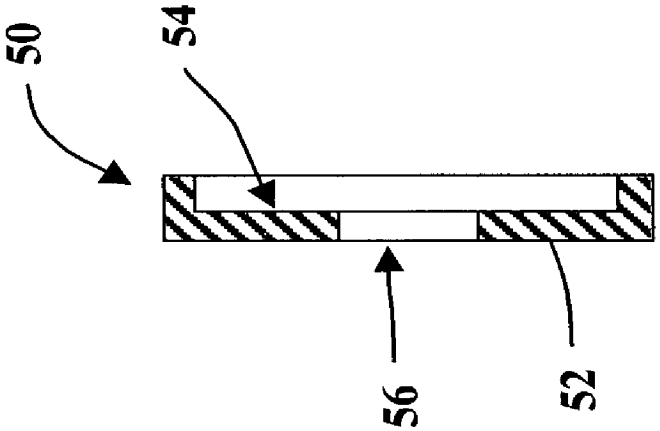


FIG. 2B

Exhibit PA-D

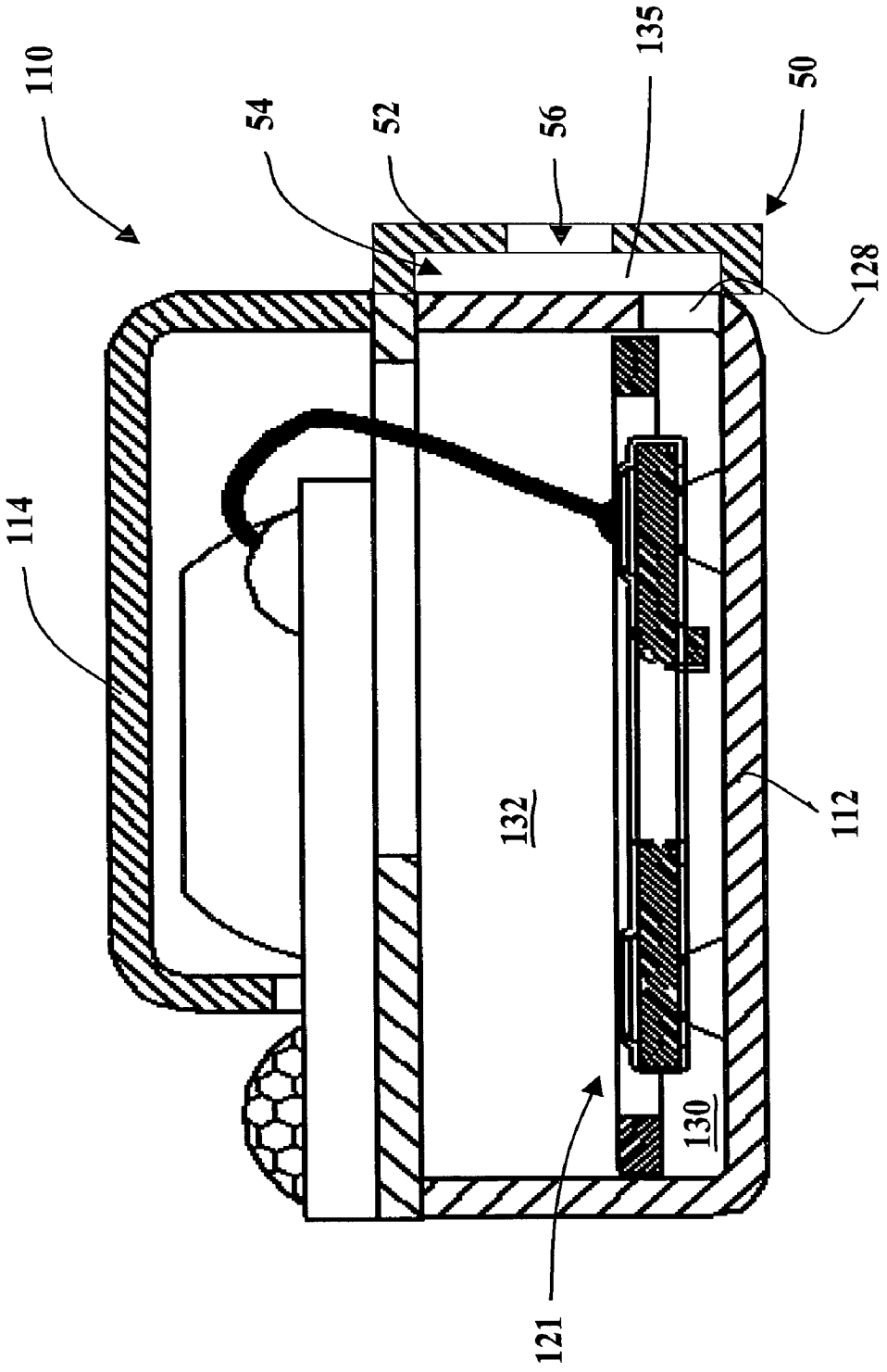


FIG. 3

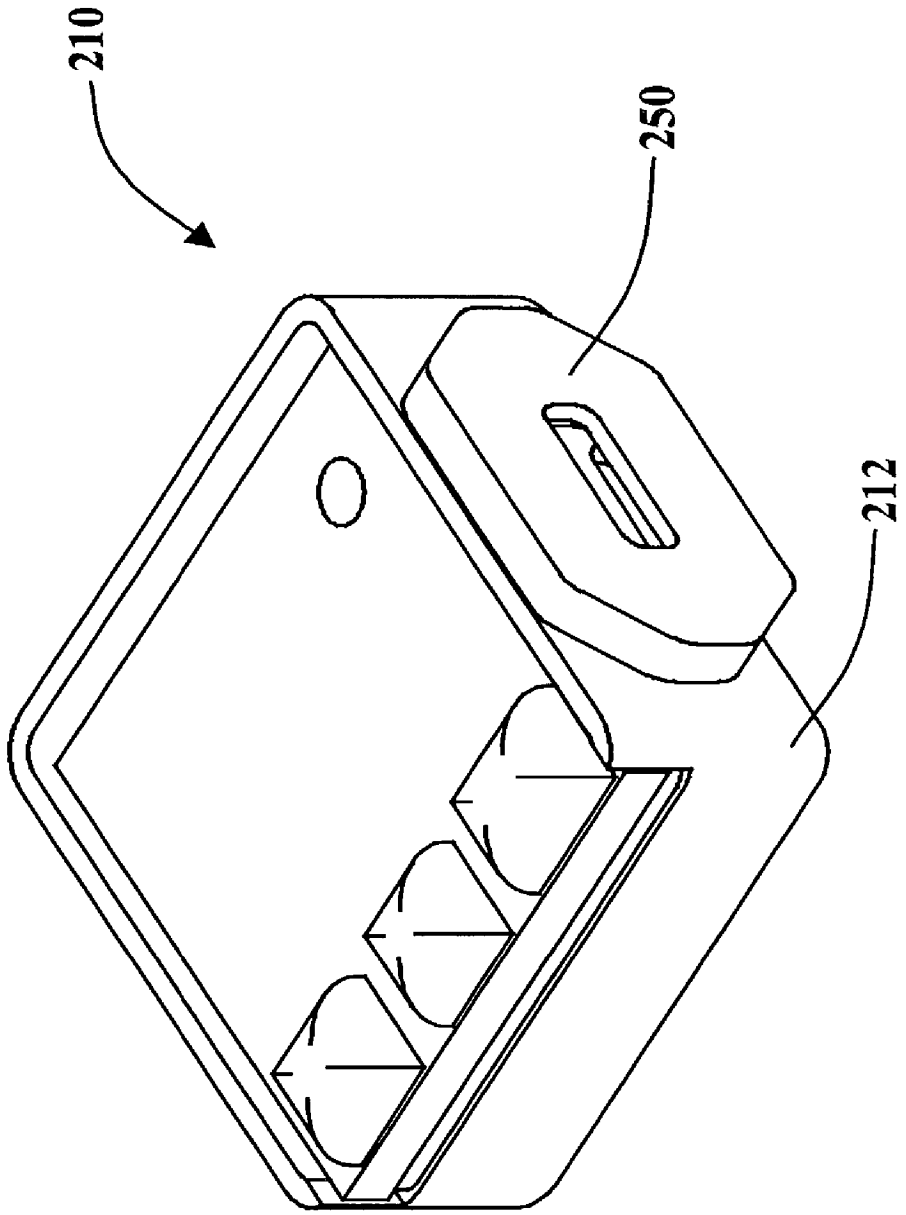


FIG. 4

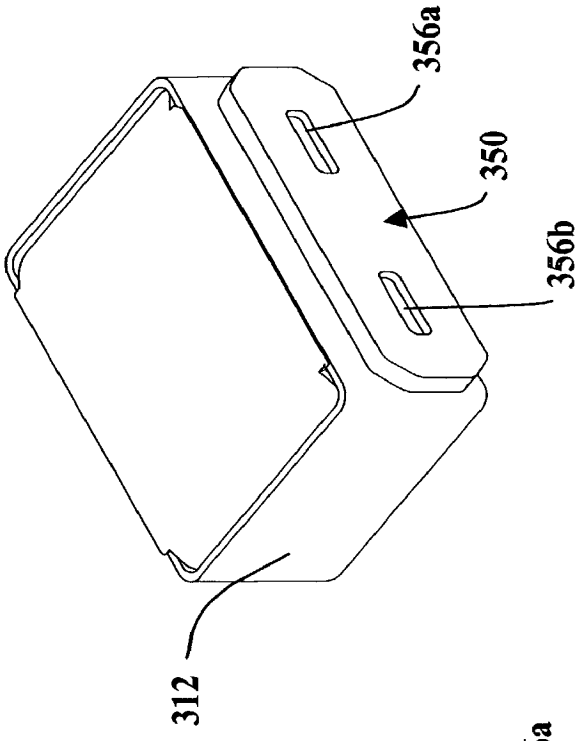


FIG. 5B

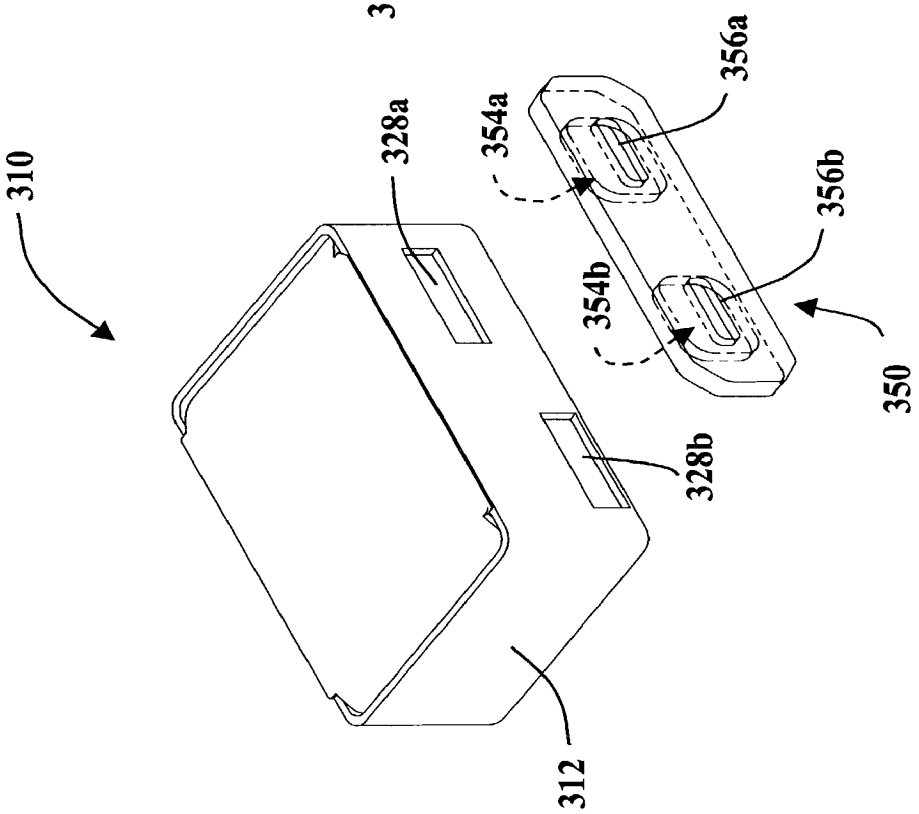


FIG. 5A

MICROPHONE WITH IMPROVED SOUND INLET PORT

FIELD OF THE INVENTION

[0001] The present invention relates generally to electroacoustic transducers and, in particular, to a microphone or listening device with an improved sound inlet port.

BACKGROUND OF THE INVENTION

[0002] Miniature microphones, such as those used in hearing aids, convert acoustical sound waves into an audio signal, which is processed (e.g., amplified) and sent to a receiver of the hearing aid. The receiver then converts the processed signal to acoustical sound waves that are broadcast towards the eardrum. In one typical microphone, a moveable diaphragm and a charged backplate convert the sound waves into the audio signal. The diaphragm divides the inner volume of the microphone into a front volume and a rear volume Sound waves enter the front volume of the microphone via a sound inlet.

[0003] Most prior art microphones, such as the prior art microphone of **FIG. 1**, have a sound inlet that includes a large inlet nozzle for receiving sound from the ambient environment. The large dimensions of the inlet nozzle can be a problem because hearing aids often have very limited space.

[0004] Further, the front volume and back volume within the microphone housing are typically of different sizes, causing the inlet nozzle, which is placed near the front volume, to be located asymmetrically on one of the exterior surfaces Mounting a microphone having an inlet nozzle asymmetrically located on its exterior surface can be problematic in some types of hearing aids because the inlet nozzle must be aligned with the hearing aid's opening to the ambient environment in the hearing aid while the microphone is positioned in a spatially constrained location.

SUMMARY OF THE INVENTION

[0005] The present invention solves the aforementioned problems by providing a novel sound inlet plate that mates with the microphone The microphone comprises a housing with an inner volume and a first exterior surface with an aperture leading to the inner volume The microphone includes a transducing assembly within the housing for converting sound into an electrical signal

[0006] The inventive sound inlet plate is mounted on the first exterior surface and defines (possibly in combination with the first exterior surface) a passageway for transmitting sound to the aperture leading to the inner volume. The passageway receives the sound from an opening in the sound inlet plate, the opening being offset from the location at which the aperture is positioned on the first exterior surface. The sound inlet plate is made very thin so that it does not extend substantially away from the housing. Further, the location of the opening of the sound inlet can be offset to a more desirable position (e.g., the midpoint of the microphone exterior surface) to ease installation of the microphone in the hearing aid.

[0007] The inventive sound inlet plate is useful on omnidirectional and directional microphones

[0008] The above summary of the present invention is not intended to represent each embodiment, or every aspect, of the present invention. This is the purpose of the Figures and the detailed description which follow.

BRIEF DESCRIPTION OF THE DRAWINGS

[0009] The foregoing and other advantages of the invention will become apparent upon reading the following detailed description and upon reference to the drawings.

[0010] **FIGS. 1A and 1B** illustrate a prior art microphone.

[0011] **FIG. 2A** is a front view of a sound inlet plate according to the present invention

[0012] **FIG. 2B** is a cross-sectional view of the sound inlet plate of **FIG. 2A**.

[0013] **FIG. 3** is a cross-sectional view that illustrates the plate of **FIG. 2** mounted on a microphone.

[0014] **FIG. 4** is an isometric view of a sound inlet plate mounted on a microphone according to another embodiment of the present invention.

[0015] **FIGS. 5A and 5B** illustrate another embodiment of the present invention wherein a sound inlet plate is used on a directional microphone.

[0016] While the invention is susceptible to various modifications and alternative forms, specific embodiments have been shown by way of example in the drawings and will be described in detail herein. It should be understood, however, that the invention is not intended to be limited to the particular forms disclosed. Rather, the invention is to cover all modifications, equivalents, and alternatives falling within the spirit and scope of the invention as defined by the appended claims.

DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

[0017] FIGS. **1A-1B** illustrate a typical prior art microphone **10** The microphone **10** includes a case or housing **12**, a cover **14**, and a sound inlet nozzle **16** through which sound enters the housing **12**. Within the housing **12**, a backplate **18** having a charged electret layer works in conjunction with a moveable diaphragm **20** to convert (i.e., transduce) the sound into an electrical signal. The combination of the backplate **18** and the diaphragm **20** is generally referred to as an electret-type transducing assembly **21**, although the present invention is useful with other types of transducing assemblies, as well

[0018] A printed circuit board **22** is mounted on a mounting plate **24** The signal from the transducing assembly **21** is sent to the printed circuit board **22** via a wire connection **23** The signal is processed on the printed circuit board **22** (e.g. amplified) to produce an output signal Because only a portion of the printed circuit board **22** is covered by the cover **14**, one of the set of contacts **25***a* **(FIG. 1B)** on the printed circuit board provides the output signal from the microphone **10**. The other two contacts **25***b*, **25***c* **(FIG. 1B)** are a ground connection and an input power connection, respectively.

[0019] In this typical prior art microphone **10**, sound reaches the interior of the housing **12** via the sound inlet nozzle **16** and the aperture **28**. The inlet nozzle **16** may have

a screen **29** to provide dampening and to serve as a shield for keeping foreign objects from entering the housing **12** Within the housing **12**, the sound propagates through a front volume **30** and acts upon the diaphragm **20**, which separates the front volume **30** from a back volume **32**. The diaphragm **20** moves relative to the backplate **18** in response to the sound causing the backplate **18** to generate the electrical signal corresponding to the pressure change associated with the sound

[0020] One embodiment of the present invention is disclosed in **FIGS. 2A and 2B**. A cup-shaped sound input plate **50** includes an outer wall **52** and an interior recess **54**. A sound port **56**, which is an opening in the outer wall **52**, leads to the interior recess **54**. The interior recess **54** is configured to be large enough to overlap with an aperture **60** (shown in dashed lines) in the housing of a microphone leading to the transducing assembly, like the aperture **28** in **FIG. 1A** Because the plate **50** is designed for miniature microphones, the largest dimensions of the plate **50** are on the order of millimeters. By way of example only, the plate **50** in **FIG. 2** may have a width of about 2 mm to 3 mm and a height of only about 1 mm. The overall thickness is less than 0.5 mm and the interior recess **54** has a depth that is between 0.1 mm and 0.2 mm. The area of the sound inlet port **56** is generally less than about 0.5 mm^2 and typically about 0.2 mm^2 to about 0.3 mm^2.

[0021] **FIG. 3** illustrates the sound inlet plate **50** of **FIG. 2** mounted on a microphone **110**, which is substantially identical to the microphone **10** of **FIG. 1**, but includes 100-series reference numerals. The microphone **110** includes a housing **112** and a cover **114** for the housing **112**. The housing **112** includes an aperture **128** that transmits sound to the transducing assembly **121**, which divides the interior of the housing **112** into a front volume **130** and a back volume **132**. The sound inlet plate **50** can be welded to the housing **112** or attached via an adhesive or glue.

[0022] In operation, the sound inlet plate **50** receives sound through the sound inlet port **56** in its exterior wall **52**. The sound propagates through a passageway **135** that is defined by the interior recess **54** of the plate **50** and the exterior wall of the housing **112** adjacent to the aperture **128**. Eventually, the sound is transmitted through the aperture **128** and acts upon the transducing assembly **121**.

[0023] Unlike prior art systems where the sound inlet extends substantially away from the housing of the microphone (such as the nozzle **16** in **FIGS. 1A and 1B**), the sound inlet plate **50** only protrudes slightly away from the housing **112**. One design aspect leading to the minimal protrusion feature of the plate **50** is the fact that the sound port **56** is simply an opening or hole in the outer wall **52**. For example, in one embodiment, the plate **50** protrudes less than 0.5 mm from the housing **112** and, preferably, about only 0.3 mm from the housing **112**. Relative to the housing **112**, the thickness of the plate **50** is usually less than approximately four times the wall thickness of the housing **112** and, preferably, only about twice the wall thickness of the housing **112**, as shown in the cross-sectional views of **FIGS. 2B and 3**. Thus, the microphone **110** has a more compact design as compared with the prior art microphone **10** of **FIG. 1**.

[0024] Another benefit of the design of the sound inlet plate **50** is that it can be designed to provide a sound passageway leading from a hearing aid sound receptacle that is offset from the aperture **128** in the housing **112**. In other words, the hearing aid's sound receptacle receiving sound from the ambient environment may not be in alignment with the aperture **128** in the housing **112**. By locating the sound port **56** at a point on the exterior wall **52** of the sound inlet plate **50** that is in alignment with the hearing aid's sound receptacle, the interior recess **54** and the exterior surface of the housing **112** immediately adjacent thereto define an appropriate passageway **135** leading to the aperture **128**. In sum, the sound port **56** can be vertically and/or horizontally offset from the aperture **128** in the housing **112**.

[0025] Further, because the performance of some microphones dictate that the front volume **130** be a much smaller size than the back volume **132**, the aperture **128** is usually near a corner of a surface of the housing **112**, substantially offset from the central region on the exterior surface of the housing **112**. Thus, a sound inlet plate **50** can be selected for a particular microphone **110** so as to locate the sound inlet port **56** in the central region of the microphone **110**, providing more symmetry to the location of the sound inlet relative to that exterior surface of the microphone **110**. This can facilitate easier orientation of the microphone **110** while it is being mounted within the hearing aid.

[0026] A further benefit is that a manufacturer of microphones may need only one style of a sound inlet plate **50** for one or more types of microphones. The manufacturer can then maintain a large inventory of such plates **50** that lack the sound inlet port **56**. Once a design specification or order is received from a hearing aid manufacturer dictating the offset of the sound inlet port **56** relative to the aperture **128**, the manufacturer can then form the sound inlet ports **56** in the plates **50** at the appropriate position in the exterior wall **52**.

[0027] Additionally, the sound inlet plate **50** can be designed to have an acoustic inertance that helps to dampen the peak frequency response of the microphone **110**. This can be accomplished by locating the sound inlet port **56** at a certain location relative to the aperture **128** and/or by providing a specific configuration to the interior recess **54**. For example, instead of the recess **54** having the shape of a rounded rectangle, as shown in **FIG. 2**A, it could have an "S" shape, "C" shape, or any other type of shape that creates an elongated, narrow passageway leading to the aperture **128** in the housing **110**. This passageway(s) can be in series or in parallel when leading to the aperture **128**, so as to have a specific effect on the overall frequency response of the microphone **110** Alternatively, the depth of the interior recess **54** can be modified as well to affect the frequency response, and possibly be variable along the passageway that leads from the sound inlet port **56** to the aperture **128** of the housing **112**. Further, although the size of the sound inlet port **56** is shown in the illustrative embodiment as being approximately the same size as the aperture **128** in the housing **112** (or aperture **60** in **FIG. 2**), the size of the sound inlet port **56** can be altered, as well. And, like the prior art microphone **10** in **FIG. 1**, the interior recess **54** may include a dampening material, such as a screen

[0028] The exterior of the sound inlet plate **50** can have various shapes to accommodate different microphones **110** to which it is mounted. For example, **FIG. 4** illustrates an alternative microphone **210** having a housing **212**, but

lacking a cover (like cover **114** in **FIG. 2**) The microphone **210** has a sound inlet plate **250** that has more of a flattened, ovular shape, which is different from the rounded rectangular shape of the plate **50** in **FIG. 2A**.

[**0029**] Additionally, the sound inlet plate may extend over two or more exterior housing surfaces such that the sound inlet port on the plate is adjacent to an exterior surface on the housing (or cover) that is perpendicular to the exterior surface on the housing where the aperture leading to the front volume is located. Further, the sound inlet plate may define the sound passage by itself (i e., the housing does not assist in defining the passageway) by including an interior wall opposite the exterior wall **52** that includes the sound port Such an interior would contact the housing of the microphone. And, while the present invention has been described with respect to a microphone, it can be used on other electroacoustic transducers, such as a receiver.

[**0030**] **FIGS. 5A and 5B** illustrate the invention in conjunction with a directional microphone **310**. The directional microphone **310** includes a housing **312** with two apertures **328***a*, **328***b* for passing sound into the inner volume on both sides of the transducing assembly. A sound inlet plate **350** includes two recesses **354***a*, **354***b* for placement over respective ones of the apertures **328***a*, **328***b*. Two sound ports **356***a*, **356***b* lead into respective ones of the apertures **328***a*, **328***b*.

[**0031**] The sound inlet plate **350** can be designed to increase or decrease the spacing between the sound ports **356** without changing the spacing between the apertures **328** to affect the performance of the directional microphone **310**. Additionally, the plate **350** can be formed around a plurality of exterior surfaces on the housing **312** (e.g., having an "L" shape while fitting on two exterior surfaces). And, the recesses **354***a*, **354***b* can be independently designed to attain a certain (and different, if so desired) acoustical characteristic (e.g., acoustical inertance) in each recess **354**. Finally, the plate **350** can be replaced by two independent plates, each of which leads to a corresponding one of the two sound inlet ports **356**

[**0032**] While the present invention has been described with reference to one or more particular embodiments, those skilled in the art will recognize that many changes may be made thereto without departing from the spirit and scope of the present invention Each of these embodiments and obvious variations thereof is contemplated as falling within the spirit and scope of the claimed invention, which is set forth in the following claims.

What is claimed is:

1. A microphone, comprising:

a housing defining an inner volume and including a first exterior surface with an aperture leading to said inner volume,

a transducing assembly dividing said inner volume into a front volume and a rear volume, said transducing assembly for converting sound into an electrical signal; and

a plate coupled to said first exterior surface of said housing and including a sound inlet port, said plate defining a passageway between said sound inlet port and said aperture for transmitting said sound to said front volume in said housing, said sound inlet port being an opening in a wall of said plate and being offset from said aperture in said first exterior surface

2. The microphone of claim 1, wherein said sound inlet port is vertically offset from said aperture in said first exterior surface

3. The microphone of claim 1, wherein said sound inlet port is horizontally offset from said aperture in said first exterior surface

4. The microphone of claim 1, wherein said passageway is also defined by said first exterior surface

5. The microphone of claim 1, wherein said sound inlet port has an area that is approximately the same as an area of said aperture.

6. The microphone of claim 1, wherein said wall of said plate having said sound inlet port is the primary flat surface of said plate

7. The microphone of claim 1, wherein said plate protrudes away from said first exterior surface of said housing by a distance that is less than four times a wall thickness of said housing.

8. The microphone of claim 7, wherein said plate protrudes away from said first exterior surface by about 0 3 mm

9 The microphone of claim 1, wherein said plate includes an interior recess that defines said passageway.

10. The microphone of claim 9, wherein said interior recess has a generally rectangular shape.

11 A microphone, comprising

a housing defining an inner volume and including a first exterior surface with an aperture leading to said inner volume;

a transducing assembly within said housing for converting sound into an electrical signal, and

a plate attached to said first exterior surface over said aperture and including a sound inlet port, said plate defining a passageway between said sound inlet port and said aperture for transmitting said sound to said inner volume, said passageway having a portion that transmits said sound in a direction generally parallel with said exterior surface.

12. The microphone of claim 11, wherein said aperture is near a corner of said housing and said sound inlet port is offset from said aperture toward a central region of said housing

13. The microphone of claim 11, wherein said passageway defined by said plate is elongated.

14. The microphone of claim 11, wherein said passageway is also defined by said first exterior surface.

15. The microphone of claim 11, wherein said plate has a cup shape.

16 The microphone of claim 15, wherein said cup shape has a base wall and said sound inlet port is in said base wall

17 The microphone of claim 11, wherein said microphone is a directional microphone and said housing includes a second aperture, said plate including a second sound inlet port and defining a second passageway leading from said second aperture to said second sound inlet port.

18 The microphone of claim 11, wherein said plate protrudes away from said exterior surface of said housing by a distance that is less than about 0.5 mm.

19 The microphone of claim 11, wherein said sound inlet port is on a surface of said plate that is generally parallel with said exterior surface.

20 The microphone of claim 11, wherein said plate is designed so that said passageway has an acoustical inertance for achieving a selected dampening of a frequency response of said microphone

21. An electroacoustic transducer, comprising:

a housing defining an inner volume and including a first exterior surface with an aperture therein;

a transducing assembly within said housing for transducing between an acoustic signal and an electrical signal; and

a plate located on said first exterior surface and over said aperture, said plate, in combination with said first exterior surface, defines a passageway for transmitting sound between said aperture and an opening in said plate that is offset from said aperture.

22. The electroacoustic transducer of claim 21, wherein said electroacoustic transducer is a microphone and said plate is a sound inlet plate.

23. The electroacoustic transducer of claim 21, wherein said plate has a cup shape.

24 The electroacoustic transducer of claim 21, wherein said transducing assembly includes a backplate and a movable diaphragm

25 The electroacoustic transducer of claim 21, wherein said plate protrudes away from said exterior surface of said housing by a distance that is less than about 0.5 mm

26 The electroacoustic transducer of claim 21, wherein said opening is on a surface of said plate that is generally parallel with said exterior surface.

27. The electroacoustic transducer of claim 21, wherein said plate includes an interior recess that defines said passageway and said opening in said plate leads to said interior recess

28 The electroacoustic transducer of claim 27, wherein said electroacoustic transducer is a microphone and said recess is designed so that said passageway has an acoustical inertance for achieving a selected dampening of said frequency response for said microphone.

29. The electroacoustic transducer of claim 28, wherein said recess is of a generally rectangular shape

30. The electroacoustic transducer of claim 28, wherein said passageway defined by said recess is elongated

31. A directional microphone, comprising

a housing defining an inner volume and including a first exterior surface with two apertures leading to said inner volume,

a transducing assembly dividing said inner volume into a front volume and a rear volume, said transducing assembly for converting sound into an electrical signal; and

a plate system coupled to said first exterior surface of said housing and including two sound inlet ports each corresponding to a respective one of said two apertures, said plate system defining two passageways each located between respective ones of said sound inlet ports and said apertures for transmitting said sound to said inner volume in said housing, said sound inlet ports being openings in at least one wall of said plate system, at least one sound inlet port being offset from said respective one of said two apertures in said first exterior surface.

32. The microphone of claim 31, wherein said plate system includes interior recesses that define said passageways.

33 The microphone of claim 31, wherein said plate system protrudes away from said exterior surface of said housing by a distance that is less than about 0.5 mm

34 The microphone of claim 31, wherein said plate system is designed so that said passageways have an acoustical inertance for achieving a selected dampening of a frequency response of said microphone.

35 The microphone of claim 31, wherein said plate system includes two independent plates.

36 A method of assembling a microphone, comprising:

positioning a transducing assembly within a housing of said microphone, said housing including an aperture through which sound passes leading to said transducing assembly, and

attaching a sound inlet plate to said housing, said sound inlet plate including an interior recess positioned over said aperture and an exterior wall, said exterior wall having an opening leading into said recess, said opening being vertically or horizontally offset from said aperture.

37. The method of claim 36, wherein said sound inlet plate protrudes from said housing by a distance of less than 0.5 mm.

38. A method of altering the frequency response of a microphone, comprising.

selecting a sound inlet plate having a recess that defines a sound passageway leading to an aperture in a housing of said microphone, said sound inlet plate having an opening in a wall leading to said passageway, said passageway having a geometry resulting in a certain acoustic inertance; and

attaching said sound inlet plate to said housing.

39. The method of claim 38, wherein said sound inlet plate protrudes from said housing by a distance of less than 0.5 mm

* * * * *

Exhibit PA-F

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Beasley (43) **Pub. Date: Jul. 13, 2006**

(54) **HEADSET AUDIO BYPASS APPARATUS AND METHOD**

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(21) Appl. No.: **11/328,890**

(22) Filed: **Jan. 10, 2006**

Related U.S. Application Data

(60) Provisional application No. 60/642,842, filed on Jan. 10, 2005.

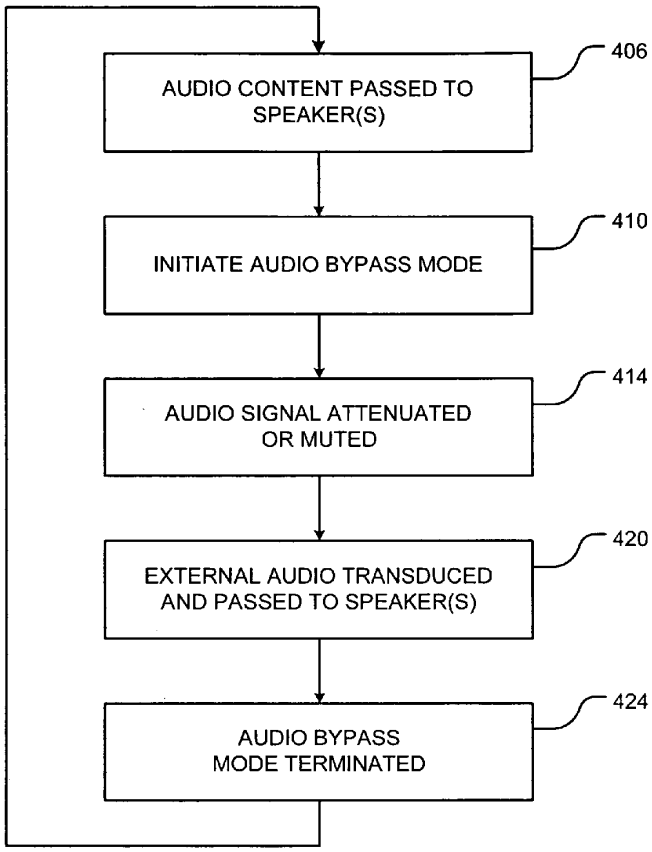
Publication Classification

(51) **Int. Cl.**
H03G 3/20 (2006.01)
H03G 9/00 (2006.01)

(52) **U.S. Cl.** **381/57**; 381/102

(57) **ABSTRACT**

An apparatus and method provides control over the relative level of audio and background sound for an audio device such as a headset or headphones. An input sound transducer can be provided and configured to convert ambient acoustical pressure into an electrical background signal representing the background sound. Noise cancellation circuitry create a cancellation signal that is the inverse of the electrical background signal. A summing component is coupled to the noise cancellation component and combines the cancellation signal with an audio signal representing audio program content and provides the combined electrical signal to an output sound transducer. A controller is coupled to the noise cancellation component and is configured to control the level of a cancellation signal relative to the audio signal, thereby controlling the mix of cancellation signal and audio signal reaching the output sound transducer.



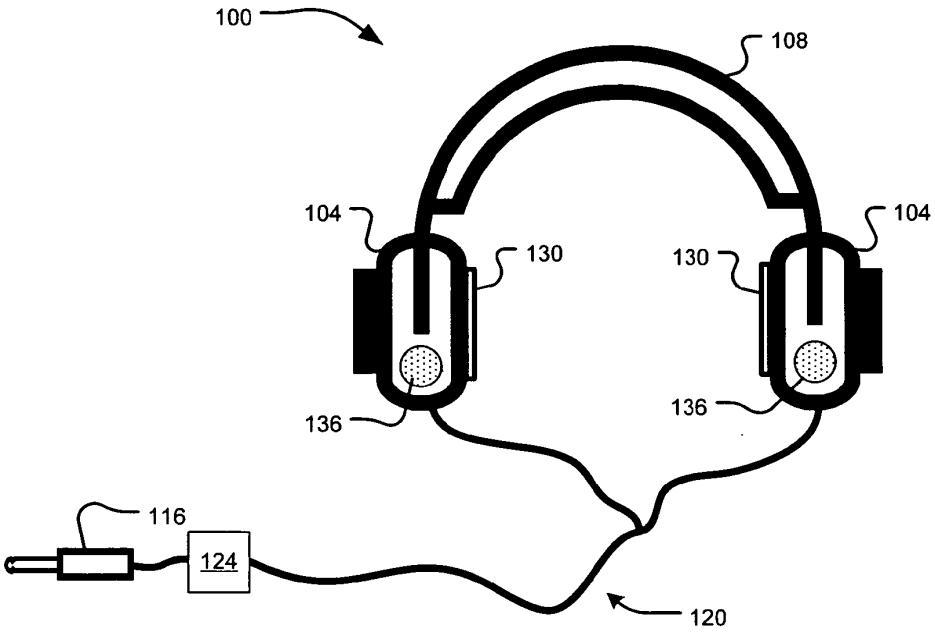

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Fig. 1

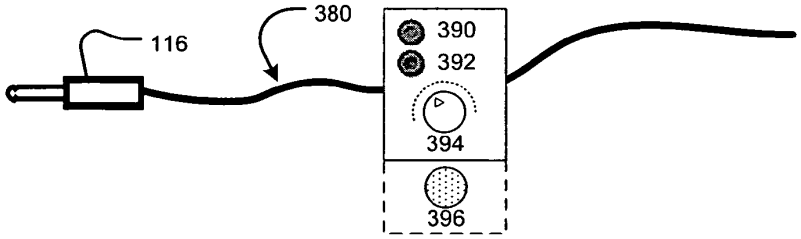

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Fig. 11

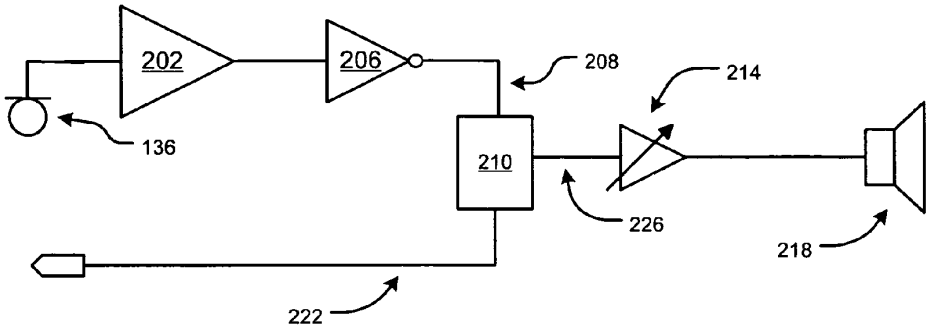

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Fig. 2

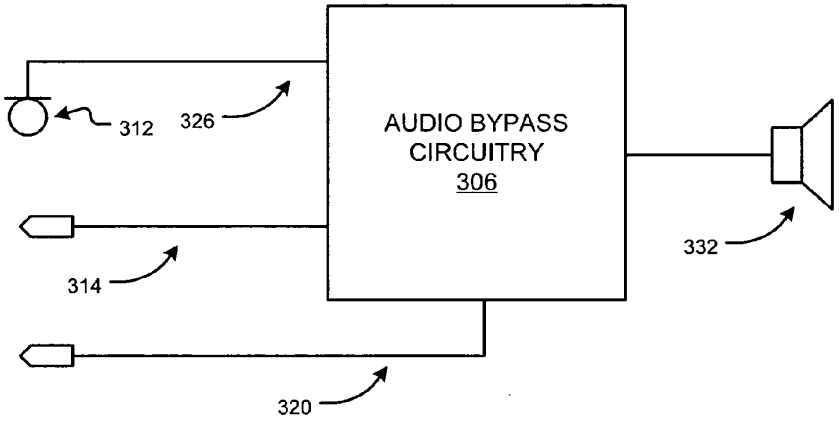

312
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AUDIO BYPASS
CIRCUITRY
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Fig. 3

Exhibit PA-F

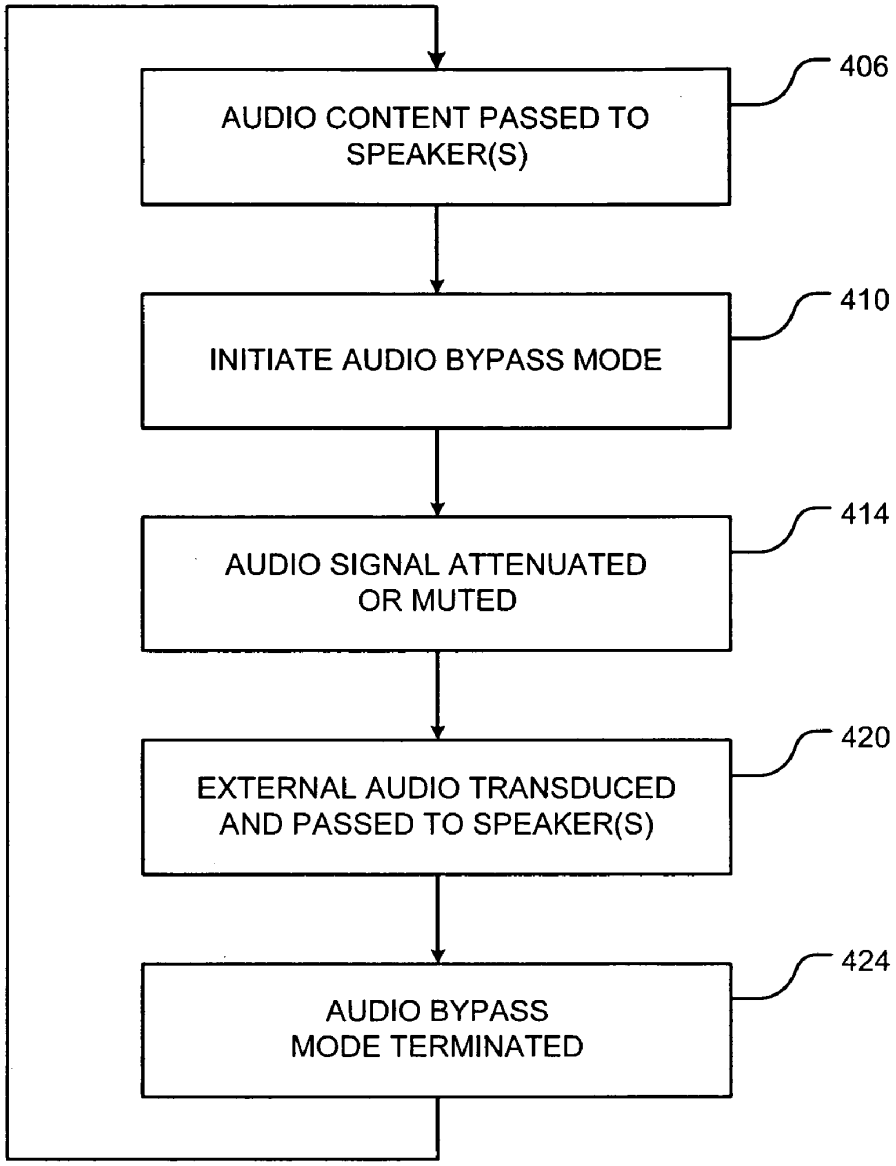


Fig. 4

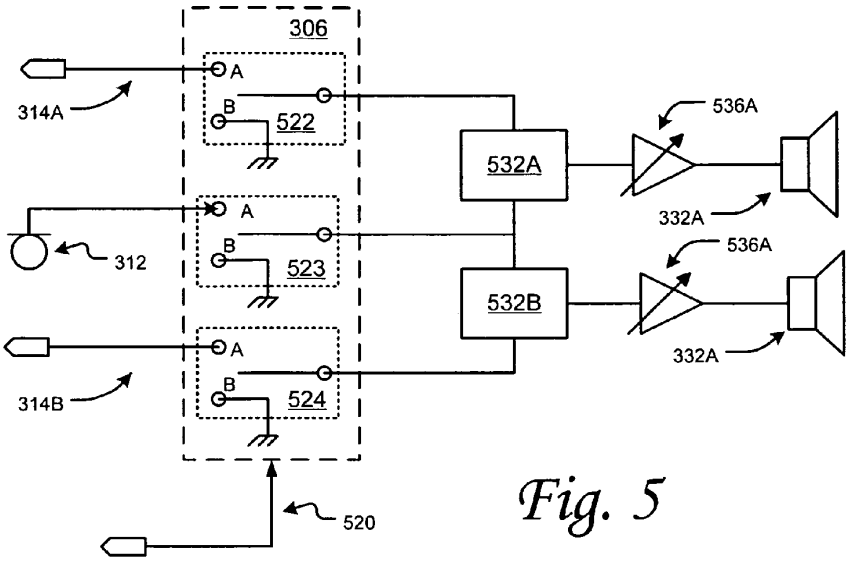

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520

Fig. 5

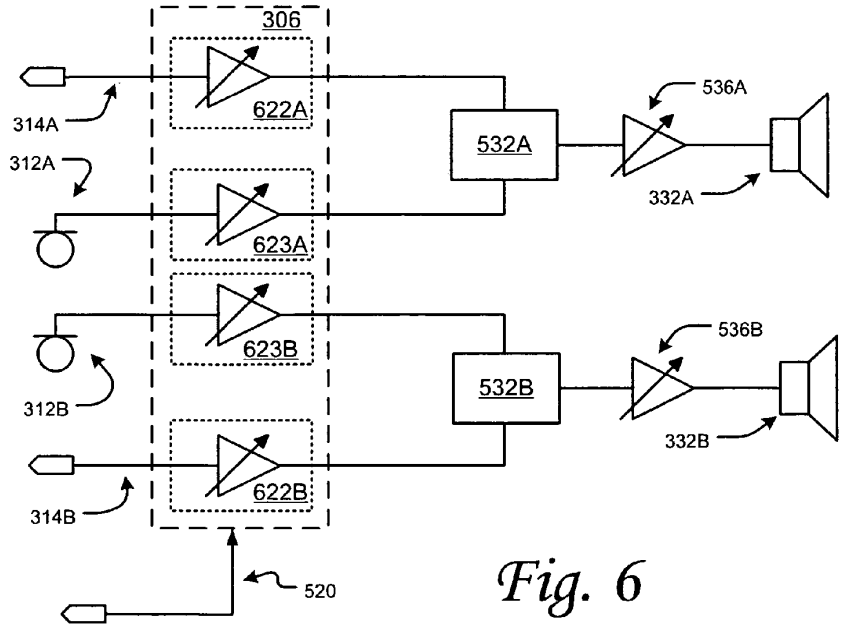

306
622A
314A
312A
532A
536A
332A
623A
623B
536B
532B
312B
332B
622B
314B
520

Fig. 6

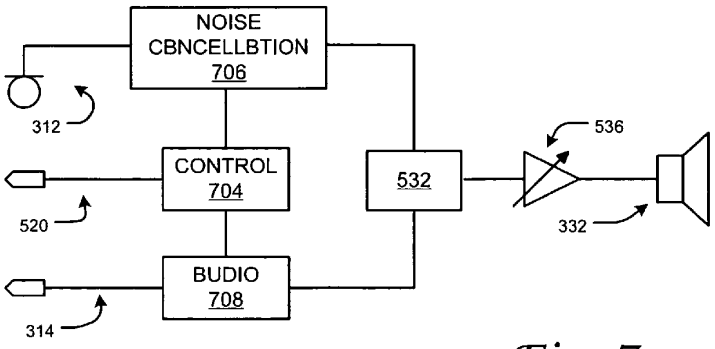

NOISE
CBNCELLBTION
706
312
CONTROL
704
520
BUDIO
708
314
532
536
332

Fig. 7

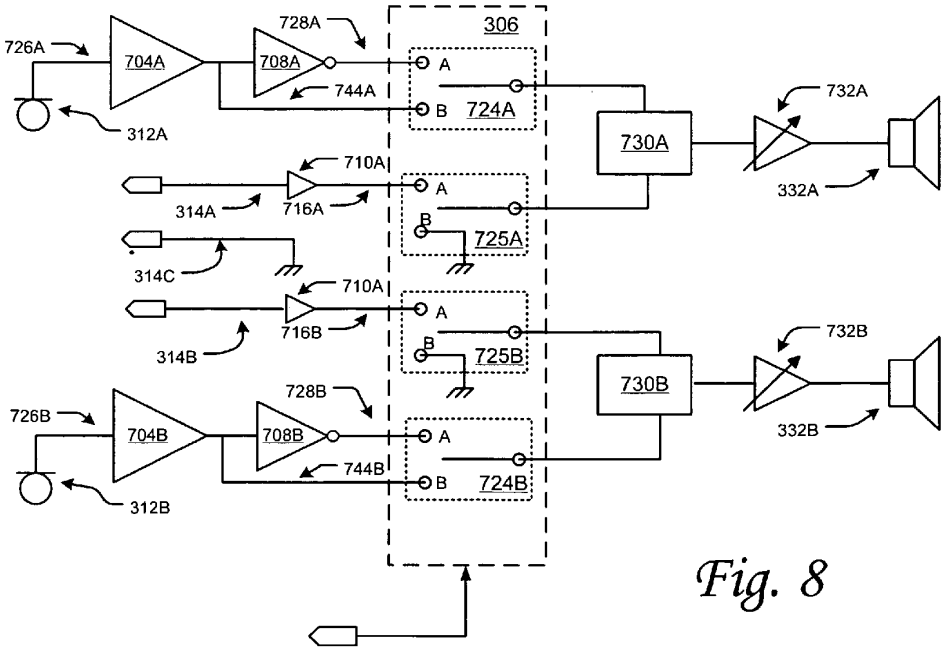

306
726A
704A
728A
708A
744A
A
B
724A
312A
730A
732A
332A
710A
314A
716A
725A
314C
716B
314B
725B
732B
332B
726B
704B
728B
708B
744B
724B
312B

Fig. 8

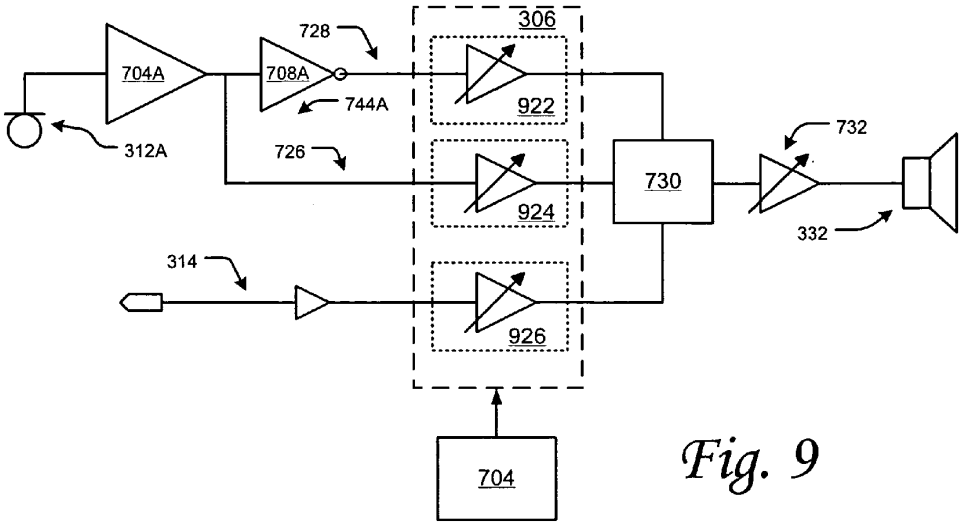

306
728
704A
708A
744A
922
312A
726
730
732
924
332
314
926
704

Fig. 9

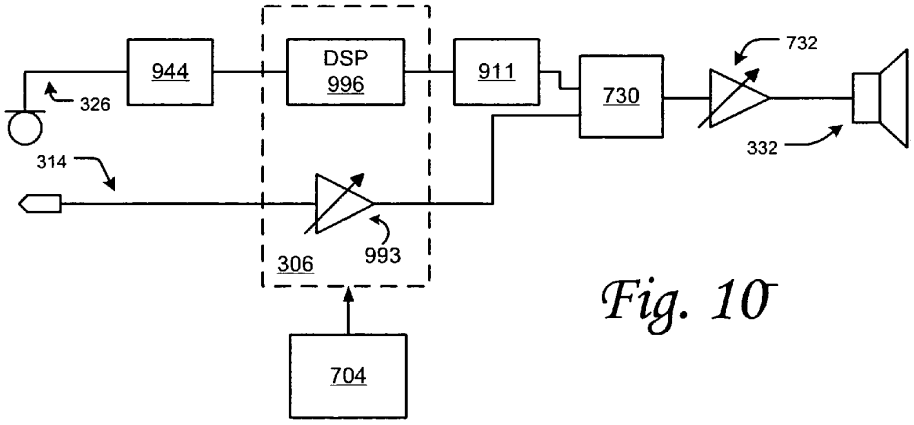

944
DSP
996
911
730
732
326
332
314
306
993
704

Fig. 10

HEADSET AUDIO BYPASS APPARATUS AND METHOD

[0001] This application claims priority to U.S. Patent Provisional Application Ser. No. 60/642,842, filed on Jan. 10, 2005, the disclosures of which are herein incorporated by reference in their entirety.

BACKGROUND OF INVENTION

[0002] 1. Field of the Invention

[0003] The present invention relates to audio headsets, and more particularly to an apparatus and method to facilitate enhanced usability thereof.

[0004] 2. Description of Related Art

[0005] Our contemporary society enjoys numerous electronic devices that help to make our lives more productive, more comfortable and more efficient. One such device is the audio headset, which is a listening device that allows a user to hear audio material. Early examples of an audio headset included the monaural earpiece that was often seen accompanying the early transistor radio. Modern examples of an audio headset include stereo headphones used by casual listeners and audiophiles alike to enable a personal listening experience even in a public setting.

[0006] With the proliferation of portable audio and video players, the headset market has also exploded. People on the go or at home use an audio headset to allow them to listen to their electronic devices without external interruptions and without disturbing others. For example, on any given airline flight, one will see numerous passengers with his or her own audio or audio/video source such as, for example, a DVD player, MP3 player or the like. Headsets used with these and other devices allow the user to not only play the source material without disturbing others nearby, but also allow the user to at least partially screen out unwanted noises or audible distractions.

[0007] In fact, many contemporary headsets offer active or passive noise rejection or noise cancellation features to provide an enhanced listening experience. For example, passive systems may utilize a relatively tight seal around the perimeter of the ear or around the circumference of the ear canal to insulate the listener from unwanted noises. Active systems, on the other hand, may utilize electronic cancellation of undesirable background audio or noise.

[0008] Unfortunately, the advantages offered by many headsets also result in a disadvantage as well. That is, when listening to audio program content via a headset the user often experiences difficulty participating in a conversation with others around him or her or otherwise hearing certain background audible content that may be of interest. This is especially true with passive noise insulating or active noise cancellation devices, which make it particularly difficult for the user to hear a conversation, listen to someone speaking to them or otherwise hear outside audio information that may be of interest.

SUMMARY OF THE INVENTION

[0009] In accordance with one embodiment, an apparatus to provide control over the relative level of audio and background sound includes: an input sound transducer configured to convert ambient acoustical pressure into an electrical background signal representing the background sound; noise cancellation circuitry coupled to the input sound transducer and configured to create a cancellation signal that is the inverse of the electrical background signal; a summing component coupled to the noise cancellation component and configured to combine the cancellation signal with an audio signal representing audio program content and for providing the combined electrical signal to an output sound transducer; and a controller coupled to the noise cancellation component and configured to control the level of a cancellation signal relative to the audio signal, thereby controlling the mix of cancellation signal and audio signal reaching the output sound transducer.

[0010] In accordance with another embodiment, an apparatus to provide control over the relative level of audio and background sound includes: an input sound transducer configured to convert ambient acoustical pressure into an background audio signal representing the background sound; an audio input line configured to accept an audio content signal representing audio program content; and audio bypass circuitry coupled to the input sound transducer and the audio input line and configured to adjust the level of electrical background signal relative to the audio signal. The audio bypass circuitry can be implemented, for example, using at least one of a switch, an amplifier, and a processor. A summer can also be included to combine the audio content signal with the background audio signal.

[0011] Additionally, cancellation circuitry can be provided to adjust the electrical background signal such that it provides a level of cancellation of the background sound when played through an output audio transducer. The cancellation circuitry can include at least one of an inverter, a compensation filter, and a digital signal processor.

[0012] In accordance with yet another embodiment, an audio headset can be provided that includes: an input sound transducer configured to convert ambient acoustical pressure into an background audio signal representing the background sound; an audio input line configured to accept an audio content signal representing audio program content; audio bypass circuitry coupled to the input sound transducer and the audio input line and configured to adjust the level of electrical background signal relative to the audio signal; a summer configured to combine the audio content signal with the background audio signal to create a combined output signal; and an output transducer configured to convert the combined output signal into an audible signal.

[0013] The audio headset can include cancellation circuitry to adjust the electrical background signal such that it provides a level of cancellation of the background sound when played through the output transducer.

[0014] In accordance with yet another embodiment, a method for controlling the level of audible program content relative to a background sound is provided. The method converts a background sound into a background audio signal representing the background sound; receives an audio content signal representing audio program content; adjusts the level of electrical background signal relative to the audio content signal; combines the audio content signal and the electrical background signal into a combined output signal; and provides the combined signal to a speaker or other audio driver.

BRIEF DESCRIPTION OF THE DRAWINGS

[0015] The present invention, in accordance with one or more various embodiments, is described in detail with reference to the following figures. The drawings are provided for purposes of illustration only and merely depict typical or example embodiments of the invention. These drawings are provided to facilitate the reader's understanding of the invention and shall not be considered limiting of the breadth, scope, or applicability of the invention. It should be noted that for clarity and ease of illustration these drawings are not necessarily made to scale.

[0016] **FIG. 1** is a diagram illustrating an example environment in which the invention can be implemented.

[0017] **FIG. 2** is a diagram illustrating an example implementation for noise cancellation circuitry that can be implemented in accordance with one embodiment of the invention.

[0018] **FIG. 3** is a diagram illustrating an example implementation of audio bypass circuitry in accordance with one embodiment of the invention.

[0019] **FIG. 4** is a diagram illustrating an example operation of an audio bypass feature in accordance with one embodiment of the invention.

[0020] **FIG. 5** is a diagram illustrating an example implementation of audio bypass circuitry in accordance with one embodiment of the invention.

[0021] **FIG. 6** is a diagram illustrating an example implementation of audio bypass circuitry in accordance with one embodiment of the invention.

[0022] **FIG. 7** is a diagram illustrating an example implementation of audio bypass in accordance with one embodiment of the invention.

[0023] **FIG. 8** is a diagram illustrating an example implementation of audio bypass circuitry in accordance with one embodiment of the invention.

[0024] **FIG. 9** is a diagram illustrating an example implementation of audio bypass in accordance with one embodiment of the invention.

[0025] **FIG. 10** is a diagram illustrating an example implementation of audio bypass in accordance with one embodiment of the invention.

[0026] **FIG. 11** is a diagram illustrating an example controller in accordance with one embodiment of the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0027] The present invention is directed toward a system and method for providing an audio bypass feature for various headset applications that allow outside sounds or other background audio information to pass through the headset to the user's ears. In one embodiment, this audio bypass feature is enabled selectively by the user, but in other embodiments, user intervention is not utilized. The present invention is described herein in terms of an example application in a set of noise-canceling audio headphones. As will become apparent to one of ordinary skill in the art after reading this description, the present invention can be implemented in other headset applications where it may be useful or desirable for a user to allow outside audio information such as, for example, a conversation to pass through the headset so this information can be heard by the user.

[0028] Thus, before describing the invention in detail, this example application of a noise-canceling audio headset is first described. **FIG. 1** is a diagram illustrating one embodiment of a noise-canceling audio headset, which forms an example application for the present invention. Referring now to **FIG. 1**, this example noise-canceling audio headset **100** includes two ear pieces **104**, a headband **108**, an audio jack **116**, audio cables **120**, and optionally a switch or other control device **124**. Typically contained within ear pieces **104** are speakers or other audio drivers, (not illustrated in **FIG. 1**) which can be used to translate electrical representations of an audio signal into an audible audio content that is capable of being heard by the human ear. Ear pieces **104** may also include insulating structure **130** (for example, padded rings around the perimeter) to provide more effective sound insulation, thus helping to better shield the user from background audio or other unwanted background noise originating outside the headset that may detract from the desired audio content being played through the audio headset **100**.

[0029] Control device **124** may include controls such as, for example, a volume control, an on/off switch, a mute button, or other controls that allow the user to customize his or her listening experience. Control device **124** also provides a convenient location in which a battery, batteries, or other power source may be included to provide power to noise-cancellation headset **100**. Alternatively one or more of these items can be included elsewhere in audio headset **100** including, for example, on or within ear pieces **104**, headband **108** and so on.

[0030] In audio headsets **100** utilizing active noise-cancellation techniques, an input transducer **136**, such as for example a microphone or other device, can be included in either or both earpieces **104** for transducing acoustic pressure outside of ear pieces **104** (e.g., audible background noise) to a corresponding audio signal, which is used by the noise-cancellation system to filter out or otherwise cancel the background noise.

[0031] **FIG. 2** is a simplified block diagram illustrating an example implementation for noise cancellation circuitry that can be included with noise-canceling headset **100**. Referring now to **FIG. 2**, microphone **136** transduces the acoustical pressure generated by the unwanted background noise into an electrical signal that is enhanced by an amplifier such as pre-amp **202**. The electrical signal is inverted by an inverter **206** to create an inverse of the original signal representing the background noise, thus creating a cancellation signal **208**. Cancellation signal **208** can then be amplified via an amplifier **214**, which is preferably a variable gain amplifier, and provided to drive a speaker **218** to cancel the external noise that may penetrate the ear piece **104**. In the embodiment illustrated in **FIG. 2**, cancellation signal **208** is summed with the audio signal **222** via summer **210** to create a composite signal **226** to amplifier **214** to drive speaker **218** with an audio signal that includes the cancellation signal **208**, thus canceling, at least partially, the unwanted noise.

[0032] Cancellation signal **208** can be generated using an inverter **206**, a compensation filter, a DSP, or other circuitry, devices or techniques to create a signal that when provided

to speaker **218** cancels all or part of the background noise transduced by microphone **136**. The example application illustrated in **FIG. 2** shows the use of an inverter **206**. However, it will become apparent to one of ordinary skill in the art after reading this discussion how alternative techniques such as those provided in the foregoing examples can be implemented.

[0033] Having thus described an example environment in which the present invention can be applied, the present invention will now be described in greater detail in terms of this example environment. After reading this description, it will become apparent to one of ordinary skill in the art how to implement the invention in its various forms and embodiments in this or alternative environments in which it may be desirable to utilize the features and aspects of the present invention. **FIG. 3** is a high-level block diagram illustrating an example implementation of the audio bypass feature according to one embodiment of the invention. Referring now to **FIG. 3**, the present invention includes audio bypass module **306**, which is implemented to allow background sounds originating outside the headset to more readily be heard by the user of an audio headset. More particularly, in accordance with one embodiment of the invention, audio bypass module **306** can be implemented to allow a user to selectively engage or disengage the audio bypass feature to thereby allow selected background sounds to be heard by the user while the user is wearing an audio headset. In alternative embodiments, audio bypass module **306** can be implemented to automatically engage or disengage the audio bypass feature. Audio bypass module **306** in this and the other embodiments described herein can be implemented utilizing hardware, firmware, software, or a combination thereof, as may be desirable based on the given application.

[0034] An audio input is provided to accept the traditional audio signal **314** for the audio program content a user may desire to hear via a speaker **332**, such as, for example, the audio stream from a CD player, MP3 player, radio, DVD player or other audio source. The audio input may be implemented using a jack, banana plug or any other wired connector, and could alternatively be implemented utilizing a wireless interface.

[0035] A control signal **320**, typically generated by user interaction, can be utilized by audio bypass module **306** to allow an external audio signal transduced by a microphone **312** to pass through to speaker **332** such that that external audio signal **326** can be heard by the user without removing the audio headset. As discussed in the various embodiments presented herein, one or more control signals **320** can be generated by user interaction (e.g., via a control panel or other user interface) or generated automatically depending on the implementation.

[0036] **FIG. 4** is an operational flow diagram illustrating an example process for implementing audio bypass in accordance with one embodiment of the invention. Referring now to **FIGS. 3 and 4**, in a step **406**, in conventional operation, audio content is passed to speaker **332** by audio bypass module **306**. That is, for example, an audio signal **314** representing audio content from an electronic device can be heard via the ear pieces **104** in a user's noise-cancellation headset **100**.

[0037] In a step **410**, the audio bypass mode is initiated. In one embodiment, this can be accomplished by user interaction such as, for example, the user depressing a button or operating a switch to activate audio bypass module **306**. In other embodiments, this may be accomplished, for example, by automated means such as, for example, voice or speech recognition, background audio level threshold detection or other techniques.

[0038] In a step **414**, with audio bypass mode initiated, the audio signal **314** can be either attenuated or muted such that the audio program content is either softened before it reaches speaker **332** or is prevented from actually reaching speaker **332** so that it is diminished or not heard by the user during audio bypass mode.

[0039] In a step **420**, the external audio transduced by a microphone **312** (which can be implemented as, for example, microphone **136** in the example environment) is passed to speaker **332**, which converts the electrical signal into an audible signal that can be heard by the user via the headset speaker or speakers. In an environment of a noise cancellation headset **100**, it may be desirable to diminish or inhibit the noise cancellation functionality so that the background audio can be better heard by the user.

[0040] In a step **424**, when the audio bypass mode is terminated, audio signal **314** is returned to normal levels and the original audio content is once again passed to speaker **332** such that the user can resume listening to the audio content. Similar to initiating audio bypass mode, termination of the audio bypass mode can be accomplished by a manual means (e.g., via a button or switch etc.) or through automatic means (e.g., voice or speech recognition or audio level threshold detection and so on).

[0041] As will become apparent to one of ordinary skill in the art after reading this description, audio bypass module **306** can be implemented utilizing a number of different techniques. A few example embodiments for implementing audio bypass module **306** are now described. **FIG. 5** is a diagram illustrating one example implementation of audio bypass module **306** in accordance with one embodiment of the invention. In the example illustrated in **FIG. 5**, audio signal **314** is a stereo signal and is thus provided as a left audio signal **314**A and a right audio signal **314**B. A microphone **312** transduces the acoustical pressure generated by the background noise (e.g., the background audio) thereby generating a background audio signal **526**. In this example embodiment, audio bypass module **306** is illustrated as being implemented utilizing switches **522**, **523**, and **524**. Switches **522**, **523**, and **524** can be implemented utilizing various technologies including, for example, FET switches or other switching techniques. Because various audio signals are being switched through switches **522**, **523**, and **524**, it is desirable to utilize technologies that exhibit the appropriate frequency response characteristics for handling such audio signals. It is also often desirable that low cost means be used when implementing switches **522**, **523**, and **524**. Additionally, it is important to consider isolation of the various signals in the device.

[0042] One or more control signals **520**, as illustrated, are utilized to provide control information to configure switches **522**, **523**, and **524** to initiate and/or terminate the audio bypass mode. For example, to initiate the audio bypass mode, switches **522** and **524** would be set to the B position and switch **523** set to the A position. In this configuration, background audio signal **526** is passed through audio bypass

module **306** and on to speakers **332**A and **332**B. Alternatively, when not in audio bypass mode, switches **522** and **524** are set to position A and switch **523** is set to position B, thus allowing audio signal **314**A and audio signal **314**B to pass to its respective speaker **332**A and **332**B.

[**0043**] Summers **532**A and **532**B can be used to combine the respective audio signals **314**A and **314**B with the background audio signal **526**. Variable gain amplifiers **536**A (or fixed gain amplifiers) can be utilized to provide amplification of the resultant signal prior to driving speakers **332**A and **332**B.

[**0044**] In the embodiment illustrated in **FIG. 5**, a single microphone **312** is illustrated as providing a background audio signal from a single source (i.e., microphone **312**) to both speakers **332**A, **332**B in a stereo headset. As will become apparent to one of ordinary skill in the art after reading this description, a second microphone (not illustrated) and a second switch (not illustrated) can be provided in addition to microphone **312** and switch **523** to provide a separate background audio signal for each of the left and right channels.

[**0045**] **FIG. 6** is a simplified schematic diagram illustrating another example implementation of audio bypass module **306** in accordance with one embodiment of the invention. Referring now to **FIG. 6**, in the embodiment illustrated, switches **522**, **523**, and **524** have been replaced by amplifiers **622** and **623**. More specifically, in the embodiment illustrated in **FIG. 6**, these amplifiers are illustrated as variable gain amplifiers. Note, however, that the embodiment illustrated in **FIG. 6** includes two microphones **312**A and **312**B for left and right channel operation.

[**0046**] The operation of this embodiment is similar to the operation of **FIG. 5**, wherein one or more control signals **520** are utilized to provide input to audio bypass module **306** to selectively allow a background audio signal to pass through audio bypass module **306** to be heard by the user via speakers **332**. Thus, when the audio bypass mode is initiated, control signal **520** can be utilized to cause audio signals **314**A and **314**B to be attenuated by variable gain amplifiers **622**A and **622**B, and background or outside audio signals **312**A and **312**B to be amplified by variable gain amplifiers **623**A and **623**B. Thus, in this embodiment, the signal level, and thus ultimate volume, of background audio signal **312** is increased relative to that of audio signal **314**, thus enabling the user to better hear the background audio signal **312** over the audio program content **314** from the audio source of the electronic device. Also illustrated in the example provided in **FIG. 6** are summers **532**A, **532**B and output amplifiers **536**A, **536**B. In the configuration illustrated in **FIG. 6**, two microphones **312**A and **312**B are provided such that a separate background audio signal can be provided for each channel of the headset. However, as would be apparent to one of ordinary skill in the art after reading this description, the embodiment can be implemented using a single microphone **312** and a single amplifier **623** to transduce and provide a single background audio signal to both channels of the listening device.

[**0047**] The embodiments described above with reference to **FIGS. 5 and 6** illustrate example applications in which audio bypass module **306** is used in conjunction with headset devices having no noise-cancellation circuitry or utilizing passive noise attenuation techniques to filter out unwanted background noise from the listening environment. Now described are a plurality of embodiments that can be utilized in applications such as, for example, the example environment of a noise-cancellation headset **100** utilizing active noise-cancellation techniques. **FIG. 7** is a simplified block diagram illustrating at a high level the implementation of an audio bypass mode with a noise-cancellation headset. Referring now to **FIG. 7**, the example implementation includes controller **704** that can be utilized to control a noise cancellation component **706** and an audio component **708** of the noise-canceling headset. In this example embodiment, when the audio bypass mode is initiated, controller **704** attenuates or mutes audio signal **314** and allows the background noise signal to pass through noise cancellation portion **706**. This can be accomplished by bypassing or inhibiting the noise cancellation circuitry, allowing the external signal transduced by microphone **312** to be passed to speakers **332**. Alternatively, this can also be accomplished by allowing noise cancellation portion **706** to continue unaltered or unadjusted. This is true especially, for example, in configurations where noise cancellation portion **706** is utilized to filter out only low frequency noises and does not affect, at least not to a great extent, noise in the frequency ranges of ordinary human speech. Also illustrated in the embodiment shown in **FIG. 7** are a summer **532**, amplifier **536** and speaker **332** to provide the selected audio content to the user. Although one audio channel is illustrated, it will become apparent to one of ordinary skill in the art how to implement the invention utilizing multiple channels.

[**0048**] **FIG. 8** is a simplified schematic diagram illustrating an example implementation of the audio bypass feature in a noise-cancellation headset in accordance with one embodiment of the invention. Referring now to **FIG. 8**, the example implementation illustrated herein is that of a stereo or other dual-channel headset in which first and second audio signals **314**A, **314**B (e.g., left and right audio signals) are received and played via speakers **332**. After reading this description, it will be apparent to one of ordinary skill in the art how to implement this embodiment in a monaural application or in applications having an alternative number of channels.

[**0049**] In normal operation, audio signals **314**A and **314**B are received from the audio source and can be conditioned utilizing buffer amps **710**A and **710**B, respectively. Audio signals **314**A and **314**B are typically received from the audio source via an audio jack such as, for example, an eighth inch stereo plug or other audio connecter. However, other input techniques, both hardwire and wireless, can be utilized. Conditioned audio signals **716**A and **716**B are switched by audio bypass module **306** to drivers **732**A and **732**B via mixers **730**A and **730**B such that they can be heard by the user on speakers **332**A and **332**B.

[**0050**] With the noise cancellation feature turned on, acoustical pressure generated by background noise is transduced by microphones **312**A and **312**B to create a background noise signals **726**A and **726**B that can be conditioned by optional driver/pre-amp **704**A and **704**B and ultimately inverted via inverters **708**A and **708**B to create noise-cancellation signals **728**A and **728**C. Noise-cancellation signals **728**A and **728**C are summed with conditioned audio signal **716**A and **716**B via summer **730** and amplified by driver **732** to provide an audio signal to the user that includes both the original audio program content and a component

representative of the inverse of the background noise. Thus, when the background noise is combined with its inverse and this signal is used to drive speaker **332**, the background noise is canceled such that it is not heard by the user (or it is at least diminished).

[**0051**] In the embodiment illustrated in **FIG. 8**, audio bypass module **306** comprises a plurality of switches similar to those utilized in the embodiment illustrated in **FIG. 5**. Thus, when the audio bypass mode is initiated, switches **724**A and **724**B are placed in the B position, as are switches **725**A and **725**B. Thus, in the audio bypass mode, the input audio signals **314**A and **314**B are muted and the inverters **708**A and **708**B utilized to create the cancellation signal are bypassed, thus allowing the background noise signals **726**A and **726**B to pass through to speakers **332**A and **332**B. In one example embodiment of this configuration, the gain of amplifiers **704** or amplifier **732** (or both) can be increased in the audio bypass mode to allow the background audio signal to be amplified to a greater extent so that it can be heard by the user. Thus, in this configuration, in the audio bypass mode, the background audio signal (for example, a conversation within earshot of the headset user) can be picked up by microphones **312**A and **312**B, amplified by the circuitry, and readily heard by the user via speakers **332**A and **332**B. When the audio bypass mode is terminated, switches **724**A, **724**B, **725**A, and **725**B can be returned to the A position such that background noise cancellation once again commences and the audio signals **314**A, **314**B can be passed through the respective componentry to speakers **332**A, **332**B. As discussed above, an inverter **708** can be implemented utilizing a number of different techniques to provide an inverse or complementary audio signal utilized to cancel out the background audio noise. This can be accomplished using inverters, compensation filters, and other known techniques.

[**0052**] Additionally, in this and other embodiments instead of an inverter **708**, a band pass or other filter can be utilized to filter out unwanted background noise. For example, in the environment of an aircraft cabin, the unwanted background noise is often predominantly made up of low frequency components resulting from the roar of the aircraft engines. Thus, for applications utilized in such environments, it is possible to implement the noise-cancellation function a band pass filter to filter out low frequency sounds, thus allowing only higher frequency sounds to pass. Thus, in this embodiment, in addition to or in place of inverters **708**, band pass filters can be utilized to limit audio at certain frequencies. Additionally, circuitry components or instrumentality can be utilized in signal paths **744**A and **744**B to enhance or amplify signals falling within the frequency range of normal human speech. Thus, for example, an amplifier and a band pass filter can be included where the transfer response of the band pass filter is tuned to allow frequency components within the range of human speech to be transferred to the speakers **332**A, **332**B for playback to the listener. Thus, in this embodiment, amplifying the range of frequencies covered by human speech can enhance the ability of the user to hear a conversation that is occurring outside the ear pieces and, thus, allow the user to participate in or listen to that conversation without necessarily removing the headphones.

[**0053**] **FIG. 9** is a simplified schematic diagram illustrating yet another application of the invention in the environment of a noise-canceling headset in accordance with one embodiment of the invention. **FIG. 9** illustrates one audio channel; however, as described above, this embodiment can be implemented in applications utilizing a plurality of audio channels including, for example, a stereo headset. Referring now to **FIG. 9**, bypass module **306** is implemented using a plurality of amplifiers, which in the illustrated embodiment are variable gain amplifiers **922**, **924**, and **926**. Variable gain amplifiers **922**, **924**, and **926** can be implemented in place of switches **724** and **725** illustrated in **FIG. 8**. More particularly, in this embodiment, controller **704** can be utilized to control the gain of amplifiers **922**, **924**, and **926** to selectively pass cancellation signal **728**, background audio signal **726**, or the audio signal **314**. Preferably, when the audio bypass mode is initiated, the gain of variable gain amplifiers **922** and **926** is attenuated such that noise cancellation signal **728** and audio signal **314** are attenuated and the gain of variable gain amplifier **924** is increased such that the background audio signal **726** is amplified and readily heard by the user through speaker **332**. Conversely, when the bypass mode is terminated, the gain of variable gain amplifier **924** is turned down such that the background audio signal **726** is no longer amplified (or is attenuated to some extent). Additionally, the gain of variable gain amplifier **926** is increased such that audio signal **314** is amplified and can be readily heard by the user via speaker **332**. Also, the gain on variable gain amplifier **922** can be increased such that noise cancellation signal **728** is amplified and can be utilized in conjunction with audio signal **314** to minimize the amount of background noise heard by the user.

[**0054**] In yet another alternative embodiment, audio bypass module **306** can be implemented utilizing DSP technology. For example, in this embodiment, one or more analog/digital converters (ADC) can be utilized to digitize either or both of background noise signal **326** and audio signal **314**. Once digitized, these signals can be processed utilizing a digital signal processor to, for example, amplify the speech signal, diminish the background noise, and inhibit the audio signal **314** from passing through to speaker **332**. One example implementation utilizing DSP technology is illustrated in **FIG. 10** in accordance with one embodiment of the invention. Referring now to **FIG. 10**, A/D converter **944** is utilized to digitize background noise signal **326**. A digital signal processor **996** can be utilized to provide the noise-cancellation functions desired to diminish or eliminate undesirable background noise from the content heard by the user. D/A converter **911** is used to perform a digital to analog conversion such that the resultant noise cancellation signal can be converted to an audible signal via speaker **332**. Additionally, a variable gain amplifier (or, alternatively, a switch or other component or instrumentality) **993** can be included to attenuate or amplify audio signal **314**. Thus, when the audio bypass mode is initiated, the gain of variable gain amplifier **993** can be decreased to minimize the level of audio signal **314** while the operation of DSP **996** occurs such that the noise cancellation features are inhibited and the background audio signal **314** is allowed to pass through to speaker **332**. More particularly, in one embodiment, the DSP **996** can be utilized to actually enhance the background audio signal such that the conversation or other audio that the user desires to hear in the bypass mode is better heard via speaker **332**.

[**0055**] The above description provides details on a plurality of embodiments that utilize some form of audio bypass

module **306** that optionally attenuates an audio signal **314** such that a background signal **326** can be heard by the user via speaker **332**. In the illustrated embodiments, a microphone **312** (or **136** in the example application) is included to detect and transduce the acoustical pressure generated by the background noise that the user wishes to hear. In the example environment illustrated in **FIG. 1**, microphone **136** is provided in ear piece **104**. In fact, in the embodiment illustrated in **FIG. 1**, there are two microphones **136**, one in each ear piece **104**. This is often the configuration found with noise-cancellation headsets **100** with which the invention can be implemented. Alternatively, microphones **312** can be implemented internal to ear piece **104** as opposed to externally, as illustrated in **FIG. 1**. In this embodiment, it may be more difficult to detect background sounds that are external to the ear pieces, as the ear pieces **104** may provide insulative properties shielding an internal microphone from such conversations. However, in such an environment, various combinations of filtering and amplification as discussed above may be utilized to further enhance the audibility of background audio picked up by internal microphones. Although not illustrated, in yet another embodiment, one or more microphones can be provided in other locations such as, for example, on or in proximity to headband **108**, somewhere along cord **120**, or integrated with control device **124**. Although not required, it is anticipated that in noise cancellation applications, microphones **136** will be positioned somewhere in the proximity of ear pieces **104** such that, when they are performing the noise-cancellation function, they do a better job of approximating the sound that is to be canceled from the ultimate signal.

[**0056**] Although some of the above embodiments discuss utilizing as microphones **312** existing microphones **136** that are also utilized for noise cancellation, it is also possible to implement the invention utilizing one or more additional microphones **312** rather than utilizing the existing microphones **136** provided for noise-cancellation purposes. For example, it may be desirable to provide one or more microphones **312** on or near control device **124**, headband **108**, cord **120** or jack **116**. One advantage, depending on the alternative placement, is that it may allow the microphone(s) **312** to be directed toward the source of the background audio for better pickup.

[**0057**] A user interface device can be included to allow the user to control one or more features of the audio bypass function. **FIG. 11** is a diagram illustrating an example implementation of such a control device in accordance with one embodiment of the invention. Referring now to **FIG. 11**, illustrated is a control device **372** that is utilized to provide manual control of one or more of the features of the invention. The control device **372** illustrated in **FIG. 11** is illustrated as being positioned along audio cables **380**; however, other locations are possible, including on either or both ear pieces **104** or on headband **108** or in conjunction with jack **116**. In one embodiment, control device **372** is hardwired in with the headset; however, wireless control devices **372** are also contemplated.

[**0058**] The features included with control device **372** in the illustrated example can include a mute button **390**, an audio bypass button **392**, a volume control **394**, and a speaker **396**, although each of these need not be present with a control device **372** and alternatives are possible, depending on the desired implementation. Mute button **390** can be utilized to mute the audio signal sent to the headset, thereby silencing in its entirety the audio content provided by the speakers. Alternatively, or in addition, a mute button can be provided that mutes only the original audio program content such that background audio can be heard without having to be heard above the audio signal (or vice versa). An audio bypass button **392** can be provided to allow the user to manually enter the audio bypass mode. Thus, depressing this button (or flicking the switch, etc.) allows the user to manually enter the audio bypass mode and utilize the features discussed above.

[**0059**] Volume control **394** can be included and utilized to adjust the audio volume presented by the headset. In one embodiment, this can adjust the volume for all features; however, one or more volume knobs **394** can be included to adjust individually if desired the audio content playback level, the background audio level, the noise cancellation level, and so forth.

[**0060**] As discussed above, a microphone **396** can also be included on the control device which can be used in addition to or as an alternative to other microphones that may be positioned elsewhere such as, for example, on ear pieces **104**. Although jack **116** in this embodiment is illustrated as a standard stereo plug, it is contemplated that other connection devices can be utilized to receive the audio playback signal such as, for example, USB connectors, wireless communication ports, and so on.

[**0061**] In addition to utilizing a manual control to enter or exit the audio bypass mode, an automatic mode can be included wherein the audio bypass mode is controlled automatically based on the occurrence of one or more events. For example, in one embodiment, voice or speech recognition techniques can be utilized to allow the headset to enter the audio bypass mode in response to a command that is spoken by the user. Thus, for example, the user can program the device to enter the audio bypass mode when a particular word or phrase is spoken. For example, the headset in this embodiment could be programmed to respond to a phrase such as "Could you repeat that?" Thereby enabling the audio bypass mode when the user asks another person to repeat what they have just said. As another example, the device may be programmed to enter automatically the audio bypass mode when a background audio level is sensed that is louder or at a higher level than the ambient background noise or, for example, when a particular threshold is reached. Thus, for example, in this application, when the device senses that the user has spoken (as words spoken by the user may be sensed by the device at a level louder than the background noise), the device could automatically enter the audio bypass mode.

[**0062**] Additionally, timeout features can be provided to allow the device to automatically terminate the audio bypass mode upon the passage of a specified amount of time. This could be preprogrammed or user selected and controlled, depending on the implementation.

[**0063**] Various embodiments of the invention and implementation examples have been described above. However, it is understood that these various embodiments and examples are exemplary only and should not serve to limit the scope of the invention. It is also readily understood by those of ordinary skill in the art how to design and implement the disclosed embodiments using alternative architectures, processes, functionality, structures, and implementations. In

sum, after reading this description, various modifications of and alternatives to the preferred embodiments described above can be implemented by those of ordinary skill in the art, without undue experimentation. These various modifications and alternatives are contemplated to be within the spirit and scope of the invention.

[0064] As an example of such modifications and alternatives and without limiting the generality of the foregoing, the example embodiments described herein illustrate a configuration utilizing a particular configuration of amplifiers and summers for combining and driving the various signals. As will be apparent to one of ordinary skill in the art after reading this disclosure, amplifiers could be provided elsewhere in the chain and summers can be provided as summing amplifiers. Also as would be apparent to one of ordinary skill in the art after reading this description, the circuitry can also include appropriate drivers, pre-amps, buffer amps, or other components or instrumentalities, as desired or appropriate to further condition the various audio signals. Furthermore, the schematic diagrams are provided to illustrate example implementations of various embodiments of the functionality of the audio bypass feature, and one of ordinary skill in the art after reading this description will understand how to implement these features and functions utilizing alternative configurations of hardware, software, firmware or combinations thereof. Likewise, particular implementations of switching functionality are disclosed herein in terms of FET switches and variable gain amplifiers. It would be apparent to one of ordinary skill in the art after reading this description how to implement the functionality of selecting and deselecting, or amplifying and attenuating selected audio signals to achieve the desired effects.

[0065] Terms and phrases used in this document, and variations thereof, unless otherwise expressly stated, should be construed as open ended as opposed to limiting. As examples of the foregoing: the term "including" should be read as mean "including, without limitation" or the like; the term "example" is used to provide exemplary instances of the item in discussion, not an exhaustive or limiting list thereof; and adjectives like "conventional,""traditional, ""normal,""standard," and terms of similar meaning should not be construed as limiting the item described to a given time period or to an item available as of a given time, but instead should be read to encompass conventional, traditional, normal, or standard technologies that may be available now or at any time in the future. Likewise, a group of items linked with the conjunction "and" should not be read as requiring that each and every one of those items be present in the grouping, but rather should be read as "and/or" unless expressly stated otherwise.

What is claimed is:

1. An apparatus to provide control over the relative level of audio and background sound, comprising:

an input sound transducer configured to convert ambient acoustical pressure into an electrical background signal representing the background sound;

noise cancellation circuitry coupled to the input sound transducer and configured to create a cancellation signal that is the inverse of the electrical background signal;

a summer coupled to the noise cancellation circuitry and configured to combine the cancellation signal with an audio signal representing audio program content and for providing the combined electrical signal to an output sound transducer; and

a controller coupled to the noise cancellation circuitry and configured to control the level of a cancellation signal relative to the audio signal, thereby controlling the mix of cancellation signal and audio signal reaching the output sound transducer.

2. The apparatus of claim 1, wherein the cancellation circuitry comprises at least one of an inverter, a compensation filter, and a digital signal processor.

3. The apparatus of claim 1, wherein the controller is configured to attenuate or mute the audio signal.

4. The apparatus of claim 1, wherein the controller is configured to attenuate or mute the cancellation signal.

5. The apparatus of claim 1, wherein the controller is configured to inhibit or diminish functionality of the noise cancellation circuitry.

6. The apparatus of claim 1, wherein the controller is configured to cause the electrical background signal to bypass the noise cancellation circuitry.

7. An apparatus to provide control over the relative level of audio and background sound, comprising:

an input sound transducer configured to convert ambient acoustical pressure into an background audio signal representing the background sound;

an audio input line configured to accept an audio content signal representing audio program content; and

audio bypass circuitry coupled to the input sound transducer and the audio input line and configured to adjust the level of electrical background signal relative to the audio signal.

8. The apparatus of claim 7, wherein the audio bypass circuitry comprises at least one of a switch, an amplifier, and a processor.

9. The apparatus of claim 7, further comprising a summer configured to combine the audio content signal with the background audio signal.

10. The apparatus of claim 7, further comprising cancellation circuitry coupled to the input sound transducer and configured to adjust the electrical background signal such that it provides a level of cancellation of the background sound when played through an output audio transducer.

11. The apparatus of claim 10, wherein the cancellation circuitry comprises at least one of an inverter, a compensation filter, and a digital signal processor.

12. The apparatus of claim 10, wherein the audio bypass circuitry is configured to cause the electrical background signal to bypass the noise cancellation circuitry.

13. The apparatus of claim 7, wherein the audio bypass circuitry is configured to attenuate or mute the audio signal.

14. The apparatus of claim 7, wherein the audio bypass circuitry is configured to attenuate or mute the cancellation signal.

15. The apparatus of claim 7, wherein the audio bypass circuitry is configured to inhibit or diminish functionality of the noise cancellation circuitry.

16. The apparatus of claim 7, wherein engagement of the audio bypass circuitry is controlled by at least one of voice recognition, speech recognition, and background audio level threshold detection.

17. The apparatus of claim 7, further comprising a user interface device configured to allow a user to control the audio bypass circuitry.

18. The apparatus of claim 7, further comprising means for automatically engaging the audio bypass circuitry.

19. The apparatus of claim 18, wherein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection.

20. An audio headset, comprising:

an input sound transducer configured to convert ambient acoustical pressure into an background audio signal representing the background sound;

an audio input line configured to accept an audio content signal representing audio program content;

audio bypass circuitry coupled to the input sound transducer and the audio input line and configured to adjust the level of electrical background signal relative to the audio signal;

a summer configured to combine the audio content signal with the background audio signal to create a combined output signal; and

an output transducer configured to convert the combined output signal into an audible signal.

21. The audio headset of claim 20, wherein the audio bypass circuitry comprises at least one of a switch, an amplifier, and a processor.

22. The audio headset of claim 20, further comprising cancellation circuitry coupled to the input sound transducer and configured to adjust the electrical background signal such that it provides a level of cancellation of the background sound when played through the output transducer.

23. The audio headset of claim 22, wherein the cancellation circuitry comprises at least one of an inverter, a compensation filter, and a digital signal processor.

24. The audio headset of claim 20, wherein the audio bypass circuitry is configured to cause the electrical background signal to bypass the noise cancellation circuitry.

25. The audio headset of claim 20, wherein the audio bypass circuitry is configured to attenuate or mute the audio signal.

26. The audio headset of claim 20, wherein the audio bypass circuitry is configured to attenuate or mute the cancellation signal.

27. The audio headset of claim 20, wherein the audio bypass circuitry is configured to inhibit or diminish functionality of the noise cancellation circuitry.

28. The audio headset of claim 20, wherein engagement of the audio bypass circuitry is controlled by at least one of voice recognition, speech recognition, and background audio level threshold detection.

29. The audio headset of claim 20, further comprising a user interface device configured to allow a user to control the audio bypass circuitry.

30. The audio headset of claim 20, further comprising means for automatically engaging the audio bypass circuitry.

31. The audio headset of claim 30, wherein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection.

32. A method for controlling the level of audible program content relative to a background sound, the method comprising the steps of:

converting a background sound into a background audio signal representing the background sound;

receiving an audio content signal representing audio program content;

adjusting the level of electrical background signal relative to the audio content signal;

combining the audio content signal and the electrical background signal into a combined output signal; and

providing the combined signal to a speaker or other audio driver.

33. The method of claim 32, wherein the step of adjusting causes the electrical background signal to bypass the noise cancellation circuitry.

34. The method of claim 32, wherein the step of adjusting attenuates or mutes the audio signal.

35. The method of claim 32, wherein the step of adjusting attenuates or mutes the cancellation signal.

36. The method of claim 32, wherein the step of adjusting inhibits or diminishes functionality of the noise cancellation circuitry.

37. The method of claim 32, wherein the step of adjusting is controlled by at least one of voice recognition, speech recognition, and background audio level threshold detection.

38. The method of claim 32, wherein the step of adjusting is controlled by a user.

39. An apparatus for controlling a relative level of audio and background sound in an audio playback device, comprising:

means for converting ambient acoustical pressure into an electrical background signal representing background sound;

means for controlling the signal level of an electrical background signal relative to the level of an electrical audio signal representing audio program content;

means for receiving an electrical audio signal representing audio program content;

means for combining an electrical background signal with an electrical audio signal and The method of claim 32, wherein the step of adjusting is controlled by providing the combined electrical background and audio signal to an output sound transducer.

* * * * *

Exhibit OTH-A

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Reexamination of:

Inventors: Steven Wayne Goldstein, et al.

U.S. Patent No. 11,244,666

Issued: February 8, 2022

For: Method and Device for Acute Sound Detection and Reproduction

Mail Stop *Ex Parte* Reexamination
Attn: Central Reexamination Unit
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

REQUEST FOR REEXAMINATION UNDER 35 U.S.C. §§ 302-307 AND 37 C.F.R. § 1.510

**DECLARATION OF NATHANIEL POLISH, PH.D.
IN SUPPORT OF REQUEST FOR *EX PARTE* REEXAMINATION
OF U.S. PATENT NO. 11,244,666**

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A. Grounds Challenging Claims 1-11 of the '666 Patent 48

B. Ground 1: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast 50

1. A person having ordinary skill in the art would have been motivated to combine Nemirovski and Rast 50

2. Claim 1 58

a. 1[pre] "1. An earphone comprising:" 58

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;" 1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;" 60

c. 1[c] "a speaker configured to play an audio signal;" 71

d. 1[d] "a memory that stores instructions; and" 1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:" 71

e. 1[f] "receiving a first microphone signal from the first microphone;" 1[g] "receiving a second microphone signal from the second microphone;" 82

f. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;" .. 83

g. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;" 90

h. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and" 94

i. 1[k] "sending the mixed audio signal to the speaker." .. 96

3. Claim 2 97

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and" 97

b. 2[b] "determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals." 101

4. Claim 3 102

a. "3. The earphone according to claim 2, where the operations further comprise: determining whether the acute sound is a warning sound or siren by analyzing the spectrum of at least one of the first microphone signal or the second microphone signal or a combination of both signals." 102

5. Claim 4 105

a. "4. The earphone according to claim 3, where the operations further comprise: decreasing a volume of the audio playback when a warning or siren is detected." 105

6. Claim 5 106

a. "5. The earphone according to claim 4, where the operations further comprise: sending a notification signal to the speaker." 106

7. Claim 6 107

a. "6. The earphone according to claim 4, wherein the warning is at least one of a bell or the sound of an emergency vehicle or the sound of a security system or a combination." 107

8. Claim 7 108

a. "7. The earphone according to claim 4, wherein the siren is at least one of a police siren or an ambulance siren or a car honking or a combination." 108

9. Claim 8 109

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected." 109

10. Claim 9 110

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal." 110

11. Claim 10 111

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal." 111

12. Claim 11 113

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section." 113

C. Ground 2: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Dijkstra 114

1. Independent Claim 1 114

2. Dependent Claims 2-11 121

D. Ground 3: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Van Doorn 122

1. Independent Claim 1 123

2. Dependent Claims 2-11 129

E. Ground 4: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Dijkstra and Beasley 130

1. A person having ordinary skill in the art Would Have Been Motivated to Combine Victorian and Beasley 130

2. Claim 1 134

a. 1[pre] "1. An earphone comprising:" 134

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;" 1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;" 135

c. 1[c] "a speaker configured to play an audio signal;" ... 139

d. 1[d] "a memory that stores instructions; and" 1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to

the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:" 140

e. 1[f] "receiving a first microphone signal from the first microphone;" 144

f. 1[g] "receiving a second microphone signal from the second microphone;" 144

g. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;" 145

h. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;" 146

i. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and" 149

j. 1[k] "sending the mixed audio signal to the speaker." 150

3. Claim 2 151

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and" 151

b. 2[b] "determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals." 153

4. Claim 8 156

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected." 156

5. Claim 9 157

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal." 157

6. Claim 11 158

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section." 158

F. Ground 5: Claim 10 is Obvious in View of Victorian in Combination with Dijkstra, Beasley, and Andrea 160

1. Claim 10 160

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal." 160

G. Ground 6: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Van Doorn and Beasley 168

1. Independent Claim 1 169

2. Dependent Claims 2, 8-9, and 11 171

H. Ground 7: Claim 10 is Obvious in View of Victorian in Combination with Van Doorn, Beasley, and Andrea 172

1. Claim 10 172

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal." 172

I. Ground 8: Claims 1 and 11 are Obvious in view of Dijkstra in combination with Boersma 173

1. A person having ordinary skill in the art Would Have Been Motivated to Combine Dijkstra and Boersma 173

2. Claim 1 180

a. 1[pre] "1. An earphone comprising:" 180

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;" 1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;" 182

c. 1[c] "a speaker configured to play an audio signal;"... 184

d. 1[d] "a memory that stores instructions; and" 185

e. 1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:" 185

f. 1[f] "receiving a first microphone signal from the first microphone;" 186

g. 1[g] "receiving a second microphone signal from the second microphone;" 187

h. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;" 187

i. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;" 190

j. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and" 192

k. 1[k] "sending the mixed audio signal to the speaker." 193

3. Claim 11 194

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section." 194

J. Ground 9: Claims 2, 8-9 are Obvious in view of Dijkstra in combination with Boersma and Beasley 195

1. Claim 2 195

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and 195

b. 2[b] determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals." 199

2. Claim 8 202

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected." 202

3. Claim 9 203

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal." 203

K. Ground 10: Claims 1 and 11 are Obvious in view of Dijkstra, Boersma, and Van Doorn 204

1. Independent Claim 1 205

2. Claim 11 207

L. Ground 11: Claims 2 and 8-9 are Obvious in view of Dijkstra, Boersma, Van Doorn, and Beasley 208

XI. CONCLUSION 208

I, Nathaniel Polish, declare as follows:

I. INTRODUCTION

1. I have been retained by Samsung Electronics Co., Ltd. and Samsung Electronics America, Inc. ("Samsung" or "Requesters") as an independent expert consultant in connection with Samsung's Request for *Ex Parte* Reexamination of U.S. Patent No. 11,244,666 (the "'666 patent").

2. I am being compensated at a rate of $675/hour for my services, which is my regular and customary rate.

3. My compensation is in no way contingent on the nature of my findings, the disposition of Samsung's *Ex Parte* Reexamination Request, or the results of any subsequent reexamination of the '666 patent.

4. I have been asked to consider whether certain references disclose or suggest the features recited in the claims of U.S. Patent No. 11,244,666 ("the '666 patent") (PAT-A).[1] My opinions are set forth below.

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[1] Where appropriate, I refer to exhibits I understand will be submitted with the request for *ex parte* reexamination of the '666 patent (the "Request").

II. BACKGROUND AND QUALIFICATIONS

5. I am an independent consultant. All of my opinions stated in this declaration are based on my own personal knowledge and professional judgment. In forming my opinions, I have relied on my education, experience, and knowledge regarding electrical engineering, computer science, and audio and acoustic sciences.

6. I am over 18 years of age and, if I am called upon to do so, I would be competent to testify as to the matters set forth herein. A copy of my current curriculum vitae, which details my education and professional and academic experience, as well as a list of all publications I have authored in the past ten years, is included as Ex. OTH-B in this proceeding. The following provides an overview of some of my experience that is relevant to the matters set forth in this declaration.

7. I expect to testify regarding my background, qualifications, and experience relevant to the issues in this litigation. I have a Ph.D. in Computer Science from Columbia University. I hold the following four degrees from Columbia, spanning the years 1980 to 1993:

- Ph.D. in Computer Science, May 1993, Thesis: Mixed Distance Measures for the Optimization of Concatenative Vocabularies in Speech Synthesis;
- M.Phil. in Computer Science, December 1989;
- M.S. in Computer Science, December 1987; and

- B.A. in Physics, Columbia College, May 1984.

8. For over thirty years, I have run a computer technology development firm that I co-founded, called Daedalus Technology Group. My primary business activity is the development of computer-related products. This activity involves understanding the business objectives of customers, designing products to suit their needs, and supervising the building, testing, and deployment of these products. I develop hardware and software as well as supervise others who do so.

9. Also, from time to time I found other companies in order to pursue particular product opportunities. I develop and ultimately sell these companies. Most of my business activity, however, is as a consulting product developer. From time to time I have also served as an expert witness on computer and software related cases. I am a named inventor on eight United States patents and am a member of several professional societies, including the IEEE and ACM.

10. I have extensive experience in several areas relevant to this case. My doctoral work was in the area of computer speech. As part of that work I developed signal processing systems and algorithms to filter and manipulate sound, especially speech. That work gave me strong practical insight into such topics as voice activity detection, digital mixing of audio and speech, as well as the general problems of digital filtering of speech signals.

11. Much of my professional work over the past thirty years has been as a computer technology product developer. Several of those products involved technologies closely related to the topics in the patent at issue in the current matter, including digital audio systems and associated signal processing. One product in particular was for a company called HeadGear in the later 1990s. That product involved developing a set of headphones that mixed ambient music from an outdoor concert with wirelessly delivered audio content. Under my supervision, the firm developed prototype versions of this product.

12. In addition to this work for HeadGear, the firm also designed and developed, under my supervision, a number of products or systems for delivering digital audio and performing associated signal processing. Between 1982-1991, the firm designed and built, under my supervision, all parts of four generations of interactive voice editing systems for The Dun & Bradstreet Corporation. These editing systems produced output used in an extensive, high quality speech synthesis system that was part of a larger information delivery system. The systems were built on networks of computers in a variety of programming languages and systems. The systems involved interactive user interfaces, screen windows, signal processing, and voice file systems.

13. Between 1995-1997, the firm also designed and developed, under my supervision, a series of products to deliver digital video and audio materials over

wide-area internet using consumer grade computers. These products were developed for Instant Video Technologies, Inc. and designed to fit within their existing patent portfolio. There were also new aspects of the technology that resulted in a new patent for the company.

14. In 2000, I also supervised the development, design, and deployment of an industrial strength interactive voice response (IVR) system that allows users to access streaming media via their cell phones. This system supported up to 96 streams on a single chassis and was designed for a company called Savos. Under my supervision, the firm developed for this system the web-based content management system and interface for customizable user preferences.

15. In addition to the products and systems described above, I have also developed many other systems that utilize various signal processing techniques, including both speech recognition and speech synthesis. In the late 1990s, I started a company named Soliloquy to commercialize my technology in the area of speech access to databases. That work resulted in a patent. I have also developed a number of systems that use speech recognition to control system functions. One of the more amusing systems that I developed for a client was a talking golf club.

16. Through these and other experiences, I am very familiar with the technologies at issue in this matter.

III. MATERIALS REVIEWED

17. The opinions contained in this declaration are based on the documents I reviewed, my professional judgment, as well as my education, experience, and knowledge regarding electrical engineering, computer science, and audio and acoustic sciences.

18. In forming my opinions expressed in this declaration, I reviewed the following materials:

- PAT-A U.S. Patent No. 11,244,666 ("the '666 patent")
- PAT-B File Wrapper for the '666 patent
- PAT-D Excerpts from the File Wrapper for U.S. Patent No. 10,134,377
- PA-A U.S. Patent No. 6,647,368 ("Nemirovski")
- PA-B U.S. Patent Application Publication No. 2001/0046304A1 ("Rast")
- PA-C European Patent Application Publication No. EP1640972A1 ("Dijkstra")
- PA-D U.S. Patent Application Publication No. 2004/0047486A1 ("Van Doorn")
- PA-E European Patent Application Publication No. EP1519625A2 ("Victorian")
- PA-F U.S. Patent Application Publication No. 2006/0153394A1 ("Beasley")
- PA-G U.S. Patent No. 5,251,263 ("Andrea")
- PA-H U.S. Patent No. 7,903,826 ("Boersma")

- PA-I U.S. Patent No. 6,754,359 ("Svean")
- PA-J U.S. Patent No. 7,983,433 ("Nemirovski '433")
- PA-K U.S. Patent No. 7,502,484 ("Ngia")
- PA-L U.S. Patent No. 8,351,634 ("Khenkin")
- PA-M U.S. Patent No. 7,953,241 ("Jorgensen")
- PA-N U.S. Patent No. 4,533,795 ("Baumhauer")
- PA-O U.S. Patent No. 8,014,553 ("Radivojevic")
- PA-P U.S. Patent No. 6,728,385 ("Kvaløy")
- PA-Q Excerpts from ROBERT OSHANA, DSP SOFTWARE DEVELOPMENT TECHNIQUES FOR EMBEDDED AND REAL-TIME SYSTEMS (2006) ("Oshana")
- PA-R U.S. Patent No. 8,150,084 ("Jessen")
- PA-S U.S. Patent Application Publication No. 2003/0035551A1 ("Light")
- PA-T U.S. Patent Application Publication No. 2006/0083388A1 ("Rothschild")
- PA-U U.S. Patent No. 6,671,379 ("Nemirovski '379")
- PA-V U.S. Patent No. 9,135,797 ("Couper")
- PA-W Mulgrew, Bernard et al., Digital Signal Processing: Concepts and Applications (2nd ed. 2002) ("Mulgrew")
- OTH-E Decision Regarding Institution, *Samsung Electronics Co., Ltd. v. Staton Techiya, LLC*, IPR2022-01099, Paper No. 10 (PTAB Jan. 3, 2023)

and any other materials I refer to in this declaration in support of my opinions.

19. All of the opinions contained in this declaration are based on the documents I reviewed and my knowledge and professional judgment. My opinions have also been guided by my appreciation of how a person of ordinary skill in the art would have understood the claims and the specification of the '666 patent at the time of the alleged invention, which I have been asked to initially consider as no earlier than January 22, 2007. My opinions reflect how one of ordinary skill in the art would have understood the '666 patent, the prior art to the patent, and the state of the art at the time of the alleged invention.

20. Based on my experience and expertise, it is my opinion that certain references disclose and/or suggest, alone or in combination, all the features recited in claims 1-11 ("challenged claims") of the '666 patent, as I discuss in detail below.

IV. LEVEL OF ORDINARY SKILL IN THE ART

21. I have been informed and understand that, in the context of an invalidity analysis, a person having ordinary skill in the art is a hypothetical person who looks to prior art at the time of the invention. I further understand that the factors that may be considered in determining the level of ordinary skill include: (1) the problems encountered in the art; (2) the prior art solutions to the problems encountered in the art; (3) the rapidity of innovation; (4) the sophistication of the technology; and (5) the education level of active workers in the field. I understand that these factors need

not all be considered for the analysis and that one or more of these factors may control.

22. I was asked to provide my opinion on the level of one of ordinary skill in the art with respect to the alleged invention of the '666 patent as of January 22, 2007. Based on my consideration of the factors above, I believe a person of ordinary skill in the art would have had bachelor's degree in electrical engineering, computer science, audio engineering or a similar field and two years of experience with hearing instruments, hearing aids and/or hearing protectors. The person having ordinary skill in the art could have also obtained similar knowledge and experience through other means.

23. As of January 22, 2007, I met, and in fact exceeded, the qualifications of a person of ordinary skill in the art. To be clear, all of my opinions in this declaration are from the perspective of one of ordinary skill in the art as I have defined it here during the relevant timeframe.

V. RELEVANT LEGAL STANDARDS

24. I am not an attorney and offer no legal opinions, but in the course of my work, I have had experience studying and analyzing patents and patent claims from the perspective of a person skilled in the art.

25. For the purposes of this declaration, I have been informed about certain aspects of the law that are relevant to forming my opinions. My understanding of the law is as follows:

26. Requesters' counsel have informed me that patents issued and references published more than one year prior to the alleged date of invention of the '666 patent, which I have been informed is January 22, 2007, are "prior art" to the '666 patent.

27. Requesters' counsel have informed me that to anticipate a patent claim a single reference must contain all of the elements and limitations recited in the claim either expressly or inherently.

28. Requesters' counsel have informed me that for prior art to inherently disclose a claimed limitation, the prior art need not expressly disclose the limitation, so long as the claimed limitation necessarily flows from a disclosure in the prior art.

29. Requesters' counsel have informed me that a patent claim can be considered to have been obvious to a person of ordinary skill in the art at the time the application was filed in view of the prior art. This means that, even if all of the requirements of a claim are not found in a single prior art reference, the claim is not patentable if the differences between the subject matter in the prior art and the subject matter in the claim would have been obvious to a person of ordinary skill in the art at the relevant time, which I have been informed is January 22, 2007.

30. Requesters' counsel have informed me that a determination of whether a claim would have been obvious should be based upon several factors, including, among others:

- the level of ordinary skill in the art at the time the application was filed;
- the scope and content of the prior art; and
- what differences, if any, existed between the claimed invention and the prior art.

31. Requesters' counsel have informed me that a single prior art reference can render a patent claim obvious if any differences between that reference and the claims would have been obvious to a person of ordinary skill in the art. Alternatively, the teachings of two or more references may be combined in the same way as disclosed in the claims, if such a combination would have been obvious to one having ordinary skill in the art. In determining whether a combination based on either a single reference or multiple references would have been obvious, it is appropriate to consider, among other factors:

- whether the teachings of the prior art references disclose known concepts combined in familiar ways that, when combined, would yield predictable results;

- whether a person of ordinary skill in the art could implement a predictable variation, and would see the benefit of doing so;
- whether the claimed elements represent one of a limited number of known design choices, and would have a reasonable expectation of success by those skilled in the art;
- whether a person of ordinary skill would have recognized a reason to combine known elements in the manner described in the claim;
- whether there is some teaching or suggestion in the prior art to make the modification or combination of elements claimed in the patent; and
- whether the innovation applies a known technique that had been used to improve a similar device or method in a similar way.

32. Requesters' counsel has informed me that one of ordinary skill in the art has ordinary creativity and is not an automaton. Requesters' counsel has also informed me that in considering obviousness, it is important not to determine obviousness using the benefit of hindsight derived from the patent being considered.

VI. SUMMARY OF OPINIONS

33. For the reasons I discuss below, it is my opinion that claims 1-11 of the '666 patent would have been rendered obvious by the prior art.

VII. OVERVIEW OF THE '666 PATENT

34. The '666 patent explains in the Background section that "Environmental noise is constantly presented in industrialized societies given the ubiquity of external sound intrusions":

> Since the advent of industrialization over two centuries ago, the human auditory system has been increasingly stressed to tolerate high noise levels to which it had hitherto been unexposed. Recently, human knowledge of the causes of hearing damage have been researched intensively and models for predicting hearing loss have been developed and verified with empirical data from decades of scientific research. Yet it can be strongly argued that the danger of permanent hearing damage is more present in our daily lives than ever, and that sound levels from personal audio systems in particular (i.e. from portable audio devices), live sound events, and the urban environment are a ubiquitous threat to healthy auditory functioning across the global population.
>
> Environmental noise is constantly presented in industrialized societies given the ubiquity of external sound intrusions. Examples include people talking on their cell phones, blaring music in health clubs, or the constant hum of air conditioning systems in schools and office buildings.
>
> Excess noise exposure can also induce auditory fatigue, possibly comprising a person's listening abilities. On a daily basis, people are exposed to various environmental sounds and noises within their

> environment, such as the sounds from traffic, construction, and industry.

'666 patent, 1:32-54. According to the '666 patent, people "drown out" this noise with portable audio playback devices delivering sound directly to the ear canal at high levels, and that doing so has a number of drawbacks:

> To combat the undesired cacophony of annoying sounds, people are arming themselves with portable audio playback devices to drown out intrusive noise. The majority of devices providing the person with audio content do so using insert (or in-ear) earbuds. These earbuds deliver sound directly to the ear canal at high sound levels over the background noise even though the earbuds generally provide little to no ambient sound isolation. Moreover, when people wear earbuds (or headphones) to listen to music, or engage in a call using a telephone, they can effectively impair their auditory judgment and their ability to discriminate between sounds. With such devices, the person is immersed in the audio experience and generally less likely to hear warning sounds within their environment. In some cases, the user may even turn up the volume to hear their personal audio over environmental noises. It also puts them at high sound exposure risk which can potentially cause long term hearing damage.
>
> With earbuds, personal audio reproduction levels can reach in excess of 100 dB. This is enough to exceed recommended daily sound exposure levels in less than a minute and to cause permanent acoustic trauma. Furthermore, rising population densities have continually

> increased sound levels in society. According to researchers, 40% of the European community is continuously exposed to transportation noise of 55 dBA and 20% are exposed to greater than 65 dBA.

Id., 1:55-2:14. The '666 patent contends that "[a] need therefore exists for enhancing the user's ability to 20 listen in the environment without harming his or her hearing faculties." *Id.* 2:19-21.

35. The '666 patent addresses this purported need with an "earpiece" device:

> The present invention relates to a device that monitors sound directed to an occluded ear, and more particularly, though not exclusively, to an earpiece and method of operating an earpiece that detects acute sounds and allows the acute sounds to be reproduced in an ear canal of the occluded ear.

'666 patent, 1:23-28. The '666 patent recites an embodiment of this earpiece:

> In a first embodiment, an earpiece can include an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change. The acute sound can be reproduced within the ear canal via the ECR responsive to detecting the acute sound.

Id. 2:28-36. In some embodiments, an Ear Canal Microphone (ECM) is also included in the earpiece:

> The earpiece can further include an Ear Canal Microphone (ECM) to measure an ear canal sound level (ECL) within the ear canal.

Id., 2:49-51. The ECM may also be used to determine if a sound is created by the user's voice. *Id.*, 10:8-40.

A. Prosecution History of the '666 Patent

36. The applicant filed the '396 application on August 7, 2020, which subsequently issued as the '666 patent. (PAT-B, Aug. 7, 2020 Application.) The '396 application is a continuation of U.S. Patent Application No. 16/669,490, which issued as U.S. Patent No. 10,810,989. (PAT-A, Cover; PAT-B, Aug. 7, 2020 Application Data Sheet.) As filed, the '396 application contained 14 claims. (PAT-B, Aug. 7, 2020 Application, Claims.)

37. On September 28, 2020, the Office issued a Non-Final Office Action, rejecting various claims under § 112, first paragraph, for failing to comply with the enablement requirement, rejecting a single dependent claim under § 112, second paragraph, as being indefinite, and rejecting the lone independent claim and several of the dependent claims under § 102(e) as being anticipated by U.S. Patent Application Publication No. 2008/0240458 A1 ("Goldstein '458"). (PAT-B, Sep. 28, 2020 Non-Final Office Action.) In response, on December 28, 2020, the

applicant amended the claims to address the § 112 rejections and argued that the § 102(e) rejection based on Goldstein '458 was improper on the grounds that Goldstein '458 was not prior art. (PAT-B, Dec. 28, 2020 Response to Office Action.)

38. On January 28, 2021, the Office issued another Non-Final Office Action, rejecting each of the pending claims under § 102(b) as being anticipated by U.S. Patent Application Publication No. 2004/0179694 A1 ("Alley"). (PAT-B, Jan. 28, 2021 Non-Final Office Action.) In response, on March 18, 2021, the applicant amended the claims to recite that the earphone includes two microphones, in addition to other features, and argued that the § 102(b) rejection based on Alley should be withdrawn because, *inter alia*, Alley allegedly did not disclose two microphones in an earphone. (PAT-B, Mar. 18, 2021 Response to Office Action.)

39. On April 2, 2021, the Office issued a Final Office Action, rejecting only dependent claims 2 and 3 under § 112, first paragraph, for failing to comply with the written description requirement, and indicating that each of the remaining claims were in condition for allowance. (PAT-B, Apr. 2, 2021 Final Office Action.) In response, on May 14, 2021, the applicant canceled rejected dependent claims 2 and 3 without making any additional amendments. (PAT-B, May 14, 2021 Response to Office Action.)

40. On June 1, 2021, the Office issued another Non-Final Office Action, objecting to several dependent claims for using inconsistent terminology, rejecting

claims 1, 4-9, 11, 14, and 15 under § 103(a) as being unpatentable over U.S. Patent Application Publication No. 2008/0165988 A1 ("Terlizzi") in view of U.S. Patent Application Publication No. 2006/0262938 A1 ("Gauger"), and rejecting claim 10 under § 103(a) as being unpatentable over Terlizzi in view of Gauger and further in view of U.S. Patent No. 5,208,867 ("Stites"). (PAT-B, June 1, 2021 Non-Final Office Action.) In response, on August 4, 2021, the applicant amended the independent claims to recite "wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted" and "wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted," and argued that neither Terlizzi, Gauger, nor Stites disclosed these features. (PAT-B, Aug. 4, 2021 Response to Office Action.) Notably, the applicant asserted that support for these features is allegedly found in Figures 1 and 2 of the application and that "the terms 'face toward' and 'face away' can be interpreted as an orientation of an axis running through the middle of the port intersecting with a user or not, respectively." (*Id.*, 6-7).

41. On August 16, 2021, the Office issued another Final Office Action, rejection claims 1, 4-11, and 14 under § 102(e) as being anticipated by U.S. Patent Application Publication No. 2016/0249128 A1 ("Goldstein '128"), rejecting claims 1, 4-9, 11, 14, and 15 under § 103(a) as being unpatentable over Terlizzi in view of

Dijkstra (PA-C) and Gauger, and rejecting claim 10 under § 103(a) as being unpatentable over Terlizzi in view of Dijkstra and Gauger and further in view of Stites. (PAT-B, Aug. 16, 2021 Final Office Action.) The Office noted that Dijkstra taught the "first microphone port" and "second microphone port" limitations that applicant had added via amendment on August 4, 2021. (*Id.*, 3-5.) In response, on October 25, 2021, the applicant argued that the § 102(e) rejection based on Goldstein '128 was improper on the grounds that Goldstein '128 was not prior art and requested reconsideration of the rejection under § 103(a) based on Terlizzi in light of a declaration submitted under 37 C.F.R. § 1.131 purporting to antedate Terlizzi. (PAT-B, Oct. 25, 2021 Response to Office Action.) Notably, the applicant did not dispute the Office's determination that Dijkstra taught the "first microphone port" and "second microphone port" limitations.

42. On December 22, 2021, the Office issued a Notice of Allowance. (PAT-B, Dec. 22, 2021 Notice of Allowance.) The Statement of Reasons for Allowance in the Notice did not provide any substantive discussion as to how the claims were allowable over the prior art. Rather, the Office simply indicated that the declaration submitted October 25, 2021, was sufficient to overcome the rejections based on Goldstein '128 and Terlizzi. (*Id.*, 2.) The applicants did not submit any comments on the Statement of Reasons for Allowance.

B. Prosecution History of the '377 Patent

43. According to the face of the '666 patent, the '377 patent is the great grand-parent to the '666 patent, and in this section I summarize aspects of its prosecution history relevant to my opinions. On December 18, 2014, the applicant submitted the following claim 1 for prosecution. (PAT-D, Dec. 18, 2014 Application.)

> What is claimed is:
>
> 1. A method for acute sound detection and reproduction, the method comprising: measuring an external ambient sound level (xASL); monitoring a change in the xASL for detecting an acute sound; estimating a proximity of the acute sound; and upon detecting the acute sound and it's proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece.

44. On August 26, 2016, the Office mailed a rejection of claim 1 as obvious in view of Victorian. (PAT-D, Aug. 26, 2016 Non-Final Office Action.)

45. In response, on November 28, 2016, the applicant submitted the following amendments to claim 1, arguing that "the cited portions of Victorian fail to disclose or suggest" the added claim limitations. (PAT-D, Nov. 28, 2016 Response to Office Action.)

1. (Currently Amended) A method for acute sound detection and reproduction, the method comprising:

measuring an external ambient sound level (xASL) using a microphone;

monitoring a change in the xASL for detecting an acute sound;

estimating a proximity of the acute sound by performing a cross correlation between at least two microphones, where the at least two microphones includes at least the microphone for measuring the external ambient sound level and identifying a peak in the cross correlation; and

upon detecting the acute sound and ~~it's~~ a time lag in the acute sound between the at least two microphones~~proximity~~, reproducing the acute sound ~~within an ear canal~~with an ear canal receiver, where an earpiece having the ear canal receiver is configured to ~~the ear canal is~~ at least partially occlude an ear canal ~~occluded by an earpiece~~.

46. Then, on May 10, 2017, the Office mailed a rejection of claim 1 as obvious in view of Rast in combination with U.S. Patent No. 5,867,581 ("Obara"). (*Id.*) The examiner took the position that Rast disclosed each limitation, except:

> *Rast may not explicitly teach nor restrict that his cross-correlation is used to detect a time lag in the acute sound between the at least two microphones in order to estimate a proximity of the acute sound and determining a direction from the time lag for the purpose of performing directional characteristics of sound that will enhance the safety and well-being of a person isolated from ambient sounds.*

(PAT-D, May 10, 2017 Final Office Action.) The Office stated that Obara disclosed this limitation and that a person having ordinary skill in the art would have combined Rast with Obara to "us[e] a known technique [disclosed in Obara] to improve a similar method" in Rast. (*Id.*)

47. In response, on September 11, 2017, the applicant submitted further amendments to claim 1. (PAT-D, Sep. 11, 2017 Response to Final Office Action.) The applicant argued that "the sound is determined to be an 'acute sound' if the sound has a positive value related to the threshold test." (*Id.*) This, according to the applicant, "differs substantially from what is taught by Rast in view of Obara" because "Rast teaches monitoring sounds from the ambient and comparing the sounds against sound files stored in memory. Once a sound is identified or correlated with a specific sound in memory the sound is then passed to the User." (*Id.*)

1. (currently amended) A method for acute sound detection and reproduction, the method comprising:

measuring an external ambient sound level (xASL) using a microphone;

monitoring a change in the xASL ~~for detecting an acute sound~~ with at least one processor;

detecting a sound with the microphone wherein the sound is converted to a level and wherein the sound is determined to be an acute sound when the level is above a predetermined threshold;

estimating a proximity of the acute sound by performing a cross correlation between at least two microphones with the at least one processor, wherein the at least two microphones includes ~~at least~~ the microphone for measuring the ambient sound level and identifying a peak in the cross correlation; and

upon detecting the acute sound and a time lag in the acute sound between the at least two microphones, reproducing the acute sound with an ear canal receiver, where an earpiece having the ear canal receiver is configured to at least partially occlude an ear canal.

(*Id.*)

48. On October 5, 2017, the Office mailed an Advisory Action, maintaining the rejection of the claims as obvious in view of Rast and Obara. (PAT-D, Oct. 5, 2017 Advisory Action.)

49. In response, on November 18, 2017, the applicant submitted further amendments to claim 1. (PAT-D, Oct. 6, 2017 Response to Advisory Action.)

1. (currently amended) A method for acute sound detection and reproduction, the method comprising:

measuring an external ambient sound level (xASL) using a microphone;

monitoring a change in the xASL ~~for detecting an acute sound~~ with at least one processor;

detecting a sound with the microphone wherein the sound is converted to a level and wherein the sound is determined to be an acute sound when the level is above a predetermined threshold;

estimating a proximity of the acute sound by performing a cross correlation between at least two microphones with the at least one processor, wherein the at least two microphones includes ~~at least~~ the microphone for measuring the external ambient sound level and identifying a peak in the cross correlation; and

upon detecting the acute sound and a time lag in the acute sound between the at least two microphones, reproducing the acute sound with an ear canal receiver if the acute sound is bleow a danger threshold, where an earpiece having the ear canal receiver is configured to at least partially occlude an ear canal.

(*Id.*)

50. On December 29, 2017, the Office mailed a non-final rejection, again maintaining the rejection of claim 1 in view of Rast and Obara. (PAT-D, Dec. 29, 2017 Non-Final Office Action.)

51. In response, on March 28, 2018, the applicant presented a further amendment to claim 1, arguing that Rast does not disclose the added limitations:

> *Rast does not teach, suggest, or show estimating a background noise level (BNL) in paragraph [0060] of Rast as claimed by Applicant. Rast does not teach, suggest, or show identifying at least one peak level stored (SPL) from the ambient sound stored within an audio buffer in paragraph [0060] as claimed by Applicant. Rast does not teach, suggest, or show generating a SPL-BNL difference in paragraph [0060] as claimed by Applicant. Finally, Rast does not teach, suggest, or show comparing the SPL-BNL difference to a first predetermined threshold where an acute sound is detected when the SPL-BNL difference is greater than the first predetermined threshold.*

(PAT-D, Mar. 28, 2018 Response to Non-Final Office Action.)

1. (Currently Amended) A method for acute sound detection and reproduction, the method comprising:

measuring an external ambient sound level (xASL) using a microphone wherein a processor applies signal processing to condition and compensate a signal from the microphone and outputs a processed signal;

estimating a background noise level (BNL) from the processed signal;

storing the processed signal in an audio buffer;

identifying and measuring at least one peak level (SPL) in the processed signal stored in the audio buffer;

subtracting the estimated background noise level (BNL) from the measured at least one peak level (SPL) to generate a SPL-BNL difference;

comparing the SPL-BNL difference to a first predetermined threshold wherein an acute sound has been detected when the SPL-BNL difference is greater than the first predetermined threshold;

~~monitoring a change in the xASL with at least one processor;~~

~~detecting a sound with the microphone wherein the sound is converted to a level and wherein the sound is determined to be an acute sound when the level is above a predetermined threshold;~~

estimating a proximity of the acute sound by performing a cross correlation between at least two microphones with the at least one processor, wherein the at least two

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microphones includes the microphone for measuring the external ambient sound level and identifying a peak in the cross correlation; and

upon detecting the acute sound and a time lag in the acute sound between the at least two microphones, reproducing the acute sound with an ear canal receiver after determining the acute sound was not a user-generated sound ~~if the acute sound is above a danger threshold, where an earpiece having the ear canal receiver is configured to at least partially occlude an ear canal.~~

(*Id.*)

52. On July 18, 2018, the Office mailed a Notice of Allowance. (PAT-D, Jul. 18, 2018 Notice of Allowance.)

VIII. OVERVIEW OF THE PRIMARY PRIOR ART REFERENCES

A. Nemirovski

53. Nemirovski issued on November 11, 2003, and is entitled "SENSOR PAIR FOR DETECTING CHANGES WITHIN A HUMAN EAR AND PRODUCING A SIGNAL CORRESPONDING TO THOUGHT, MOVEMENT, BIOLOGICAL FUNCTION AND/OR SPEECH." (Nemirovski (PA-A), Cover.) Accordingly, Nemirovski is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

54. Nemirovski is directed to "a sensor placed with respect to an ear" that is "used to detect physical or mental actions of a person such as speech" and "and provides an output which can be converted into system control functions." (*Id.*, 1:13-20.) As background, Nemirovski explains that there were known devices in the prior art to enable handicapped persons to control devices, such as controllers (e.g., joysticks) that a handicapped person could manipulate with their mouth or other systems that used an infrared signal that the user could manipulate by blinking. (*Id.*, 1:44-65.) According to Nemirovski, these prior art devices were disadvantageous for various reasons, including a lack of convenience and potential for injury. (*Id.*) Nemirovski further explains that voice recognition software had become available

and that typical systems required a microphone placed in front of the user's mouth. (*Id.*, 1:66-2:4.)

55. In view of the prior art, Nemirovski discloses a "more aesthetically pleasing and comfortable" device, including an "internal microphone" that is "disposed with respect to an ear of a user and detects air pressure changes within the ear." (*Id.*, 4:1-7.) The sensor can "detect a person's speech" which can be used for many purposes including voice recognition or re-transmitting the detected speech with a cell phone. (*Id.*, 4:14-23.) In addition, Nemirovski discloses implementing in the sensor an external microphone to detect "air pressure changes, or sounds, originating outside the user's ear." (*Id.*, 4:24-33.) The sensor can also include a speaker. (*Id.*, FIG. 1C, 8:6-9, 14:1-6.) Nemirovski explains that this device can be used for many purposes. (*Id.*, 5:1-27.) For example, it may be used by a handicapped person to control operation of a computer, machinery, or a video game. (*Id.*)

56. In various embodiments, Nemirovski describes a sensor 12 having an internal microphone 34 and an external microphone 54 each disposed on a housing 150 "to be inserted at least part way into the ear of a person." (*Id.*, 9:5-28.) FIG. 2B, reproduced and annotated below, "illustrates sensor 12 inserted at least partially into the ear 100 of a person" (id., 11:63-67):

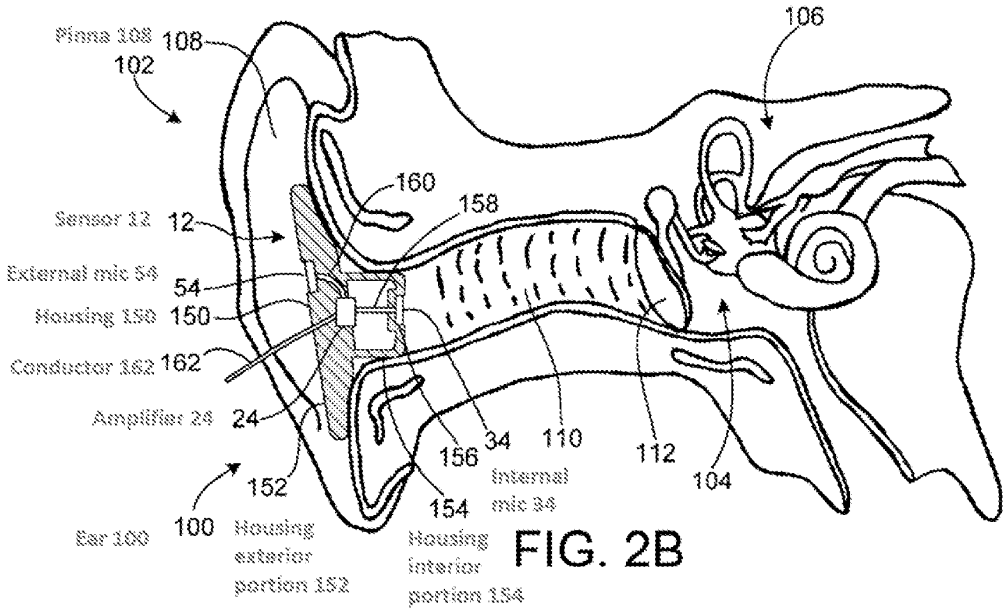


FIG. 2B

Nemirovski, FIG. 2B (annotated)[2]

57. Nemirovski discloses that the disclosed sensor is disposed within a housing 150, made from "sound absorbing (or sound proofing) material" and which "at least partially blocks externally generated soundwaves." (*Id.*, 12:19-25; 13:3-7.) Indeed, Nemirovski explains that one of the advantages of its invention is that the internal microphone "detects less environmental, or ambient, noise" than the prior art. (*Id.*, 4:2-5.).

[2] I have added all **bold**/*italics*/color emphases and annotations unless noted otherwise.

B. Rast

58. Rast is a publication of U.S. Patent Application No. 09/841,713, entitled "SYSTEM AND METHOD FOR SELECTIVE CONTROL OF ACOUSTIC ISOLATION IN HEADSETS." (Rast (PA-B), Cover.) Rast was published on November 29, 2001. (*Id.*) Accordingly, Rast is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

59. Rast is directed to "headsets with sound selective acoustical isolation from the external environment." (*Id.*, ¶[0005].) As background, Rast explains that headphones are often used in "applications for communicating between parties while blocking out sounds of the external environment" (*id.*, ¶[0007]), and that this can be inconvenient or dangerous (*id.*, ¶[0008]). For example, the user may be "unable to hear the siren of an emergency vehicle, a scream, a shout, or an associate loudly calling his/her name, while a party listening to music may miss an important phone call, be unable to hear the door bell, or miss their name being called out." (*Id.*, ¶[0008].) Thus, Rast identifies a need for "methods for controlling acoustic isolation, so that important events are not missed and non-important noises can be properly identified and ignored." (*Id.*, ¶[0009].)

60. In response, Rast discloses methods for providing controlled acoustic isolation with various forms of headsets. (*Id.*, Abstract.) Rast discloses a headset by which "the acoustic isolation of the user is modified either manually based on user

input, or automatically based on the sounds registered within the external environment and correlated to a set of stored sound characteristic." (*Id.*, ¶[0010].) The headset includes one or more microphones which "convert sounds within the external acoustical environment to electrical signals" (*Id.*, ¶[0014].) The signals are processed by a DSP/microcontroller and correlated to audio selection criterion stored in memory. (*Id.*) For example, Rast describes allowing a user to program the headset to recognize specific sounds within their specific environment, such as "alarms, screams, horns, screeching brakes, phones ringing, spoken phrases (i.e. "Hey!", "Hello", "Wearer's Name", "Help"), along with additional acoustic patterns." (*Id.*, ¶¶[0021]-[0022]; see also *id.*, ¶¶[0008], [0018], [0055], [0059], and [0062].) When the signal processing unit within, or attached to, the headset, recognizes the environmental audio as matching the criterion, then the mode of the headset is changed in response. (*Id.*, ¶[0022].) For example, in a "hearthrough" mode, sound registered by the microphones is conditioned and directed to the audio annunciators within the earcups. (*Id.*, ¶[0051].) "Additionally, the audio source being listened to may be either attenuated, or disconnected, such that the amplified external sounds from the microphones may be more clearly heard by the wearer." (*Id.*; see also *id.*, ¶[0058].) Rast also explains that the change to "hearthrough" mode may be accomplished by "the wearer themselves saying a word or phase [sic] which has been programmed or recorded for discrimination." (*Id.*, ¶[0055].)

C. Dijkstra

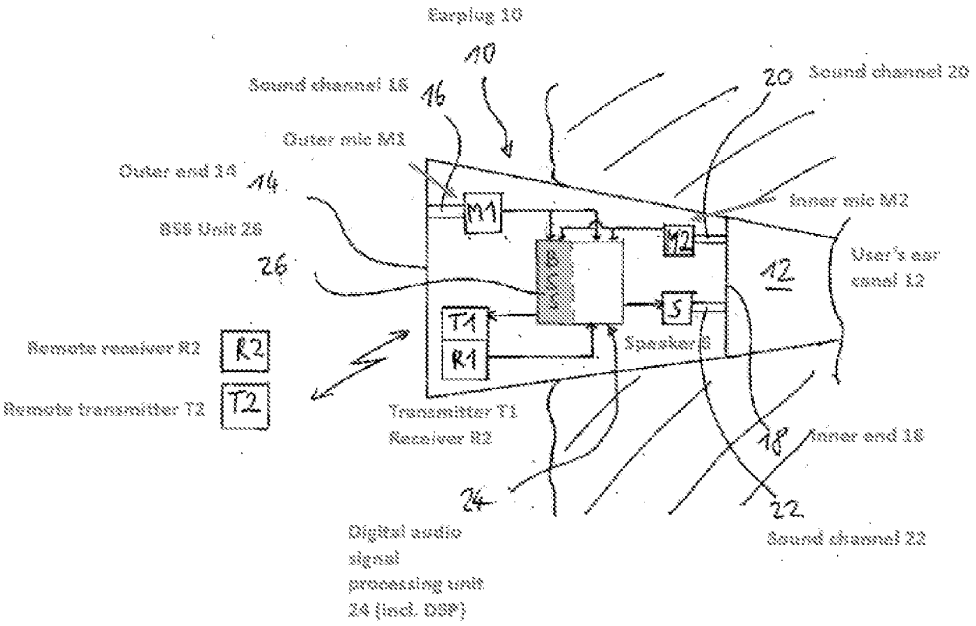
61. Dijkstra is a publication of European Patent Application No. 05028366.2, entitled "SYSTEM AND METHOD FOR SEPARATION OF A USERS VOICE FROM AMBIENT SOUND." (Dijkstra (PA-C), Cover.) Dijkstra was published on March 29, 2006. (*Id.*) Accordingly, Dijkstra is prior art to the '666 patent under at least 35 U.S.C. § 102(a).

62. Dijkstra 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." (*Id.*, ¶[0001].) Dijkstra explains that, for wireless communication between persons in a noisy environment, e.g., policemen, soldiers, and firemen, "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." (*Id.*, ¶[0002].)

63. To accomplish these objectives, Dijkstra discloses a "hearing protection device" comprising an earplug inserted into the user's ear and which implements a blind source separation ("BSS") algorithm to obtain an intelligible speech signal for use in wireless communications. (*Id.*, ¶¶[0010], [0013].) The

earplug may have a customized hard shell conforming to the inner shape of the user's ear, or a generic soft shell that adapts to the shape of the user's ear. (*Id.*, ¶[0013].)

64. The earplug includes an outer microphone M1, an inner microphone M2, and an audio signal processing unit 24 including a BSS unit 26. (*Id.*, ¶[0029.) The outer microphone M1 is open to the outer end 14 of the earplug via a sound channel 16 and primarily picks up ambient sound. (*Id.*, ¶¶[0029], [0031].) The inner microphone M2 is open to the inner end 18 of the earplug via sound channel 20 and primarily picks up the user's voice. (*Id.*) The audio signal processing unit includes a digital signal processor and program and data memory components. (*Id.*, ¶[0029]). The processing unit also includes a transceiver and receiver for communication with a mobile telephone. (*Id.*, ¶¶[0030], [0032], [0033].)



Dijkstra, FIG. 1 (annotated)

65. The signals from both microphones are passed to the BSS unit 26. (*Id.*) The output signal of the BSS unit 26 is transmitted to another user with a similar earplug or a mobile telephone. (*Id.*, ¶[0032].) Having undergone BSS, the output signal contains the user's voice separated from background noise, resulting in "good intelligibility of speech." (*Id.*, ¶¶[0007]-[0008]; see also *id.*, ¶¶[0009]-[0012].) "Such a system could be used by any persons who need to communicate in a noisy environment, such as workers, soldiers, firemen, etc." (*Id.*, ¶[0032].)

66. Dijkstra also discloses passing the signals from both microphones to the audio signal processing unit 24 for other audio signal processing besides BSS. (*Id.*, ¶[0034].) For example, the ambient sound picked up by microphone M1 may be provided to the speaker "for enabling perception of desired ambient sound during times when the noise level is low so that the hearing protection function provided by the earplug 10 is not needed." (*Id.*).

D. Van Doorn

67. Van Doorn is a publication of U.S. Patent Application No. 10/236,649, entitled "MICROPHONE WITH IMPROVED SOUND INLET PORT." (Van Doorn (PA-D), Cover.) Van Doorn was published on March 11, 2004. (*Id.*). Thus, Van Doorn is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

68. Van Doorn relates to "a microphone or listening device with an improved sound inlet port." (*Id.*, ¶[0001].). As background, Van Doorn explains that "[m]ost prior art microphones" "have a sound inlet that includes a large inlet nozzle for receiving sound from the ambient environment." (*Id.*, ¶[0003]; *see also id.*, ¶¶[0017]-[0019], FIG. 1.) According to Van Doorn, "[t]he large dimensions of the inlet nozzle can be a problem because hearing aids often have very limited space." (*Id.*).

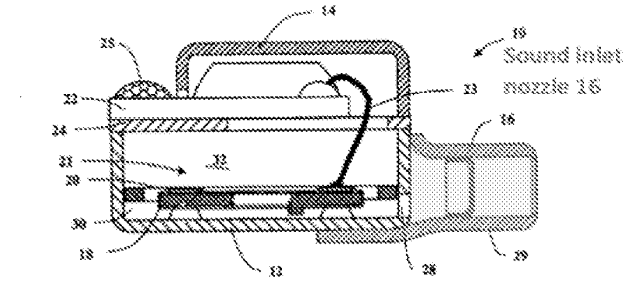


FIG. 1A
Prior Art

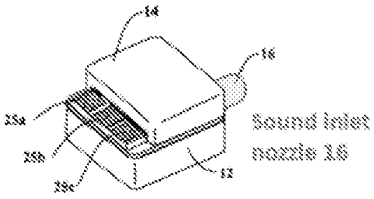


FIG. 1B
Prior Art

Van Doorn, FIGs. 1A-B (annotated)

69. Van Doorn's solution is a microphone with a "sound inlet plate" that is "made very thin so that it does not extend substantially away from the housing." (*Id.*, ¶[0006].) "The inventive sound inlet plate is useful on omni-directional and

directional microphones." (*Id.*, ¶[0007]). More particularly, Van Doorn discloses a miniature microphone with a "cup-shaped sound input plate 50" with "an outer wall 52" and a "sound port 56, which is an opening in the outer wall 52." (*Id.*, ¶[0020].)

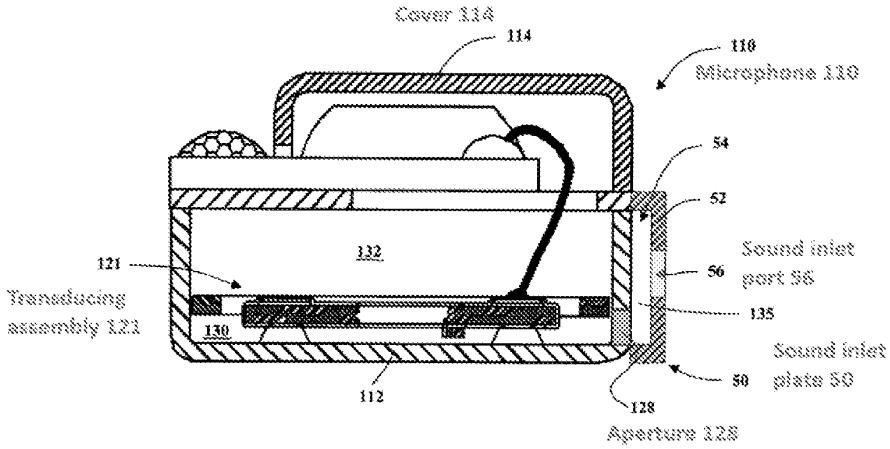


FIG. 3

Van Doorn, FIG. 3 (annotated)

70. "In operation, the sound inlet plate 50 receives sound through the sound inlet port 56 in its exterior wall 52. The sound propagates through a passageway 135 that is defined by the interior recess 54 of the plate 50 and the exterior wall of the housing 112 adjacent to the aperture 128. Eventually, the sound is transmitted through the aperture 128 and acts upon the transducing assembly 121." (*Id.*, ¶0022].) "Unlike prior art systems where the sound inlet extends substantially away from the housing of the microphone (such as the nozzle 16 in FIGS. 1A and 1B), the sound inlet plate 50 only protrudes slightly away from the housing 112. One design aspect

leading to the minimal protrusion feature of the plate 50 is the fact that the sound port 56 is simply an opening or hole in the outer wall 52." (*Id.*, ¶[0023].)

E. Victorian

71. Victorian is a publication of European Patent Application No. 04255520.1, entitled "EXTERNAL EAR CANAL VOICE DETECTION." (Victorian (PA-E), Cover.) Victorian was published on March 30, 2005. (*Id.*) Accordingly, Victorian is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

72. Victorian is directed to "ear-level communication systems" for communicating with remote devices such as a cellular telephone via full duplex audio. (*Id.*, ¶[0001]-[0002].) Victorian discloses, for example, an "ear-level device 110." (*Id.*, ¶[0019].) FIG. 1 of Victorian, which is reproduced and annotated below, depicts an ear-level communication system 100 including ear-level device 110:

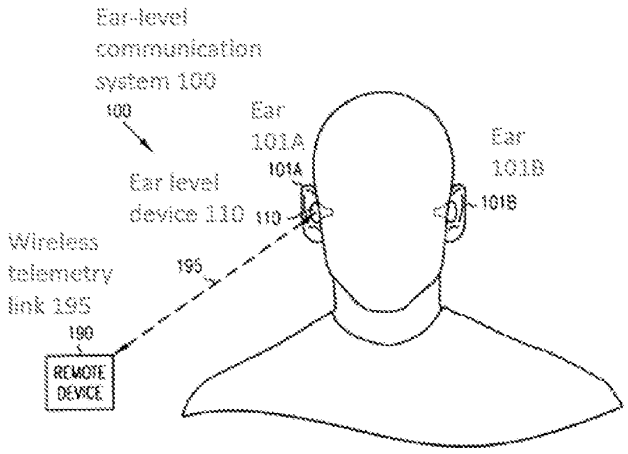


FIG. 1

Victorian, FIG. 1 (annotated)

Victorian discloses various configurations for "ear-level device 110," including an "in-the-canal (ITC) device 310B." (*Id.*, ¶[0025]). FIG. 3B of Victorian, which is reproduced and annotated below, depicts ear-level device 110 with an ITC device 310B:

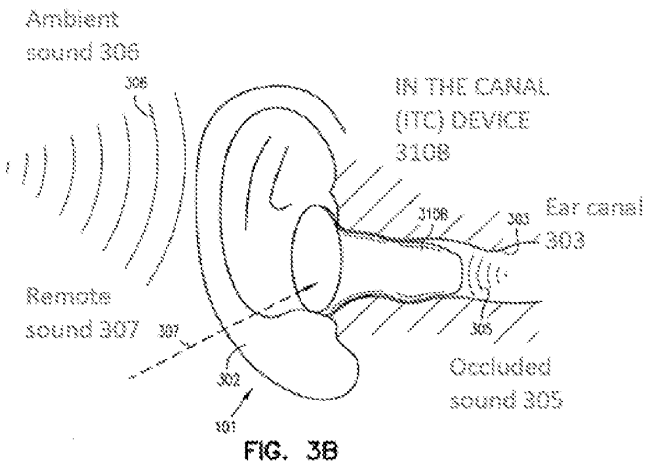


FIG. 3B

Victorian, FIG. 3B (annotated)

73. By reference to FIG. 4A, which is reproduced below, Victorian discloses the circuit of the single ear-level device 110. *Id.*, [0029].

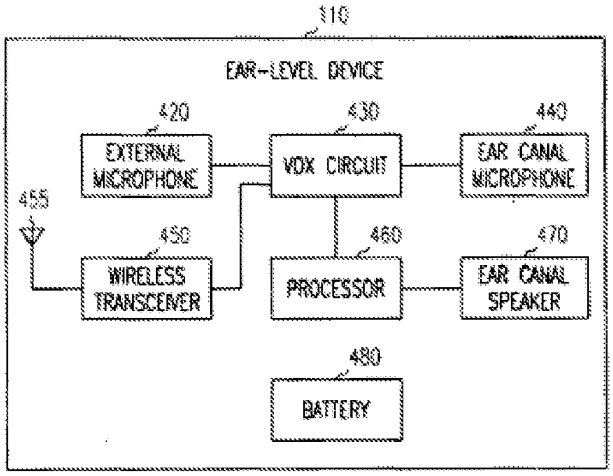


FIG. 4A

Victorian, FIG. 4A

74. The circuit includes, *inter alia*, the following features:

- an "external microphone 420," which "picks up ambient sound";
- an "ear canal microphone 440," which detects the "speech of the person wearing ear-level device 110";
- an "ear canal speaker 470" which "transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal";
- a "processor 460" which "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"; and

- a voice operated exchange (VOX) circuit 430.

(*Id.*, ¶[0030].) Victorian discloses that processor 460 of ear-level device 100 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. (*Id.*)

75. By reference to FIG. 4B, which is reproduced and annotated below, Victorian discloses the operation of VOX 430. (*Id.*, ¶[0031].) Victorian uses gating modules to reduce echo or ringing caused by the feedback loop between the ear canal speaker and ear canal microphone 440. (*Id.*) If a module is "on," that means sound is passed through; if "off," that sound is blanked or attenuated. (*Id.*, [0032].)

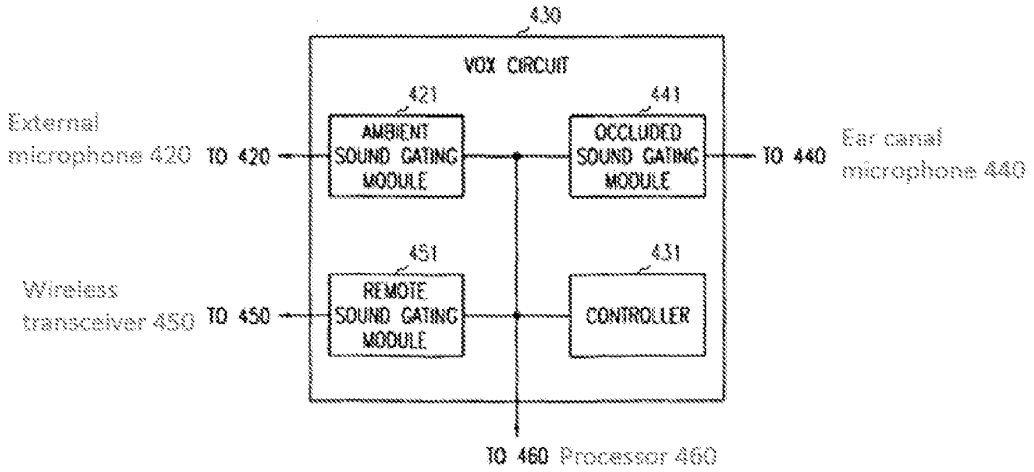


FIG. 4B

Victorian, FIG. 4B (annotated)

76. Victorian discloses a gating rule that allows the transmission of both the remote sound and the ambient sound to the ear canal. (*Id.*, ¶[0038].)

F. Beasley

77. Beasley is a publication of U.S. Patent Application No. 11/328,890, entitled "HEADSET AUDIO BYPASS APPARATUS AND METHOD." (Beasley (PA-F), Cover.) Beasley was published on July 13, 2006. (*Id.*) Accordingly, Beasley is prior art to the '666 patent under at least 35 U.S.C. § 102(a).

78. Beasley is directed to techniques for "control[ling] the relative level of audio and background sound for an audio device such as a headset or headphones." (*Id.*, Abstract.) Beasley explains that "many contemporary headsets offer active or passive noise rejection or noise cancellation features to provide an enhanced listening experience." (*Id.*, ¶[0007].) These systems may for example "utilize electronic cancellation of undesirable background audio or noise." (*Id.*) Unfortunately, according to Beasley, the advantages offered by active noise cancellation headsets also result in a disadvantage as well. (*Id.*, ¶[0008].) For example, "the user often experiences difficulty participating in a conversation with others around him or her or otherwise hearing certain background audible content that may be of interest." (*Id.*)

79. In view of these disadvantages, Beasley discloses "[a]n apparatus and method [that] provides control over the relative level of audio and background sound

for an audio device such as a headset or headphones." (*Id.*, Abstract.) More particularly, Beasley discloses "a system and method for providing an audio bypass feature" in headsets "where it may be useful or desirable for a user to allow outside audio information such as, for example, a conversation to pass through the headset so this information can be heard by the user." (*Id.*, ¶[0027].) Beasley describes providing a control signal 320 to the audio bypass module to allow an external audio signal "to pass through to speaker 332." (*Id.*, ¶[0035].) Beasley explains that the control signal "can be generated by user interaction (e.g., via a control panel or other user interface) or generated automatically depending on the implementation." (*Id.*, ¶[0035]; *see also id.*, ¶¶[0037], [0040], [0061].) "[F]or example, the user can program the device to enter the audio bypass mode when a particular word or phrase is spoken"—such as "Could you repeat that?"—"[t]hereby enabling the audio bypass mode when the user asks another person to repeat what they have just said." (*Id.*, ¶[0061].) Thus, Beasley discloses a device configured to automatically enter an audio bypass mode "when the device senses that the user has spoken." (*Id.*)

G. Andrea

80. Andrea issued on October 5, 1993, and is entitled "ADAPTIVE NOISE CANCELLATION AND SPEECH ENHANCEMENT SYSTEM AND APPARATUS THEREFOR." (Andrea (PA-G), Cover.) Accordingly, Andrea is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

81. Andrea discloses a headset apparatus for use in an intercommunications system" such as an intercom, telephone, or radio. (*Id.*, Abstract, 5:29-34.) The headset apparatus includes a "speaker 50," a "microphone 60" and a "microphone 70." (*Id.*, 6:45-49.) The microphone 60 "is adapted to detect external background signals" such as "sirens, engine noises and crowd noises." (*Id.*, 6:46-47, 7:11-15.) The microphone 70 is for "feedback/voice transmission" and is mounted in portion 80 of the headset closest to the ear canal opening of the user. (*Id.*, 6:47-51.)

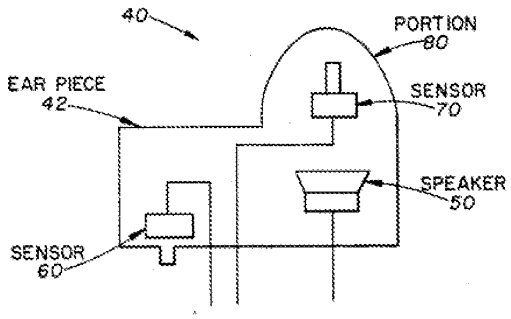
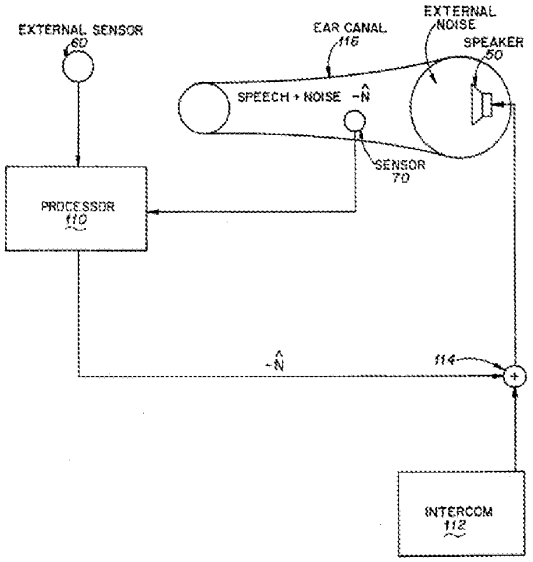


Fig. 4

Andrea, FIG. 4

82. The system also includes a "processor 100," also referred to as an "ANC unit." (*Id.*, 6:15-21.) When operating in the "receive mode," the signals from the microphones 60, 70 are supplied to the processor. (*Id.*, 7:10-15; 5:38-43, 8:39-49, FIG. 13.) The processor processes the received signals using a finite impulse response filter and outputs an anti-noise signal. (*Id.*, 7:15-25.) A combiner circuit

114 combines the anti-noise signal and signal from the intercom to the speaker 50. (*Id.*, 7:25-30.)



Andrea, FIG. 5

H. Boersma

83. Boersma issued on March 8, 2011, and is entitled "HEADSET WITH AMBIENT SOUND." (Boersma (PA-H), Cover.) Boersma was filed on March 8, 2006. Accordingly, Boersma is prior art to the '666 patent under 35 U.S.C. § 102(e).

84. Boersma relates to "a headset for electronic equipment" such as media players or mobile phones. (*Id.*, 1:6-7, 4:38-56.) Boersma identifies several disadvantages of conventional headsets, including that they "can completely block out all other sounds from the environment" which "can be undesirable or even dangerous." (*Id.*, 1:37-46.) For example, a user "may be unable to hear police or

emergency equipment sirens, thereby creating a potentially dangerous situation," and "[i]n a social situation, a user wearing a headset still may desire to hear sound from his/her surroundings." (*Id.*, 1:46-50.) Boersma notes that prior efforts have been made to reproduce a mix of audio signals and ambient sounds in a headset (*id.*, 1:51-56), but those attempts were not "comfortable or convenient" enough, had limited interoperability with different devices, did not provide sufficient control to the user, and restricted the user's ability to receive and appreciate spatial characteristics of ambient sound. (*Id.*, 1:57-67.)

85. In view of the foregoing disadvantages, Boersma discloses a headset 30 "suitable for use with a variety of types of electronic equipment" and which does not rely on processing circuitry and/or controls within a media player or mobile phone to provide a user with the ability to listen to reproduced audio yet still be able to hear ambient sound. (*Id.*, 5:51-59.) Moreover, the headset provides the user "the ability to control the respective levels of the reproduced audio and ambient sound, thereby allowing the user to adjust the listening experience to his or her preference." (*Id.*, 5:59-65.) The device can be used in a variety of environments with different levels of noise, for example in traffic, a crowded train, or in one's living room. (*Id.*, 7:36-46.)

86. In one embodiment, the headset includes a pair of earpieces 38. (*Id.*, 6:14-16.) Each earpiece includes a microphone 50 "for picking up ambient sound" (*id.*, 6:32-33) and a loudspeaker 56 (*id.*, 6:50-51).

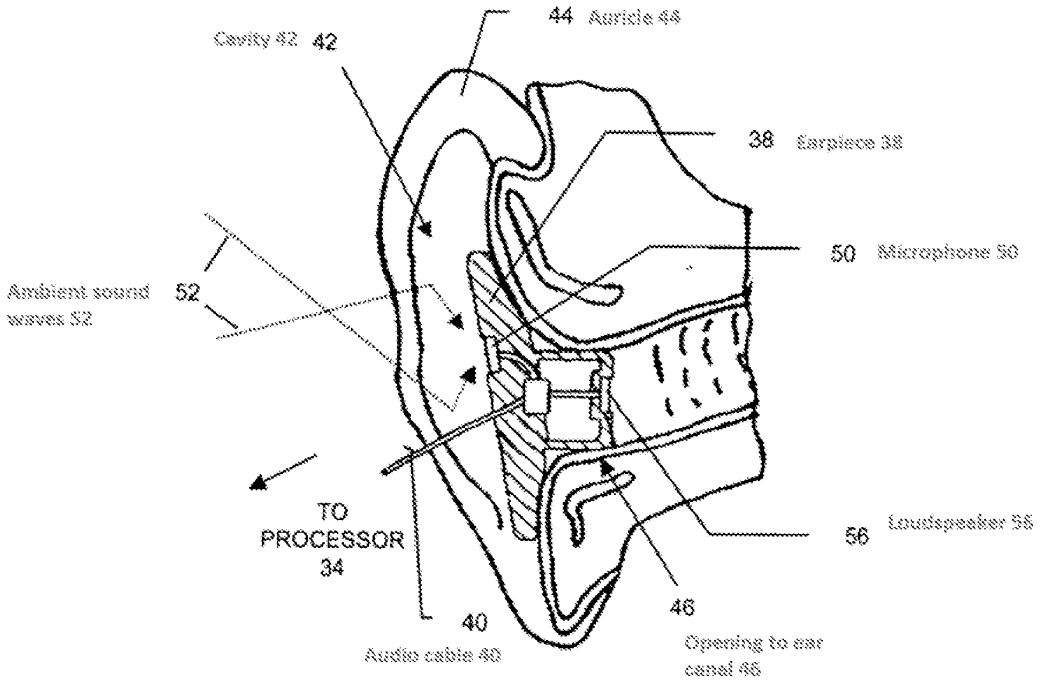


FIG. 2

Boersma, FIG. 2 (annotated)

87. The headset also includes a processor 34. (*Id.*, 6:13.) The processor 34 "functions to combine the audio signal received from the mobile phone 10 with the ambient sound signal provided by the respective microphones 50." (*Id.*, 6:58-62.) "The processor 34 then provides the combined signals to the loudspeaker 56 in each earpiece 38." (*Id.*, 6:62-63.)

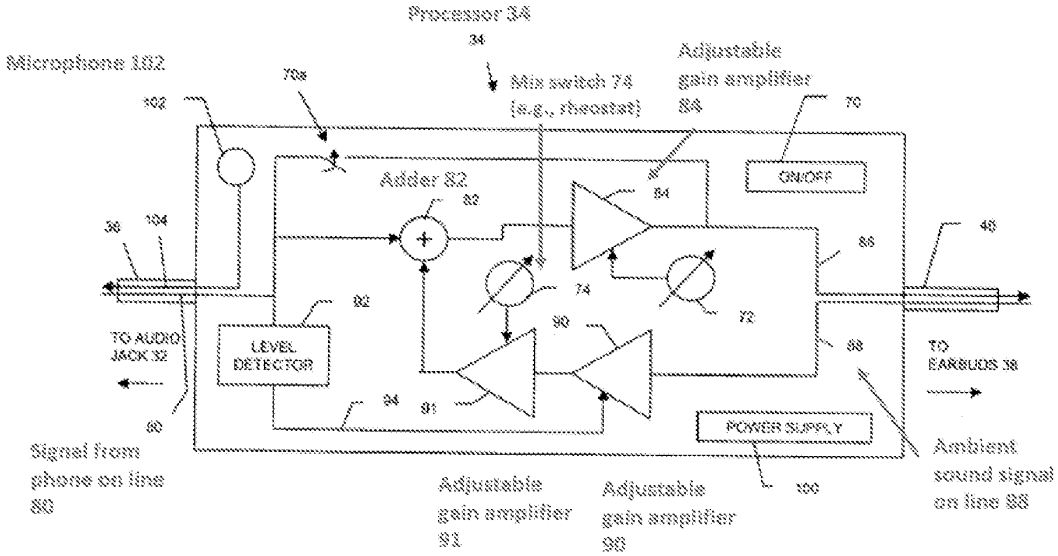


FIG. 4

Boersma, FIG. 4 (annotated)

88. A mix switch 74 controls the adjustable gain amplifier 91 and gives the user control over the "ratio by which the processor 34 combines the audio signal from the mobile telephone 10 and the ambient sound signal from the microphones 50." (*Id.*, 7:30-33, 8:15-18.) Thus, the user can choose whether the earpieces provides more or less ambient sound in relation to an incoming audio signal. (*Id.*, 7:33-36.) The user can (1) block most or all of the environmental sound, (2) retain ambient sound at a lower level, or (3) retain ambient sound at a higher level. (*Id.*, 7:36-46.)

89. A level detector 92 is used in conjunction with the adjustable gain amplifier 90 to automatically adjust the level of an ambient sound signal in

proportion to changes in the level of the received audio signal. (*Id.*, 8:33-42.) "In this manner, even if the user adjusts the audio output volume of the mobile phone 10 or other electronic equipment, the mix ratio of the audio signal and the ambient sound signal will remain substantially constant." (*Id.*, 8:42-46.)

90. Adder 82 "functions to combine the ambient sound signal with the audio signal from the mobile phone 10." (*Id.*, 8:19-21.) Volume switch 72 "functions to adjust the gain of the amplifier 84." (*Id.*, 8:27-28.) "Accordingly, the user may control the gain of the amplifier 84 which amplifies the combined audio and ambient sound signals. As a result, the user may control the overall volume of the sound reproduced by the headset." (*Id.*, 8:29-32.) The output of the amplifier 84 is provided to the earpieces 38 to be output through the speaker. (*Id.*, 8:6-7, 2:21-26, 2:65-3:5, 6:62-63.)

91. The processor also has a microphone 102 "for picking up the voice of the user during hands-free communication with the mobile phone." (*Id*, 8:63-67.) Boersma illustrates microphone 102 for picking up the user's voice as part of processor 34 in a wire/cable implementation (*see id.*, FIG. 4), but Boersma does not detail the location of microphone 102 when processor 34 is implemented within the housing of the earpiece or on the ear.

92. Boersma discloses implementing the processor 34 with "digital circuitry" using a "digital design" (*id.*, 7:54-61), implementing the processor 34

"within the housing of the earpiece(s) 38)" or "within a small housing designed to fit behind the ear of the user," (*id.*, 9:13-18), and implementing a Bluetooth version of processor 34 (*id.*, 9:1-8, FIG. 5). Boersma does not disclose the implementation details associated with implementing the processor 34 with a digital design within the housing of the earpieces 38.

IX. CLAIM INTERPRETATION

93. I have given all of the terms of the challenged claims their broadest reasonable interpretation consistent with the specification. I have taken into consideration the language of the claims and the specification.

X. SPECIFIC GROUNDS FOR CHALLENGE

A. Grounds Challenging Claims 1-11 of the '666 Patent

94. In the sections below, I explain in detail how claims 1-11 would have been rendered obvious. The table below summarizes the prior art references that I discuss below.

Reference	Filing date	Publication date	Type
US6647368B2 (Nemirovski)	7/2/2001	11/11/2003	§ 102(b)
US5251263A (Andrea)	05/22/1992	10/5/1993	§ 102(b)
EP1519625A2 (Victorian)	9/13/2004	3/30/2005	§ 102(b)
US20010046304A1 (Rast)	4/24/2001	11/29/2001	§ 102(b)
US20060153394A1 (Beasley)	1/10/2006	7/13/2006	§ 102(a)
EP1640972A1 (Dijkstra)	12/23/2005	03/29/2006	§ 102(b)
US20040047486A1 (Van Doorn)	9/6/2002	3/11/2004	§ 102(b)
US7903826B2 (Boersma)	3/8/2006	3/8/2011	§ 102(e)

95. The table below summarizes the grounds that in my opinion render the claims obvious.

Ground #	Basis	Claims
1	Obvious under § 103 in view of Nemirovski in combination with Rast	1-11
2	Obvious under § 103 in view of Nemirovski in combination with Rast and Dijkstra	1-11
3	Obvious under § 103 in view of Nemirovski in combination with Rast and Van Doorn	1-11
4	Obvious under § 103 in view of Victorian in combination with Dijkstra and Beasley	1-2, 8-9, 11
5	Obvious under § 103 in view of Victorian in combination with Dijkstra, Beasley, and Andrea	10
6	Obvious under § 103 in view of Victorian in combination with Van Doorn and Beasley	1-2, 8-9, 11
7	Obvious under § 103 in view of Victorian in combination with Van Doorn, Beasley, and Andrea	10
8	Obvious under § 103 in view of Dijkstra in combination with Boersma	1, 11
9	Obvious under § 103 in view of Dijkstra in combination with Boersma and Beasley	2, 8, 9
10	Obvious under § 103 in view of Dijkstra in combination with Boersma and Van Doorn	1, 11
11	Obvious under § 103 in view of Dijkstra in combination with Boersma, Van Doorn, and Beasley	2, 8, 9

B. Ground 1: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast

1. A person having ordinary skill in the art would have been motivated to combine Nemirovski and Rast

96. As explained further below, a person having ordinary skill in the art would have been motivated to combine the teachings of Nemirovski and Rast and would have had a reasonable expectation of success in doing so.

97. Nemirovski is directed to a sensor 12 that is inserted into the ear of a wearer and is configured to detect initiating actions, such as the speech of the person wearing sensor 12, and provide an output that corresponds to the detected action. (Nemirovski (PA-A), 9:3-28.) Nemirovski discloses that sensor 12 is disposed within a housing 150 that is made of "sound absorbing (or sound proofing) material and/or include[s] sound insulating material such as foam." (*Id.*, 12:22-25.) Nemirovski further explains that housing 150, together with the head of the user, "at least partially blocks externally generated sound waves" from reaching the internal microphone 34. (*Id.*, 13-3-7.) Nemirovski also describes implementing sensor 12 with noise-canceling functionality to further block environmental noise. (*Id.*, 9:46-49, 13:20-22, 14:19-21.) Nemirovski identifies as one of its advantages that it "detects less environmental, or ambient, noise" than the prior art. (*Id.*, 4:2-5.)

98. Nemirovski explains that, even without employing external microphone 54, using two sensors 12—one in each ear—improves performance "by

about six times due to the detection of more information." (*Id.*, 13:52-60.) Nemirovski notes, however, "that in certain circumstances, the use of a sensor 12 in both ears is discouraged (such as when using a cellular telephone while driving a car)." (*Id.*, 13:60-62.) A person having ordinary skill in the art would have understood that this is due to the acoustic isolation created by sensor 12 preventing environmental, or ambient, sound from reaching the user.

99. For example, a person having ordinary skill in the art would have understood that a driver may be unable to hear important ambient sound sufficiently. U.S. Patent Application Publication No. 2003/0035551A1 ("Light") explains that "noise cancellation headphones should not be used in certain situations because of the potential dangers of the headset wearer not hearing ambient noise adequately, such as in the case of driving an automobile." (Light (PA-S), ¶[0003].) As another example, U.S. Patent Application Publication No. 2006/0083388A1 ("Rothschild") explains "[r]ecently, digital noise reduction sound processing, and better headphone designs to isolate sound, have allowed audio headphone users to enhance their audio content listening experience… By example, if a user were driving a car, the user may want to only listen to music content in a sealed environment, meaning that only music could be heard to the exclusion of all other sounds and the experience could be further enhanced with digital noise reduction processing. ***However, prudency and in many cases the laws of the various individual states (in the United States) and***

other countries require that the user be able to hear certain outside noises while driving. For instance, the sound of a car horn or the sound of an emergency vehicle are two noises that the user would want to (and in many cases be required by law) to hear. Moreover, the user may also have a need to hear other occupants of his vehicle, but in this case, only when it is important for them to be heard." (Rothschild (PA-T), ¶[0005].)[3]

100. In view of these disclosures, a person having ordinary skill in the art would have been motivated to look for similar systems that discussed ways to modify Nemirovski's sensor 12 to enable use of a sensor 12 in both ears without sacrificing the wearer's ability to hear external sounds in circumstances where it would be desirable to do so (e.g., when using a cellular telephone while driving a car). In so looking, a person having ordinary skill in the art would have readily found Rast.

101. Rast also recognizes that using noise-canceling headsets in the circumstances described by Nemirovski may be dangerous. (Rast (PA-B), ¶[0008]

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[3] Here, I am relying on Light (PA-S) and Rothschild (PA-T) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

("It should be appreciated, however, that being isolated from every sound within the external acoustical environment can often be inconvenient, or even dangerous. A flight crew member may be unable to hear the siren of an emergency vehicle, a scream, a shout, or an associate loudly calling his/her name, while a party listening to music may miss an important phone call, be unable to hear the door bell, or miss their name being called out.").)

102. Rast discloses proposes a solution to that problem. *Rast* discloses a system and method for controllably changing the sound isolation properties of a headset in situations where the user would benefit from hearing ambient sound. (*Id.*, ¶¶[0008]-[0009].) In Rast's system, a signal generated by ambient sound microphone(s) mounted to a headset earpiece are compared to stored signals, and if a match is found, the headset can change modes, such as to a "hearthrough" mode in which ambient sounds are amplified and provided to the user while currently playing audio signals are attenuated. (*Id.*, ¶[0058] ("A 'hearthrough' button 86, provides for a manual activation of the hearthrough function wherein, external sounds picked up by the microphones are amplified and played within the earcups, preferably in concert with attenuating the audio amplitude of the programmed sound source.").)

103. A person having ordinary skill in the art would have found it obvious to modify Nemirovski's system to selectively provide ambient sounds to the user in

response to the detection of environmental audio matching specific criterion because it would have made Nemirovski's device safer, more flexible, and thus more marketable. For example, with Rast's solution implemented in Nemirovski, users could more often employ Nemirovski's superior two-sensor embodiment with less risk associated with being unable to effectively hear their surroundings.

104. A person having ordinary skill in the art would have been motivated to combine Nemirovski and Rast because the combination would have amounted to applying a known technique (Rast's sound detection technique to trigger the "hearthrough" mode) to a known device (Nemirovski's ear sensor system) ready for improvement to yield predictable results (an ear sensor system configured to selectively provide ambient sounds).

105. The combination would also have been obvious to a person having ordinary skill in the art because it amounted to the use of a known technique (Rast's technique for controlling sound isolation) to improve a similar device (Nemirovski's noise-canceling headset) in the same way (to make Nemirovski's headset both safer and more convenient for users).

106. The combination would have also amounted to using known work in one field of endeavor (Rast's controllable sound-isolation technique) for use in either the same field or a different one (Nemirovski's noise-cancelling voice control

apparatus) based on design incentives or other market forces (i.e., the ability to use Nemirovski's superior two-sensor embodiment under additional circumstances).

107. In the combined Nemirovski-Rast device, Rast's teachings related to implementing a "hearthrough" mode would be implemented in Nemirovski's system. More particularly, the signal generated by Nemirovski's external microphone would be, according to Rast's technique, compared by Nemirovski's processing circuitry 14" to stored sounds, and upon detecting a match, Nemirovski's processing circuitry 14" would generate and then amplify a signal of externally generated sounds and provide it to the wearer in concert with the audio signals previously being provided (which would preferably be attenuated to enable the external sounds to be more clearly heard by the wearer).

108. There is ample evidence that a person having ordinary skill in the art would have had a reasonable expectation of success in making the proposed combination. For example, Nemirovski and Rast share a very similar hardware and software configuration, that is, microphones and speakers operatively coupled to a processor to conduct signal processing. Moreover, Nemirovski already discloses using "feature extraction and analysis using a neural net trained to recognize respective signals/features of such signals." (Nemirovski (PA-A), 17:36-38, 18:27-32, 19:31-20:9.)

109. Other earpieces with a similar structural configuration to Nemirovski disclose a similar technique for playing ambient sound in the user's ear.

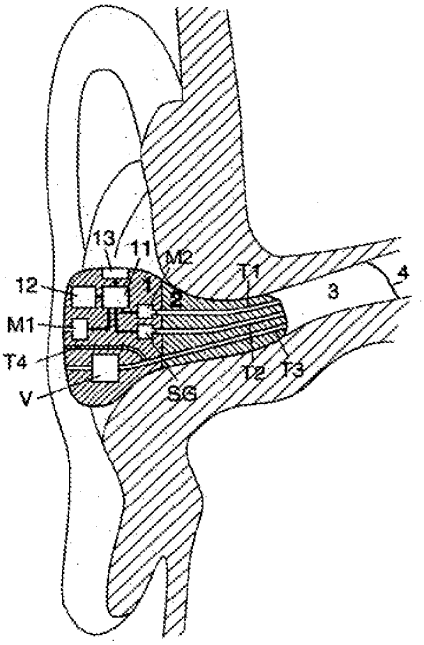


Fig. 1

Svean, FIG. 1

110. For example, Svean discloses an earpiece with an inner microphone M2 and outer microphone M1. (Svean (PA-I), 5:29-51.) Svean explains that using earplugs may cause the user to feel his own voice being distorted, which reduces the comfort of wearing hearing protectors and may cause the user to raise his/her voice. (*Id.*, 2:14-21.) To solve that problem, Svean discloses "filtering and mixing in the user's own voice picked up by either the outer or the inner microphone at one ear"

and reproducing the signal at the loudspeaker in the same ear or other ear, applying feedback cancellation if needed. (*Id.*, 2:22-29.)[4]

111. I understand that Patent Owner may argue that implementing Rast's technique in Nemirovski's system would have introduced problems with feedback between Nemirovski's speaker 38 and internal microphone 34, but a person having ordinary skill in the art would have been aware of and able to implement solutions for any potential feedback issues. For example, U.S. Patent No. 6,671,379 ("Nemirovski '379") discloses "a circuit coupled to receive the microphone signal and the speaker signal and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker to produce a corrected microphone signal." (Nemirovski '379 (PA-U), Abstract.) Similarly, the Victorian reference also discloses a potential solution to potential feedback, i.e., gating rules. (Victorian (PA-E), ¶[0031] ("The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing

[4] Here, I am relying on Svean (PA-I) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 1.

caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440.").) Svean also explains that reproducing ambient sound in the same ear with an inner microphone is possible as long as "feedback cancellation" is applied. (Svean (PA-I), 2:25-27.)[5]

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

112. To the extent the preamble is limiting, Nemirovski discloses or suggests an earphone. Nemirovski discloses "a system 52 for detecting initiating actions and providing an output which corresponds to the detected action." (Nemirovski (PA-A), 9:3-5.) As explained with respect to FIG. 1E, system 52 includes a sensor 12 having an internal microphone 34 and an external microphone 54 each disposed on a housing 150 "to be inserted at least part way into the ear of a person." (*Id.*, 9:5-28.) FIG. 2B, reproduced and annotated below, "illustrates sensor 12 inserted at least partially into the ear 100 of a person" (*id.*, 11:63-67):

[5] Here, I am relying on Nemirovski '379 (PA-U), Victorian (PA-E), and Svean (PA-I) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

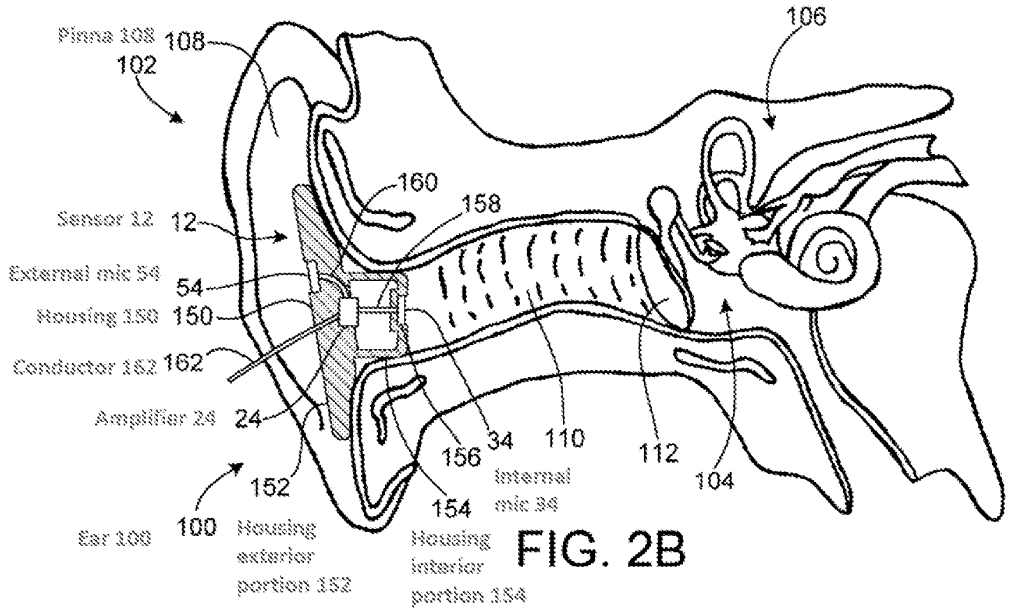


FIG. 2B

Nemirovski, FIG. 2B (annotated)

113. Nemirovski discloses that, "[a]lthough not illustrated in FIG. 2B, a speaker 38 (FIG. 1C) can be placed on or within the housing 150 of the sensor 12 to transmit sounds to the individual." (*Id.*, 14:1-3.) Nemirovski also explains that the exterior portion of housing 150 can take on a number of configurations including one that "resembles the housing of a hearing aid" and that the exterior portion 152 can also take on various configurations, including to "engage the ear 100 to cover the whole pinna 108 (e.g., similar to that of a common headphone assembly." (*Id.*, 12:3-16.)

114. A person having ordinary skill in the art would have understood Nemirovski's system 52 including sensor 12 to comprise "[a]n earphone" at least

because it is configured to be inserted at least part way into the ear of a person and because it includes a speaker 38 configured to transmit sounds to the individual. Nemirovski's sensor 12 is also of the same general construction as the "earphone" (or "earpiece") described in the '666 patent. (PAT-A, FIG. 1, 4:52-65.)

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;"

1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;"

115. Nemirovski discloses or suggests these features.

116. **Addressing element 1[b] first**, Nemirovski discloses system 52 including an "internal microphone 34" that is disposed on housing 150 and is inserted "into a cavity defined by an external portion of the ear." (Nemirovski (PA-A), 9:10-15; *see also id.*, FIGs. 1E & 2B.) Nemirovski explains that "internal microphone 34 detects air pressure changes located within the ear." (*Id.*, 9:15-17.) Nemirovski's FIG. 2B also depicts internal microphone 34 as closer to ear canal 110 than external microphone 54. (*Id.*, FIG. 2B.)

117. Nemirovski explains that the disclosure uses the term "changes in air pressure" in its broadest sense to characterize "the parameter ***being measured***." (*Id.*, 11:20-23.) Nemirovski further provides that "[c]hanges in air pressure may alternatively be characterized as sound waves." (*Id.*, 11:23-24.) Thus, a person having ordinary skill in the art would have understood Nemirovski's internal microphone 34 detecting air pressure changes located within the ear to comprise a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the external microphone 54.

118. Nemirovski's disclosure of an internal microphone that points towards the user's ear canal and detects sounds within the ear is also consistent with how the '666 patent describes the "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone." (PAT-A, 4:66-5:3, 5:22-28.)

119. Regarding the second microphone port, notably, the '666 patent does not define the term "microphone port." Indeed, the word "port" does not appear in the specification of the '666 patent. The "first microphone port" and "second microphone port" limitations were added by amendment during prosecution, and the applicant argued that Figures 1 and 2 supported the amendment. (PAT-B, Aug. 4, 2021 Response to Office Action.) The applicant explained that the amendment was

to "further distinguish the first and second microphone orientations (i.e., the associated microphone port through which measurements are taken)." (*Id.*, 6.)

120. A person having ordinary skill in the art would have understood the term "microphone port" or "port" to refer to one or more related structures. For example, the prior art uses the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted. U.S. Patent No. 7,983,433 ("Nemirovski '433") (PA-J) describes a microphone 16 disposed within earset 14 "**microphone port 28**" and a "microphone port 50" which are openings in the housing of the earset. (Nemirovski '433 (PA-J), 7:13-17, 9:22-25.) Nemirovski '433 also describes "input port 32a" and "input port 32b." (*Id.*, 7:18-19.)

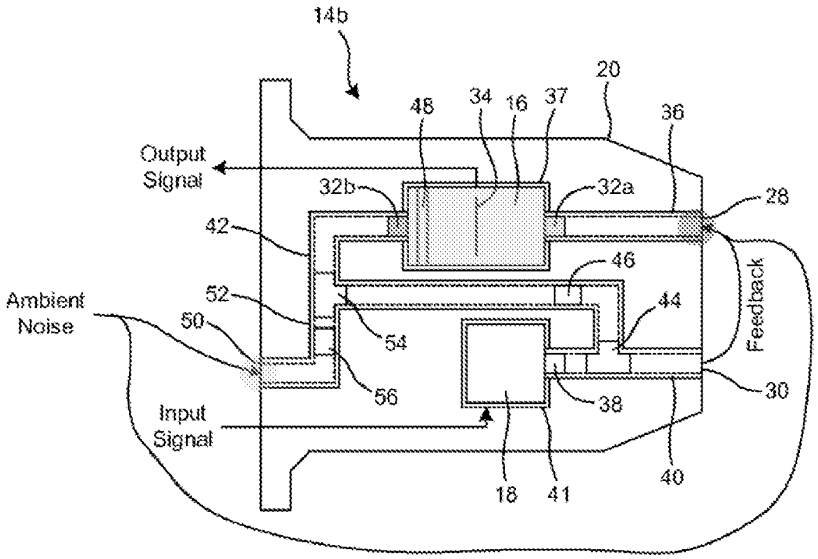


FIG. 6

Nemirovski '433, FIG. 6

121. U.S. Patent No. 7,502,484 ("Ngia") (PA-K) similarly describes a channel in an ear piece through which sound is conveyed to a microphone and referring to the opening of the ear piece at which the channel terminates as a "port 38." (Ngia (PA-K), 9:18-28.)

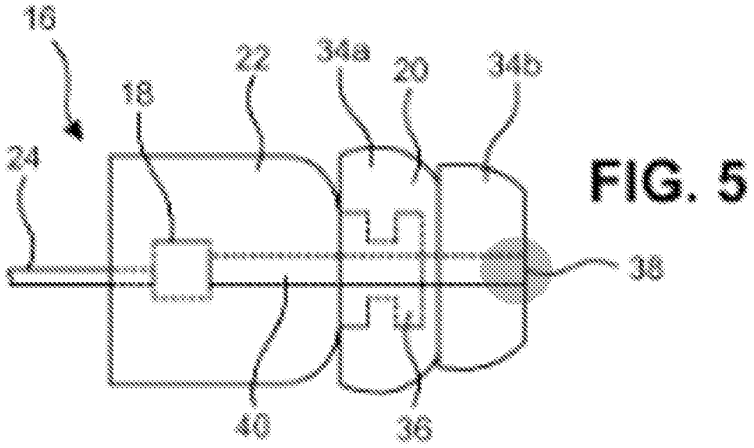


FIG. 5

Ngia, FIG. 5 (annotated)

122. U.S. Patent No. 8,351,634 ("Khenkin") (PA-L) describes a mobile phone housing 104 that "defines a microphone port 110, i.e., a hole through which sound enters the housing 104." (Khenkin (PA-L), 6:51-52.) Khenkin further discloses that "MEMS microphone package 210 should be positioned such that the

aperture 218 is close to, and oriented toward, the microphone port 110." (*Id.*, 6:57-59.)[6]

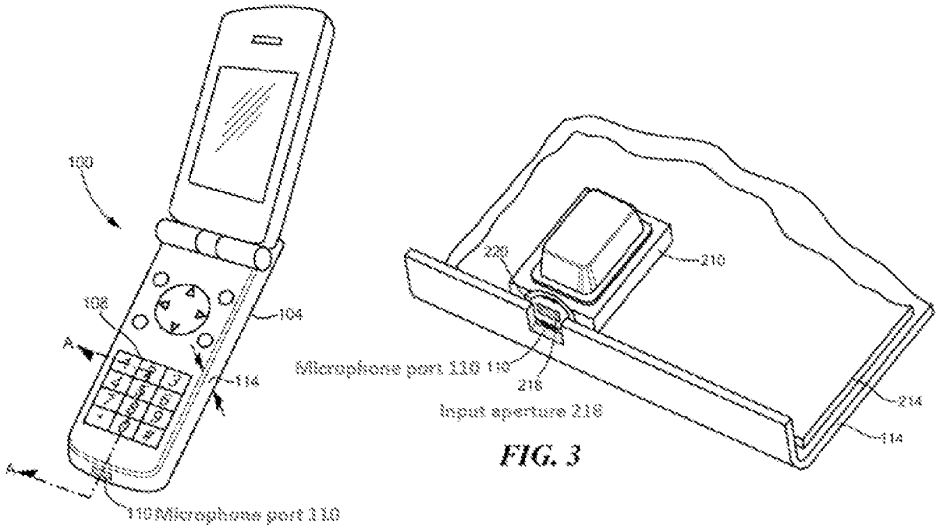


FIG. 1 FIG. 3

Khenkin, FIGs. 1, 3

123. The prior art also uses the term "port" to refer to an opening in a microphone package itself. U.S. Patent Application Publication No. 2004/0047486 ("Van Doorn") (PA-D) discloses a miniature microphone 110 with a "cup-shaped

[6] Here, I am relying on Nemirovski '433 (PA-J), Ngia (PA-K), and Khenkin (PA-L) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

sound input plate 50" with "an outer wall 52" and a "sound port 56, which is an opening in the outer wall 52." (Van Doorn (PA-D), ¶[0020].) "In operation, the sound inlet plate 50 receives sound through the sound inlet port 56 in its exterior wall 52. The sound propagates through a passageway 135 that is defined by the interior recess 54 of the plate 50 and the exterior wall of the housing 112 adjacent to the aperture 128. Eventually, the sound is transmitted through the aperture 128 and acts upon the transducing assembly 121." (*Id.*, ¶0022].)

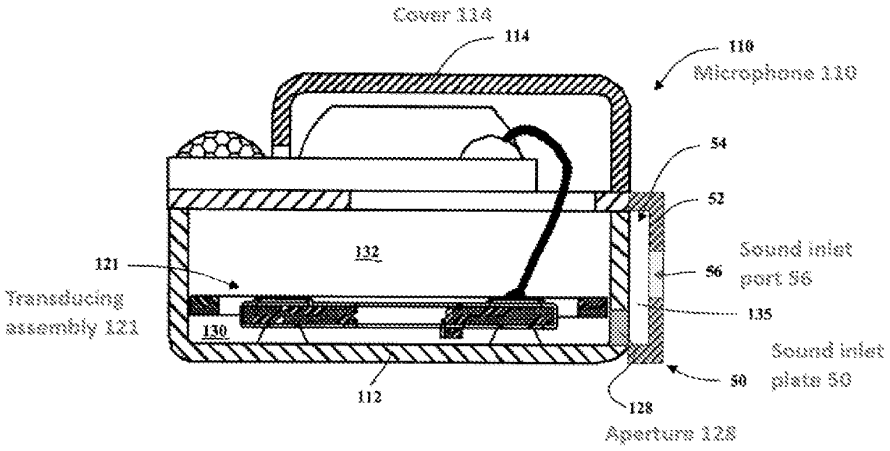


FIG. 3

Van Doorn, FIG. 3 (annotated)

124. U.S. Patent 7,953,241 ("Jorgensen") (PA-M), describes a microphone housing 1 with a "sound inlet port 5" as "positioned on top of the microphone on a surface part 15 of the microphone." (Jorgensen (PA-M), 6:28-45.)

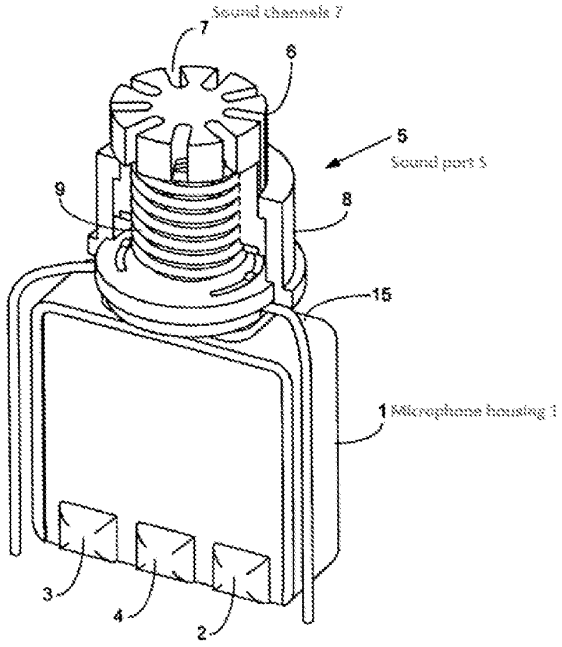


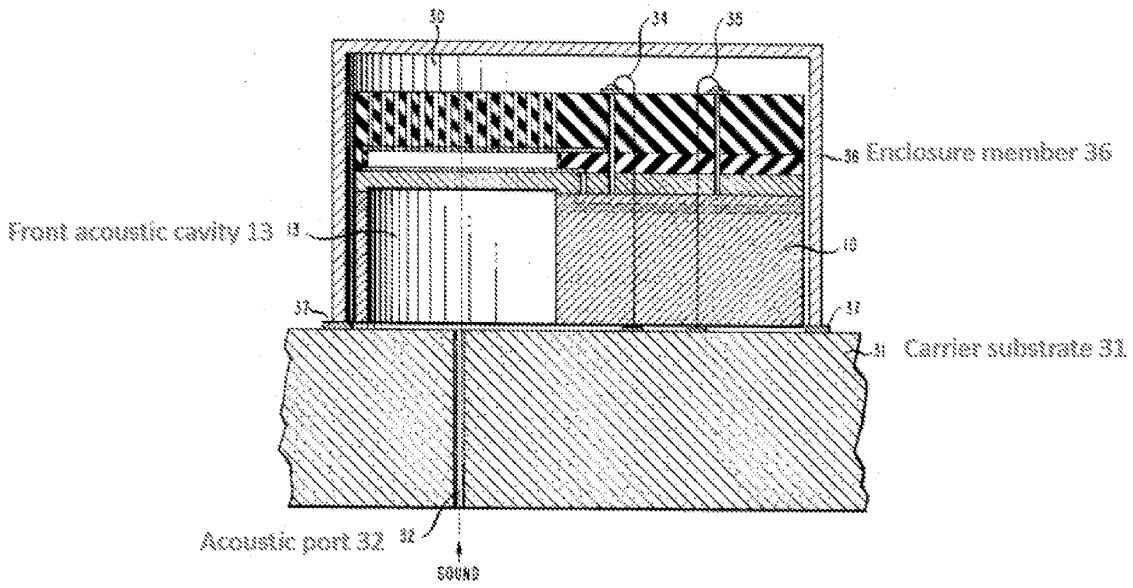
Fig. 1

Jorgensen, FIG. 1 (annotated)

125. U.S. Patent No. 4,533,795 ("Baumhauer") (PA-N) describes an "acoustic port" in a capacitive microphone as an element adjacent the front acoustic cavity that produces a desired frequency response. (Baumhauer (PA-N), 9:18-27.)[7]

[7] Here, I am relying on Van Doorn (PA-D), Jorgensen (PA-M), and Baumhauer (PA-N) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

FIG. 6



126. Thus, a person having ordinary skill in the art would have understood that, under the broadest reasonable interpretation of claim 1, any opening associated with a microphone "through which measurements are taken" qualifies as a "microphone port" within the meaning of the '666 patent. (*See* PAT-B, Aug. 4, 2021 Response to Office Action, at 6 (applicant characterizing "microphone port").) And under the broadest reasonable interpretation, Nemirovski discloses such a port.

127. Nemirovski discloses an embodiment in which "[internal] microphone 34 is disposed within a hollow cavity defined by the housing 150 and an opening in a wall of the internal portion 154 of the housing 150 is provided to establish fluid communication of the air inside the housing 150 and the air inside the ear canal 110."

(Nemirovski (PA-A), 12:64-13:2.) A person having ordinary skill in the art would have understood that this opening in the wall of the internal portion 154 to establish fluid communication of the air inside the housing 150 and the air inside the ear canal 110 would comprise a microphone port (under the broadest reasonable interpretation of that term) for internal microphone 34 that is configured to face toward the user when the earphone is inserted.

128. Accordingly, Nemirovski discloses or suggests internal microphone 34 ("a second microphone") detecting air pressure changes located within the ear ("configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone"), wherein the internal microphone 34 is disposed within a hollow cavity defined by the housing 150 and an opening in a wall of the internal portion 154 of the housing 150 ("wherein the second microphone has a second microphone port") is provided to establish fluid communication of the air inside the housing 150 and the air inside the ear canal 110 to allow internal microphone 34 to detect sounds within the ear canal ("that is configured to face toward the user when the earphone is inserted").

129. **Turning next to element 1[a]**, Nemirovski describes system 52 including an external microphone that "detects air pressure changes, or sounds, originating outside the user's ear." (*Id.*, 4:25-27.) Specifically, Nemirovski discloses an "external microphone 54" which "generally points away from the person wearing

the sensor 12" and "detects air pressure changes located outside the ear," such as air pressure changes that "are the result of externally generated noise (other people's speech, environmental noise, etc.)" and the speech of the person wearing sensor 12. (*Id.*, 9:20-28; *see also id.*, FIG. 1E.) Nemirovski also equates "environmental" noise with "ambient" noise. (*Id.*, 4:1-5.)

130. For the same reasons as discussed with respect to element 1[b], a person having ordinary skill in the art would have understood Nemirovski's external microphone 54 detecting air pressure changes located outside the ear resulting from externally generated sounds to comprise a first microphone configured to measure sound waves resulting from ambient sounds. Nemirovski's disclosure of an external microphone that points away from the user and detects sounds outside the ear is also consistent with how the '666 patent describes the "first microphone configured to measure an ambient acoustic environment." (PAT-A, 4:66-5:3, 5:28-34 6:38-46.)

131. Regarding the first microphone port, a person having ordinary skill in the art would have recognized that Nemirovski's disclosure to dispose the internal microphone 34 within a hollow cavity defined by the interior portion 154 of the housing 150 to be equally applicable to the external microphone 54 and the exterior portion 152 of the housing 150. That is, a person having ordinary skill in the art would have recognized the benefit of incorporating a hollow cavity within the

exterior portion 152 of the housing 150 and disposing the external microphone 54 within such a hollow cavity.

132. For example, a person having ordinary skill in the art would have been motivated to adopt such a structure because it would provide additional external physical protection for microphone 54 and attach the external microphone more securely to or within the external portion 152 of the housing 150. In doing so, a person having ordinary skill in the art would also have found it obvious to provide an opening in a wall of the exterior portion 152 of the housing 150 for the same reason that Nemirovski discloses with the respect to the to the internal microphone, i.e., to establish fluid communication of the air inside the hollow cavity within the exterior portion of the housing 150 and the air outside the user's ear.

133. Accordingly, Nemirovski discloses or suggests external microphone 54 ("a first microphone") detecting air pressure changes located outside the ear resulting from externally generated sound ("configured to measure an ambient acoustic environment"), wherein the external microphone 54 is disposed within a hollow cavity defined by the housing 150 and an opening in a wall of the exterior portion 152 of the housing 150 ("wherein the first microphone has a first microphone port") is provided to establish fluid communication of the air inside the housing 150 and the air outside the user's ear to allow external microphone 54 to detect sounds

outside the ear ("that is configured to face away from a user when the earphone is inserted").

c. 1[c] "a speaker configured to play an audio signal;"

134. Nemirovski discloses or suggests this feature. Nemirovski discloses sensor 12 including a speaker 38. For example, Nemirovski provides that, "[a]lthough not illustrated in FIG. 2B, a speaker 38 (FIG. 1C) can be placed on or within the housing 150 of the sensor 12 to transmit sounds to the individual. For example, when using the sensor 150 as a microphone for a communications device 36, the sensor 12 can also be used to broadcast received sounds to the user." (Nemirovski (PA-A), 14:1-6; *see also id.*, 8:6-9.) Accordingly, Nemirovski discloses or suggests a speaker 38 ("a speaker") used to transmit sounds ("configured to play an audio signal") to a user.

d. 1[d] "a memory that stores instructions; and"

1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:"

135. Nemirovski in combination with Rast discloses or suggests these features. Nemirovski discloses system 52 including "processing circuitry 14"."

(Nemirovski (PA-A), 9:29-40.) FIG. 1E, which depicts system 52 and includes processing circuitry 14" is reproduced below:

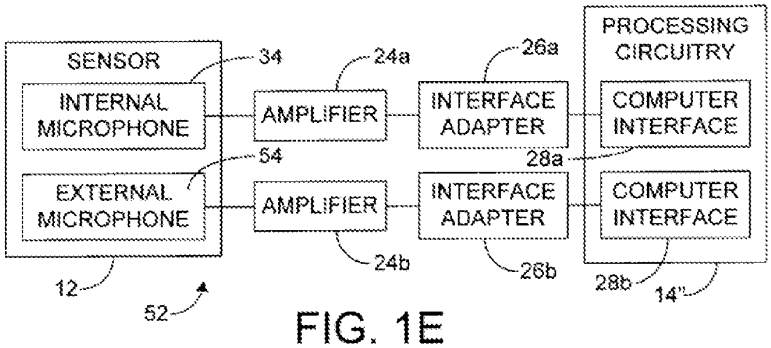


FIG. 1E

Nemirovski, FIG. 1E

136. A person having ordinary skill in the art would have found it obvious to implement Nemirovski's processing circuitry 14" as a processor or signal processing unit. For example, with respect to FIGs. 1A and 1B, Nemirovski explains that processing circuitry 14 and processing circuitry 14', respectively, include a processor 18 (or processor 18') programmed to perform various signal processing functions and execute a voice recognition software program. (*Id.*, 6:46-51, 7:35-36.) Nemirovski also notes that "like reference numerals will refer to like elements throughout." (*Id.*, 3:42-45.) Further, Nemirovski explains that "processing circuitry 14" or another device [may be] used to receive the electrical signals respectively produced by the microphones 34 and 54 in response to sensed air pressure changes." (*Id.*, 9:29-32.) As such, a person having ordinary skill in the

art would have found it obvious for processing circuitry 14" to similarly include, or be implemented as, a processor or signal processing unit, such as Nemirovski's processor 18 (or processor 18'), to perform various signal processing operations to detect the occurrence of initiating actions from the signals produced by internal microphone 34 and external microphone 54.

137. Nemirovski is silent as to the location of the processing circuitry 14". In the combined Nemirovski-Rast device, however, it would have been obvious to a person having ordinary skill in the art to have implemented processing circuitry 14" as a processor within housing 150 (along with internal microphone 34, external microphone 54, and speaker 38) in view of Rast and the knowledge of a person having ordinary skill in the art.

138. Rast discloses implementing its signal processing unit as a digital signal processor "within" its headset. (PA-B, ¶¶ [0022], [0053]; FIG. 3.) Rast describes, and depicts in FIG. 3, DSP 62 as connected, or coupled, to microphones 58*a*, 58*b* and earpiece speakers 56*a*, 56*b*. (*Id.*, ¶[0053].) Rast also describes implementing its functionality "in various mechanical configurations, such as mounted as earpieces within a helmet, single sided headset, earbuds, and so forth." (*Id.*, ¶[0013].)

139. For the reasons discussed in Section X.B.1, a person having ordinary skill in the art would have been motivated to modify Nemirovski's processing circuitry 14" (implemented as a processor or signal processing unit) to selectively

provide ambient sounds to the user in response to the detection of environmental audio matching specific criterion as performed by Rast's DSP 62. A person having ordinary skill in the art would also have been motivated to include processing circuitry 14" and a corresponding memory within Nemirovski's housing 150 as taught by Rast at least because there were well-known benefits associated with doing so.

140. For example, there was a recognized preference in the art for devices that were hands-free. U.S. Patent No. 8,014,553 ("Radivojevic") (PA-O) discloses how "[o]rdinary handsfree modules and hearing aids are very useful modules/gadgets. On the other hand, people do not like carrying separate modules while commuting and travelling. At the present, high-level integration technologies can provide a single small device which might be hosted at an external electronic device (e.g., mother phone)." (Radivojevic (PA-O), 1:58-63.)[8] Additionally, a person having ordinary skill in the art would have understood that implementing Rast's functionality within the headset would have provided the earpiece with

[8] Here, I am relying on Radivojevic (PA-O) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 1.

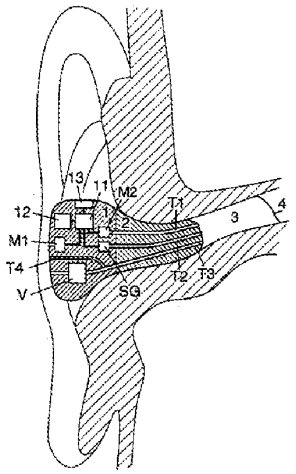
interoperability advantages because the earpiece would not be reliant on processing power derived from a separate device. More particularly, Boersma recognizes that attempts have been made "to maintain ambient sound in addition to the audio reproduced by the headsets," but that such attempts were "limited with respect to their ability to be used with a variety of types of electronic equipment." (Boersma (PA-H), 1:57-67.)[9] As a solution to this problem, Boersma taught that giving the headset its own processing capabilities would make it "suitable for use with a variety of types of electronic equipment" and that the headset would not have to "rely on processing circuitry and/or controls within the electronic equipment 10 to provide a user with the ability to listen to reproduced audio yet still be able to hear ambient sound." (Boersma (PA-H), 5:51-6:3.)

141. Rast itself also recognizes that the various operations discussed herein and listed in its Table 1—which include amplifying and providing external sounds to the wearer if certain sounds are detected while also attenuating an audio source

[9] Here, I am relying on Boersma (PA-H) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 1.

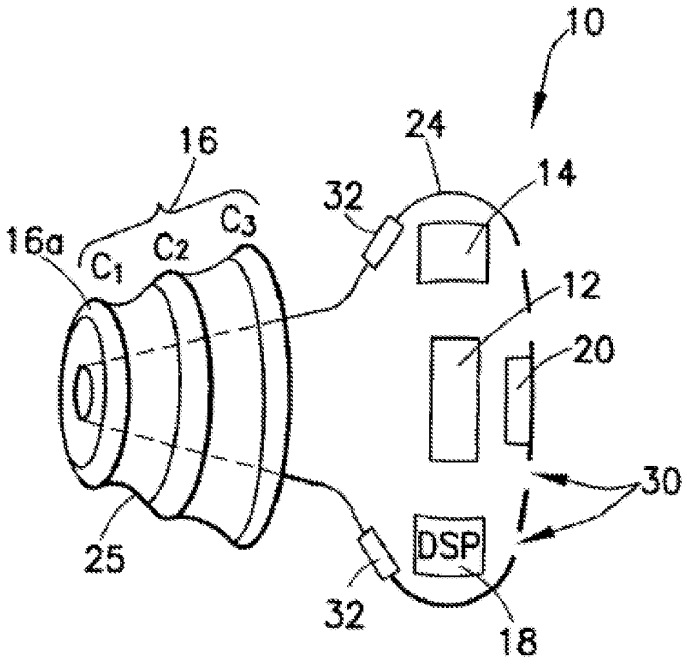
being listened to—are ***preferably implemented within the headset***." (Rast (PA-B), ¶[0059].)

142. Moreover, a person having ordinary skill in the art would have had a reasonable expectation of success, as the prior art provides multiple examples of devices with earpieces that had integrated processor and memory, including with a variety of form factors. For example, U.S. Patent No. 6,728,385 ("Kvaløy") (PA-P) discloses an apparatus "where the voice detection and discrimination is included in an earplug based hearing protective voice communication terminal," as shown in FIG. 1 reproduced below. (Kvaløy (PA-P), 3:38-41.) Furthermore, Kvaløy discloses that such apparatus includes a processing unit (including a digital signal processor) connected to memory storage means. (*Id.*, 5:1-6:49.)



Kvaløy, FIG. 1.

143. Similarly, Radivojevic discloses an "ear-device" that "when detached and placed into the human ear in a handsfree tissue mode (HTM) [acts] as an autonomous miniaturized device" comprising of "an electronic processing module (e.g., digital signal processor) for supporting functionalities of the tissue conducting sensor, the microphone and the speaker" and a memory module can be included in the ear-device if data storage is required." (Radivojevic (PA-O), 6:36-45, 8:32-50.)[10]



[10] Here, I am relying on Kvaløy (PA-P) and Radivojevic (PA-O) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

Radivojevic, FIG. 4a.

144. Notably, the claims have no limitations related to form-factor (e.g., requiring that the earphone fit entirely within the ear). Nor do the claims have any limitations related to battery power, processing speed, etc. And the '666 patent specification itself takes form factor for granted, without any hint that it would have made a difference. (PAT-A, 4:52-65, 6:21-26.) For example, the '666 patent discloses various ear piece design forms where "[t]he earpiece 100 can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, open - fit device, or any other suitable earpiece type," without any discussion. (PAT-A, 4:52-65.) Furthermore the '666 patent discloses that the earpiece may even comprise of multiple distinct devices, again without any discussion of how any different form factor would affect implementation of the disclosed functionality. (PAT-A, 6:21-26 ("The earpiece 100 can further represent a single operational device or a family of devices configured in a master-slave arrangement, for example, a mobile device and an earpiece.").)

145. Thus, any expected tradeoffs associated with a smaller (or larger) form factor would not have dissuaded a person having ordinary skill in the art from making the modifications necessary to achieve the claimed invention.

146. Accordingly, in the combined Nemirovski-Rast device, it would have been obvious to a person having ordinary skill in the art to implement Nemirovski's

processing circuitry 14" as a processor, or signal processing unit, that is located within housing 150 and configured to execute instructions to perform various operations, including amplifying and providing external sounds to the wearer if certain sounds are detected while also attenuating an audio source being listened to, as described by Rast.

147. Nemirovski discloses that each of internal microphone 34 and external microphone 54 are connected to processing circuitry 14". (Nemirovski (PA-A), 9:29-41 ("Each microphone 34 and 54 is connected to the processing circuitry 14" using separate and distinct inputs, or channels.").) As explained in Section X.B.2.a, Nemirovski also describes a speaker 38 included within the housing 150 of sensor 12 for transmitting sounds to the user received from a communications device 36. For example, Nemirovski explains that "sensor 12 and communication device 34 can be used as a bidirectional communication apparatus." (*Id.*, 8:6-9.)

148. Although Nemirovski does not explicitly disclose processing circuitry 14" coupled to speaker 38, it would have been obvious to a person having ordinary skill in the art to couple speaker 38 and processing circuitry 14" to enable sounds received from communications device 36 to pass through processing circuitry 14" for processing before being passed to speaker 38 (i.e., when implementing Nemirovski's processing circuitry 14" as a processor, or signal processing unit, located within housing 150, as taught by Rast. For example, Rast explains that, when

utilizing the headset in a digital embodiment with a digital input source, audio input would be routed through the DSP/microcontroller on the way to the speakers, enabling power to be easily derived from the digital signal or bias level embedded within the signal. (Rast (PA-B), ¶[0063].)

149. A person having ordinary skill in the art would have understood that coupling Nemirovski's speaker 38 and processing circuitry 14" would similarly allow processing circuitry 14" to derive power from signals received from, for example, a cell phone or other digital input source en route to speaker 38. As such, a person having ordinary skill in the art would have found it obvious to couple Nemirovski's processing circuitry 14" to each of internal microphone 34, external microphone 54, and speaker 38.

150. A person having ordinary skill in the art would also have understood that Nemirovski's device would include a memory to store the instructions executed by Nemirovski's processing circuitry 14", especially when processing circuitry 14" is implemented as a processor or a signal processing unit. For example, Nemirovski explains that each functional block in the figures, including processing circuitry 14" depicted in FIG. 1E, "may represent a module, segment, or portion of software code that contains one or more executable instructions to implement the specified logical function(s)." (Nemirovski (PA-A), 10:18-25.)

151. A person having ordinary skill in the art would have understood that a memory would be required to store the instructions to be executed by processing circuitry 14" to implement the disclosed functions. At the very least, it would have been obvious in view of the knowledge of a person having ordinary skill in the art for Nemirovski's device to include a memory to store the instructions executed by Nemirovski's processing circuitry 14" because this was a well-known and common way of implementing a processing unit-based system to perform operations. For example, Svean describes how "[t]he processing unit E3 is connected to memory elements" which are "used for storing computer programs, filter coefficients, analysis data and other relevant data," all of which are necessary for the processing unit to operate properly. (Svean (PA-I), 6:45-58.) Similarly, Dijkstra discloses that "[i]t is evident to the skilled person that the earplug 10 includes additional electrical components, such as … program and data memory components for the audio signal processing unit 24." (Dijkstra (PA-C), ¶[0029].)[11] Moreover, when implementing Nemirovski's processing circuitry 14" as a processor, or signal processing unit,

[11] Here, I am relying on Svean (PA-I) and Dijkstra (PA-C) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; they are not part of the combination in Ground 1.

located within housing 150, it would have been obvious to a person having ordinary skill in the art to include an internal program store (ROM) and random access memory (RAM) to store instructions and a set of audio selection criterion, respectively, as taught by Rast. (Rast (PA-B), ¶[0053])

152. Accordingly, the Nemirovski-Rast device discloses or suggests processing circuitry 14" and a memory that stores instructions executed by the processing circuitry 14" implemented as a processor or signal processing unit ("a memory that stores instructions; and a processor that is configured to execute the instructions to perform operations"), where the processor or signal processing unit is connected to internal microphone 34, external microphone 54, and speaker 38 ("wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor"). The combination of Nemirovski and Rast discloses or suggests the processor or signal processing unit performing the claimed operations, as discussed below.

e. 1[f] "receiving a first microphone signal from the first microphone;"

1[g] "receiving a second microphone signal from the second microphone;"

153. Nemirovski discloses or suggests these features. Nemirovski discloses that changes in air pressure detected by internal microphone 34 and external microphone 54 are "converted by the microphones 34 and 54 into respective

electrical signals, herein referred to as an internal signal and an external signal." (Nemirovski (PA-A), 15:18-20.) Nemirovski further describes processing circuitry 14" receiving "the electrical signals respectively produced by microphones 34 and 54 in response to sensed air pressure changes." (*Id.*, 9:29-32.) Accordingly, Nemirovski discloses or suggests processing circuitry 14" implemented as a processor or signal processing unit receiving an external signal from external microphone 54 ("receiving a first microphone signal from the first microphone") and receiving an internal signal from internal microphone 34 ("receiving a second microphone signal from the second microphone").

f. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;"

154. Nemirovski in combination with Rast discloses or suggests this feature. As discussed in Section X.B.2.b, Nemirovski discloses external microphone 54 detecting air pressure changes resulting from externally generated ambient sound. Nemirovski further discloses adapting system 52 to cancel externally generated noise from specific sounds detected by the external microphone. (Nemirovski (PA-A), 15:48-16:7.)

155. For example, Nemirovski explains that, "in most applications, it will be advantageous to isolate the component of the internal signal generated by the initiating action and, if desired, eliminate to the fullest extent possible any externally

generated component of the internal signal." (*Id.*, 15:53-57.) To do so, Nemirovski describes "cancel[ing] noise (or all sounds) detected by the external microphone 54 from the detections of the internal microphone 34. More specifically, the externally generated component of internal signal is filtered from or mathematically removed from the internal signal." (*Id.*, 15:60-65.) Thus, a person having ordinary skill in the art would have understood Nemirovski to disclose or suggest "filter[ing]" or "mathematically remov[ing]" the noise detected by external microphone 54 from the internal signal to isolate the initiating action.

156. Nemirovski further discloses performing "[s]imilar processing" on the external signal "to derive a signal of externally generated sounds, such as a to isolate another person's speech from noise and from speech of a person wearing the sensor 12." (*Id.*, 16:1-5.) To perform similar processing on the external signal (for example, to isolate an external sound or sounds from noise and speech of the wearer), a person having ordinary skill in the art would have understood Nemirovski to disclose or suggest filtering or mathematically removing the noise detected by internal microphone 34 from the external signal from external microphone 54 to derive a signal of externally generated sounds.

157. The '666 patent does not describe the step of "generating an ambient sound signal." The specification only describes the "ambient sound signal" as a signal received or captured from the ambient sound microphone. For example,

the '666 patent details with reference to Figure 4 a method for an "Acute-Sound Pass Through System (ACPTS)." ('666 Patent (PAT-A), 8:59-61). It describes that "[a]t step 402, the earpiece 100 ***captures ambient sound signals*** from the ASM 111." (*Id.*, 9:3-4.) FIG. 4, in turn, shows step 402 as "INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES."

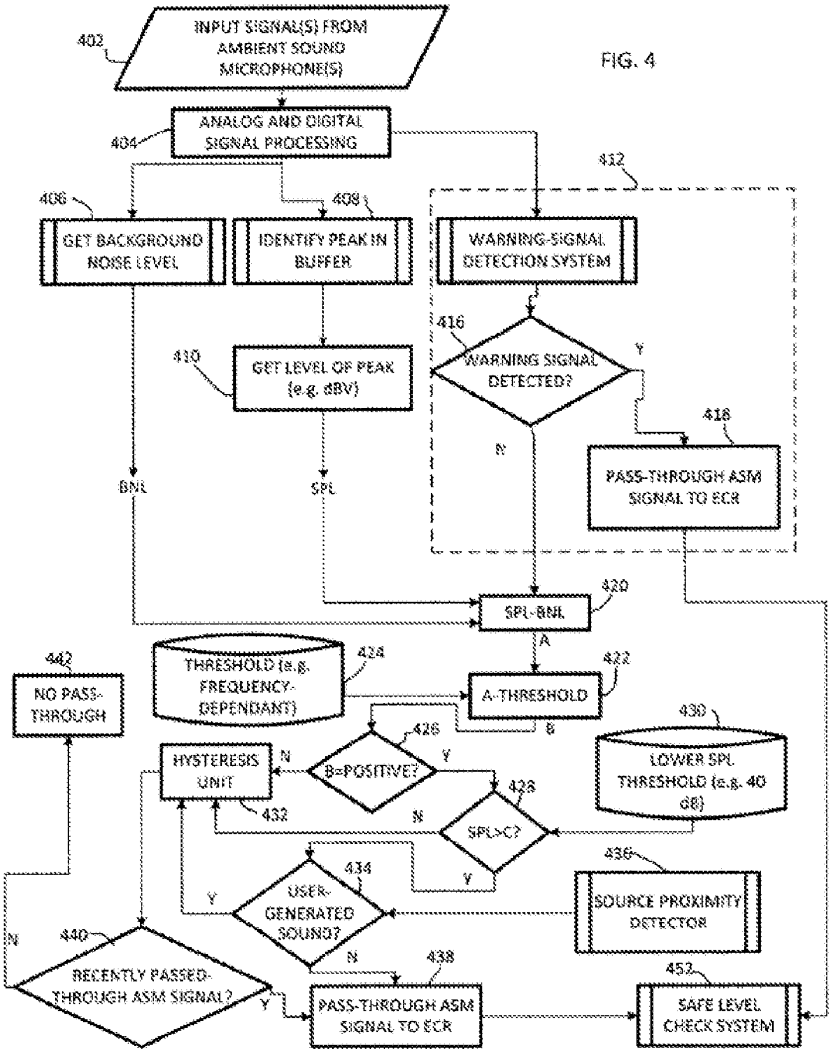


FIG. 4

'666 Patent, FIG. 4

158. Further, in the discussion of the method of Figure 4, the '666 patent equates the "ambient sound signal" with the "ASM input signal." (*Id.*, 9:42-45 ("Briefly, if a transient, high-level sound (or acute sound) is detected in ***the ambient sound signal (ASM input signal)***, then it is converted to a level, and its magnitude compared with the BNL is calculated.").)

159. As another example, in describing the method of Figure 5 for "Source Proximity Detection (SPD)," the '666 patent states that, "The SPD method 500 uses as its inputs the external ambient sound signals from left and right electro-acoustic earpiece 100 assemblies (e.g., a headphone). In some embodiments the SPD method 500 employs Ear Canal Microphone (ECM) signals from left and right earpiece 100 assemblies placed on left and right ears respectively." ('666 patent, 10:8-25.)

160. The claims of the '666 patent were amended during prosecution, however, to describe "receiving a first microphone signal from the first microphone" and separately "generating an ambient sound signal." (PAT-B, Mar. 18, 2021 Response to Office Action.) This amendment was made in response to the examiner's rejection based on U.S. Patent Application Publication No. 2004/0179694 ("Alley"). (PAT-B, Jan. 28, 2021 Response to Office Action.) The

applicant did not expressly describe the written description support for the amendment. The amendment is shown in the clip below.

Listing of Claims:

1.(currently amended) An earphone ~~communication device configured to detect a warning~~ comprising:

a first microphone configured to measure an ambient acoustic environment;

a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone;

a speaker configured to play an audio ~~content~~ signal;

a memory that stores instructions; and

a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is ~~communicatively~~ coupled to the processor, and the operations comprising:

receiving a first microphone signal from the first microphone;

receiving a second microphone signal from the second microphone;

generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;

applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;

mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and

sending the mixed audio signal to the speaker.

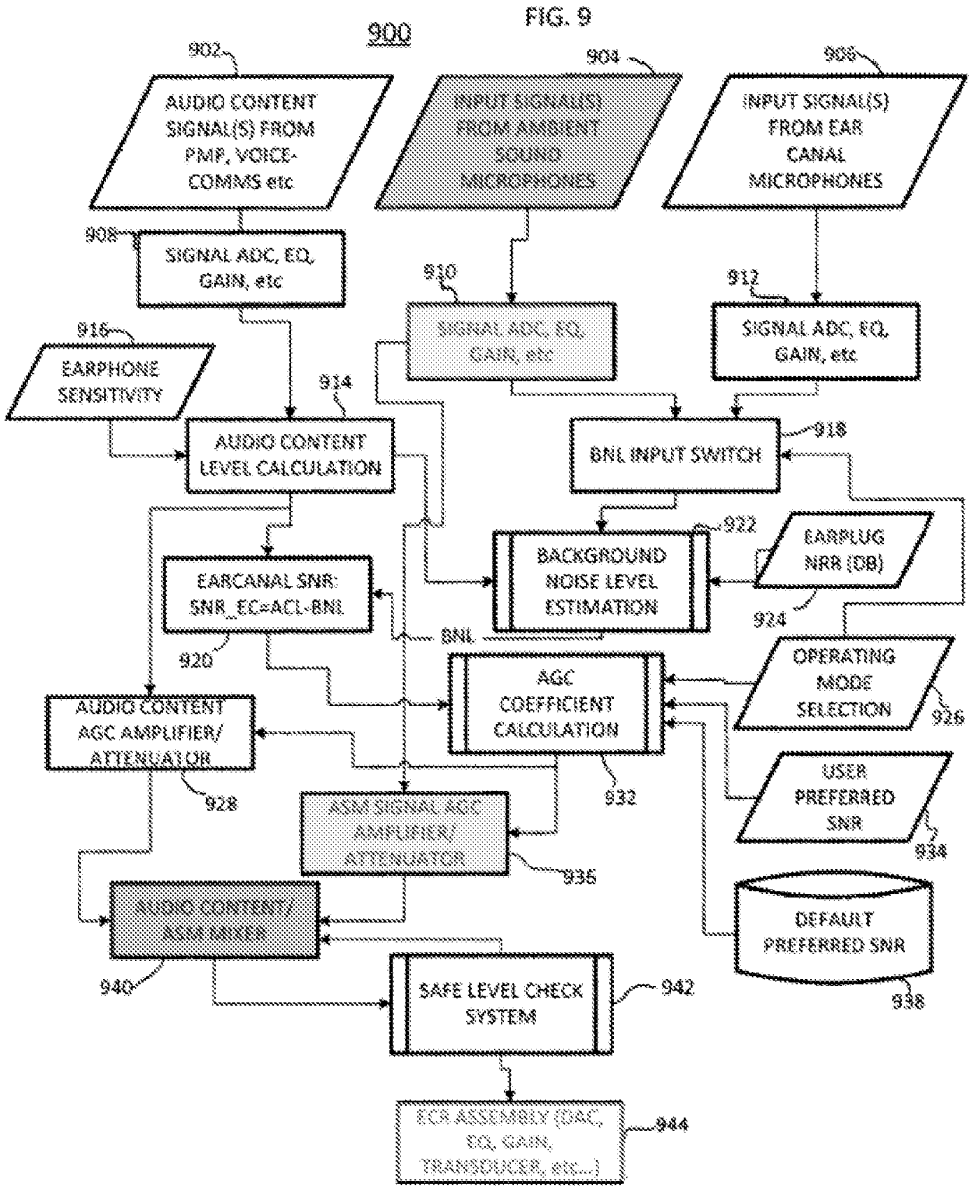
~~analyzing the microphone signal to detect a warning; and~~

~~if a warning is detected then sending a notification signal.~~

(PAT-B, Mar. 18, 2021 Response to Office Action, Amendments to the Claims)

161. In view of the above, a person having ordinary skill in the art would have understood that the broadest reasonable interpretation of the step of "generating an ambient sound signal" includes any manipulation of the signal received from the

ambient sound microphone. This is consistent with the description of the method of Figure 9, reproduced below, which describes that "an input signal is captured from the ASM 111 and ***processed*** at step 910 (e.g. ADC, EQ, gain)." (PAT-A, 12:22-33). Then at step 936 the ambient sound signal ("ASM") is amplified or attenuated (step 936), mixed (step 940) and reproduced with the ECR (step 944). (PAT-A, 12:44-64.)



'666 Patent, FIG. 9 (annotated)

162. Thus, a person having ordinary skill in the art would have understood manipulating the signal from the ambient sound microphone, for example, performing analog to digital conversion on the signal, equalizing the signal, or

applying a gain to the signal to comprise "generating an ambient sound signal." Moreover, a person having ordinary skill in the art would have understood that Nemirovski's disclosure relating to deriving a signal of externally generated sounds by filtering or mathematically removing noise detected by internal microphone 34 from the external signal from external microphone 54 to comprise "generating an ambient sound signal."

163. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest processing circuitry 14", implemented as a processor or a signal processing unit, deriving a signal of externally generated sounds ("generating an ambient sound signal") by filtering or mathematically removing the noise detected by internal microphone 34 from the external signal from external microphone 54 ("from at least one of the first microphone signal or the second microphone signal or a combination of both signals") to provide a signal of externally generated sounds that may be more clearly heard by the wearer.

g. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

164. Nemirovski in combination with Rast discloses or suggests this feature. Rast discloses that in "hearthrough mode," the "acoustics of the environment" are routed to the headset wearer at a "user selected amplification level." (Rast (PA-B),

¶[0017].) Rast further discloses that "one preferred method delivers environmental audio to the wearer during a predetermined interval by amplifying the signal from the microphone and playing it over the speakers within the earpieces." (*Id.*, ¶[0022]; *see also id.*, ¶¶[0051], [0053], [0057], [0058], [0060], [0061], and Table 1.) Rast further explains that in "hearthrough" mode, "external sounds picked up by the microphones are amplified and played within the earcups, ***preferably in concert with*** attenuating the audio amplitude of the programmed sound source." (*Id.*, ¶[0058]; *see also id.*, ¶¶[0011], [0012], [0014], [0021], [0023], [0025], [0051], [0059], [0060], Table 1.)

165. In the Nemirovski-Rast device, a person having ordinary skill in the art would have considered it obvious to amplify and mix Nemirovski's derived signal of externally generated sounds for the purpose of implementing Rast's hearthrough mode because Rast teaches playing the externally generated sounds within the speaker. And it would have been obvious to use Nemirovski's derived signal of externally generated sounds for that purpose (as opposed to the unprocessed external microphone signal) because it is a noise-reduced signal. Indeed, Rast itself suggests using a noise-reduced signal for its hearthrough feature. (*Id.*, ¶[0014] (implementing hearthrough by "amplifying the ambient sounds ***after extracting spurious noises***").)

166. Rast discloses that once a match is made between sound received at the microphones 58 and stored audio parameters, the DSP "output[s] via a digital-to-

analog converter the sound collected from the microphones to an amplifier circuit 68 which is then buffered by amplifiers 70a, 70b to the separate channels of the earpiece speakers 56a, 56b." (*Id.*, ¶[0053].) Rast further discloses that, although the described headset is an analog embodiment (e.g., where the amplifier circuit is analog), "digital headsets" are also within the invention. (*Id.*, ¶[0063].) For example, Rast notes that, when used with a digital input source, having a digital signal processor would "simplif[y] the system" because the audio input would be routed through the DSP/microcontroller on the way to the speakers, enabling power to be easily derived from the digital signal or bias level embedded within the signal. (*Id.*) Indeed, a person having ordinary skill in the art would have considered it obvious to implement the amplifier circuit and buffering amplifiers 68 and 70 using the DSP, and a person having ordinary skill in the art would have been motivated to do so.

167. Indeed, the techniques of digital signal processing and digital signal processors were well-known and widely used by January 2007, and in fact were taught to undergraduates in universities. (*See* Mulgrew, Bernard *et al.*, Digital Signal Processing: Concepts and Applications (2nd ed. 2002) ("Mulgrew") (PA-W).) Advances in the microelectronics industry greatly improved the speed, complexity, and storage capacity of DSPs by the year 2000, and certainly 2007, making DSPs an obvious choice over analog circuits.

> **The main attraction of digital processors is that their accuracy**, which is controlled by the quantisation [*sic*] step size or word length employed in the A/D converter, is extendable only at the cost of greater complexity for the ensuing processing operations, i.e. additions and multiplications. However, floating-point number representations permit the accuracy to be maintained over a wide dynamic range. **Further, digital processors are generally repeatable and much less sensitive to noise than analogue processors** as they are dealing with binary symbols and the processed outputs must always possess binary values. **The microelectronics industry has been reducing continuously silicon VLSI circuit geometrics and hence improving the speed, complexity and storage capacity. The feature size in the 1970s was 5µm while in the year 2000 we were designing VLSI circuits with 0.18µm feature sizes and single chip complexities of 40 M individual transistors, and by the year 2010 0.7µm feature size circuits are predicted to have 3GHz clock rates.**

(PA-X, xxiv.)

168. A person having ordinary skill in the art would also have understood that a widely known and common technique for amplifying an audio signal in the digital domain was to adjust (e.g., increase or decrease) its gain. More particularly, in the digital domain, amplifying a signal entails multiplying it by a factor. If that factor is more than 1, the gain of the signal would be increased. If, on the other hand, that factor is less than 1, the signal would be attenuated, i.e., the gain would be

decreased. Accordingly, a person having ordinary skill in the art would have understood that when Rast's amplifier circuit 68 and amplifiers 70 are implemented by programming processing unit 14", the gain of an audio signal could be increased or decreased.

169. Accordingly, the Nemirovski-Rast device discloses or suggests amplifying in the digital domain the signal of externally generated sounds derived by the processing unit 14" implemented as a processor or signal processing unit, which in turn represents the sound captured by external microphone 54 ("applying an ambient sound gain to the ambient sound signal"), to generate an amplified signal of externally generated sounds ("a modified ambient sound signal").

h. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

170. Nemirovski in combination with Rast discloses or suggests this feature. For example, Rast discloses that a headset may receive "audio signals" such as music programming or communications. (Rast (PA-B), ¶¶[0011], [0022], [0025], [0055], [0059], [0061].) Nemirovski similarly discloses broadcasting received sounds from a cell phone to the user. (Nemirovski (PA-A), 14:1-3.) These are each consistent with how the '666 patent describes an "audio content signal." (PAT-A, 12:26-30 (describing audio content signal 902 as "a music audio signal from a portable Media Player or mobile-phone").)

171. As discussed in Section 154, Rast explains that in "hearthrough" mode, "external sounds picked up by the microphones are amplified and played within the earcups, preferably in concert with attenuating the audio amplitude of the programmed sound source." (Rast (PA-B), ¶[0058]; see also *id.*, ¶¶[0011], [0012], [0014], [0021], [0023], [0025], [0051], [0059], [0060], Table 1.) Rast also explains that, in some embodiments, "[t]he headset preferably mixes the incoming audio with the ambient sounds so that the radio communication function is realized without the wearer losing audio contact with the environment." (*Id.*, ¶[0061].)

172. As also explained in Section 154, in the Nemirovski-Rast device, a person having ordinary skill in the art would have considered it obvious to amplify and mix Nemirovski's derived signal of externally generated sounds for the purpose of implementing Rast's hearthrough mode. And as explained in Section 154, a person having ordinary skill in the art would have understood that although Rast describes an embodiment in which the mixing is performed in an analog circuit, a person having ordinary skill in the art would have been motivated to implement the system digitally such that a processor digitally mixes the ambient sound signal and the incoming received sounds.

173. Accordingly, a person having ordinary skill in the art would have understood the combined Nemirovski-Rast device to disclose or suggest mixing the amplified signal of externally generated sounds with the received sounds from, e.g.,

a cell phone ("mixing the modified ambient sound signal with an audio content signal") to generate a signal that is broadcast to the user ("to generate a mixed audio signal").

i. 1[k] "sending the mixed audio signal to the speaker."

174. Nemirovski in combination with Rast discloses or suggests this feature. For example, Rast explains that, in the "hearthrough mode," environmental audio is delivered to the wearer during a predetermined interval by amplifying the signal from the microphone and "playing it over the speakers within the earpieces," and "[p]referably, any incoming audio source, such as music over a wired or wireless connection, is attenuated or disconnected during the period of time the environmental audio is being played." (Rast (PA-B), ¶[0022]; *see also id.*, ¶¶[0011], [0012], [0014], [0023], [0025], [0051], [0058], [0059], [0060], [0061], Table 1.)

175. Accordingly, a person having ordinary skill in the art would have understood the combined Nemirovski-Rast device to disclose or suggest sending the mixed signal comprising the amplified signal of externally generated sounds and the sounds received from a cell phone to speaker 38 ("sending the mixed audio signal to the speaker").

3. Claim 2

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and"

176. Nemirovski in combination with Rast discloses or suggests this feature. Rast discloses analyzing the ambient sound signal to detect a word spoken by the user or a specific sound in the environment to trigger "hearthrough" mode. (Rast (PA-B), ¶[0021] ("The selection criterion provides the ability for the signal processing unit to discern a variety of environmental acoustics, which by way of example and not of limitation comprise: alarms, screams, horns, screeching brakes, phones ringing, spoken phrases (i.e. 'Hey!', 'Hello', 'Wearer's Name', 'Help'), along with additional acoustic patterns."); *see also id.*, ¶[0055] ("It will be appreciated that a manual mode change may be activated by various touch patterns as well as by the wearer themselves saying a word or phase [sic] which has been programmed or recorded for discrimination .... Preferably the headsets are preprogrammed to respond to specific commands or tone sequences; which may include 'Help', the cadence and sound of a phone ringing, the sound of a doorbell, or a siren.").)

177. Rast further teaches a "method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics and changing

the mode." (*Id.*, ¶[0060].) First, "[t]he sound being recorded from the microphones is considered to be placed in a circular buffer, such as by a periodic interrupt, wherein the processor is capable of operating on information from the buffer." (*Id.*) Then "[a] pointer to the correlation sound, or trigger condition, is advanced at block 104 and the sound bite is compared at block 106." (*Id.*) "If the sound does not match this particular correlation sound (stored sound selection criterion) then a check is made to determine if this is the last correlation sound in the list at block 110. If this is not the last correlation sound, then processing returns to block 104 which advances the correlation sound, trigger condition, to the next condition, and sound comparisons continue." (*Id.*)

178. "Upon the sound bite matching the correlation sound as detected at block 108, execution reaches an extended comparison routine, wherein subsequent portions, or characteristics, of the external sound are compared with subsequent portions of the stored correlation sound at block 114." (*Id.*). "If the sound is validated then block 118 is reached which sets the mode of the headset according to the configuration as set by the user." (*Id.*) Thus, Rast describes conducting a sound bite-by-sound bite comparison to stored sounds. If an initial sound bite match is found, then an "extended comparison routine" is performed. If not, the system proceeds to check the next potential sound, if any.

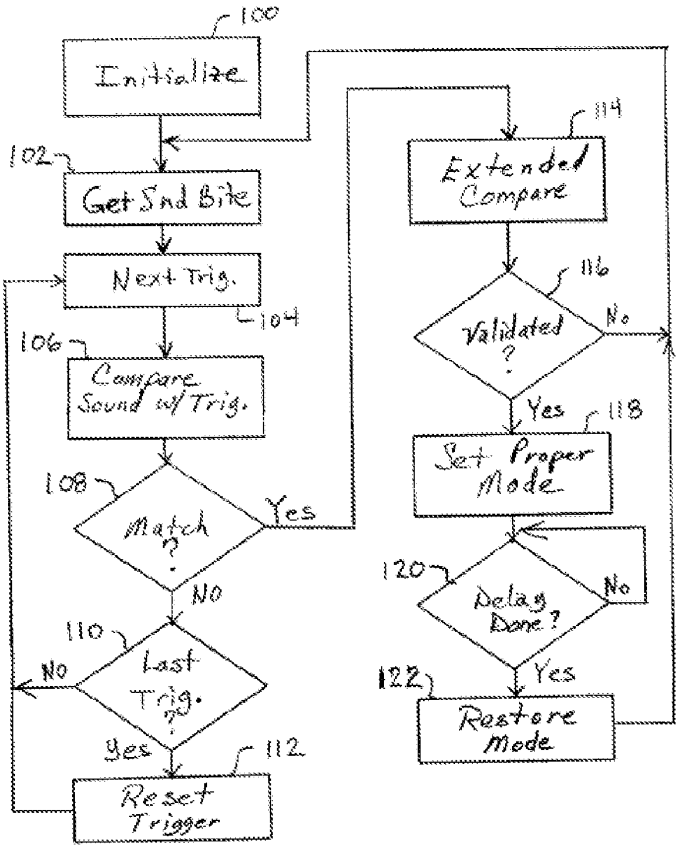


FIG. 5

Rast, FIG. 5

179. This is consistent with how the '666 patent describes this feature. The '666 patent explains that the "the processor 206 can search for on-sets (e.g., fast rising amplitude wave-front) of an acute sound or other abrupt feature characteristics without initially attempting to initially identify or recognize the sound source. That is, the processor 206 is actively listening for a presence of acute sounds before identifying the type of sound source." (PAT-A, 7:52-58; *see also id.*, FIG. 3.) A

person having ordinary skill in the art would have understood Rast's sound bite-by sound bite routine to be consistent with searching for the on-set of a programmed sound.

180. As discussed in Section X.B.1, a person having ordinary skill in the art would have been motivated to modify Nemirovski's system based on Rast to selectively provide ambient sounds to the user in response to the detection of environmental audio matching specific criterion. To modify Nemirovski's system to selectively provide ambient sounds to the user in addition to a user's speech for the reasons discussed in Section X.B.1, a person having ordinary skill in the art would have found it obvious to modify Nemirovski's system to utilize the technique described in Rast that permits the detection of both the user's voice as well as other specific sounds in the environment, such as alarms, screams, or a siren. A person having ordinary skill in the art would have been motivated to do so because it would have amounted to nothing more than applying a known technique (Rast's sound detection technique to trigger the "hearthrough" mode) to a known device (Nemirovski's ear sensor system) ready for improvement to yield predictable results (an ear sensor system configured to selectively provide ambient sounds).

181. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest processing circuitry 14" implemented as a processor or signal processing unit being programmed to

conduct a sound bite-by-sound bite comparison of sounds received from the external microphone to stored sounds ("detecting an acute sound by analyzing … the first microphone signal ….").

b. 2[b] "determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals."

182. Nemirovski in combination with Rast discloses or suggests this feature. As explained in Section X.B.3.a, Rast discloses conducting a sound bite-by-sound bite comparison to stored sounds. If there is a match between a sound bite and a stored sound, "execution reaches an extended comparison routine, wherein subsequent portions, or characteristics, of the external sound are compared with subsequent portions of the stored correlation sound at block 114." (Rast (PA-B), ¶[0055].) And "[i]f the sound is validated then block 118 is reached which sets the mode of the headset according to the configuration as set by the user." (*Id.*) Rast further discloses determining whether the sound received by the external microphone is a word uttered by the user. (*Id.*, ¶[0055] ("It will be appreciated that a manual mode change may be activated by various touch patterns as well as ***by the wearer themselves saying a word or phase*** [*sic*] which has been programmed or recorded for discrimination.").)

183. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to also disclose or suggest validating, for example, whether the user themselves said a programmed word or phrase ("determining whether the acute sound is a user's voice") via an extended comparison routine wherein subsequent portions, or characteristics, of the external sound are compared with subsequent portions of the stored sound ("by analyzing … the first microphone signal ….").

184. For the reasons discussed in Sections X.B.1 and X.B.3.a, a person having ordinary skill in the art would have been motivated to combine the teachings of Nemirovski and Rast.

4. Claim 3

a. "3. The earphone according to claim 2, where the operations further comprise: determining whether the acute sound is a warning sound or siren by analyzing the spectrum of at least one of the first microphone signal or the second microphone signal or a combination of both signals."

185. Nemirovski in combination with Rast discloses or suggests this feature. As discussed in Section X.B.3, Rast discloses analyzing the ambient sound signal to detect a word spoken by the user or a specific sound in the environment in order to trigger "hearthrough" mode. (Rast (PA-B), ¶[0021] ("The selection criterion provides the ability for the signal processing unit to discern a variety of

environmental acoustics, which by way of example and not of limitation comprise: alarms, screams, horns, screeching brakes, phones ringing, spoken phrases (i.e. 'Hey!', 'Hello', 'Wearer's Name', 'Help'), along with additional acoustic patterns.").) In particular, Rast provides that "[p]referably the headsets are preprogrammed to respond to specific commands or tone sequences; which may include 'Help', the cadence and sound of a phone ringing, the sound of a doorbell, or ***a siren***." (*Id.*, ¶[0055]; *see also id.*, ¶¶[0059], [0062], Table 1.) Rast further discloses that, to identify a siren, the processing unit may use "an algorithm that is capable of discerning a wide variety of common sirens in use." *Id.*, ¶[0018].

186. Rast also discloses analyzing environmental audio using "numerous current methods" including "parametric analysis, waveform comparisons, Fourier analysis, and in a larger context voice recognition" (*id.*, ¶[0021]), and "waveforms, waveform characteristics, or characteristics of sound, depending on the algorithms that are being utilized for providing correlation" (*id.*, ¶ [0060].)

187. Rast further discloses that the sounds may be stored using sound signaturing. (*Id.*, ¶[0016] ("For simplicity, this retained information is often referred to as "stored sounds", or "sound characterizations", however, it is to be recognized that these "stored sounds" may comprise any combination of sound representations, sound characterizations, sound related algorithms, sound correlation programming, sound scripting, ***sound signaturing***, along with additional methods of storing

information about audio waveforms.").) A person having ordinary skill in the art would have understood discerning sounds based on stored sound signatures to include performing a spectrum analysis. Indeed, U.S. Patent No. 9,135,797 ("Couper") (PA-V) discloses a system for detecting sound signatures in ambient sound similar to those that Rast is concerned with (e.g., sirens and other emergency sounds). (Couper (PA-V), Abstract, 4:60-5:4.)[12] Couper explains that its "signature detector 115" "can extract information from the detected sound and compare that information to the sound signatures 120" and that "[e]xamples of the sort of data that can be extracted or determined from the detected sound can include, but are not limited to, volume or sound-pressure-level information, the frequency range of the detected sound, the amount of energy detected in different frequency bands of interest, ***a spectrum analysis***, transient characteristics, the actual waveform of the detected sound, a Fourier Transform or FFT information, formant information, or the like." (*Id.*, 5:18.)

[12] Here, I am relying on Couper (PA-V) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 1.

188. Accordingly, a person having ordinary skill in the art would have understood Rast to disclose or suggest storing sounds using sound signaturing, and that matching a detected sound with the stored sound signature for a siren commonly in use would include performing a signal spectral analysis. Therefore, a person having ordinary skill in the art would have found it obvious to use an algorithm that analyzes the spectrum of the external microphone signal to discern whether the external microphone signal includes one of a wide variety of sirens in use.

189. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest identifying a siren within an external microphone signal ("determining whether the acute sound is a warning sound or siren") by applying an algorithm capable of discerning a wide variety of common sirens in use to the external microphone signal ("by analyzing the spectrum of … the first microphone signal ….").

5. Claim 4

a. "4. The earphone according to claim 3, where the operations further comprise: decreasing a volume of the audio playback when a warning or siren is detected."

190. Nemirovski in combination with Rast discloses or suggests this feature. Rast discloses that, if external sound is correlated with programmed sound, sound registered by the microphones is conditioned and directed to the audio annunciators

within the earcups. (Rast (PA-B), ¶[0051].) As explained in Sections X.B.3 and X.B.4, the programmed sound may comprise a siren. (*Id.*, ¶[0055].) In addition to the registered sound being conditioned and directed to the user, "the audio source being listened to may be either ***attenuated***, or disconnected, such that the amplified external sounds from the microphones may be more clearly heard by the wearer." (*Id.*, ¶[0051].)

191. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest attenuating an audio source being listened to ("decreasing a volume of the audio playback") if external sound is correlated with the sound of a siren ("when a warning or siren is detected").

6. Claim 5

a. "5. The earphone according to claim 4, where the operations further comprise: sending a notification signal to the speaker."

192. Nemirovski in combination with Rast discloses or suggests this feature. Rast explains that, if external sound is correlated with programmed sound (such as a siren), "the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound." (Rast (PA-B), ¶[0060].) Rast further discloses that "the wearer may be alerted with ***additional warning tones***, or a playback of the sound taken from the environment or the correlation sound." (*Id.*; *see also id.*, ¶[0017] ("It is anticipated that the wearer

will typically choose to have the headset temporarily change to a hearthrough mode which may be augmented with sound echo, ***alerts***, and so forth."), ¶[0051] ("It will be appreciated, however, that the wearer may alternatively, or additionally be alerted to the correlated sound in a number of ways, ***such as alerts***, and playing of the correlated sound, or the recorded correlation sound.").) A person having ordinary skill in the art would have understood that an additional warning tone or alert in addition to the earpiece switching to "hearthrough" mode comprises a notification signal sent to the speaker.

193. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest alerting a wearer with additional warning tones or alerts ("sending a notification signal to the speaker") when a siren within an external microphone signal is identified.

7. Claim 6

a. "6. The earphone according to claim 4, wherein the warning is at least one of a bell or the sound of an emergency vehicle or the sound of a security system or a combination."

194. Nemirovski in combination with Rast discloses or suggests this feature. Rast discloses detecting a correlation between environmental sounds from microphones 58a, 58b and stored sound parameters. (Rast (PA-B), ¶[0025], [0053].) Rast discloses that the stored sound may comprise specific commands or tone

sequences, such as "the sound of a doorbell, or a siren." (*Id.*, ¶[0055]; *see also id.*, ¶[0018], [0021], [0059], [0062], Table 1.) Rast also explains that a "siren" may comprise "the siren of an emergency vehicle." (*Id.*, ¶[0008].)

195. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest identifying the sound of a doorbell or the siren of an emergency vehicle ("wherein the warning is at least one of a bell or the sound of an emergency vehicle ….").

8. Claim 7

a. "7. The earphone according to claim 4, wherein the siren is at least one of a police siren or an ambulance siren or a car honking or a combination."

196. Nemirovski in combination with Rast discloses or suggests this feature. Rast discloses detecting a correlation between environmental sounds from microphones 58a, 58b and stored sound parameters. (Rast (PA-B), ¶[0025], [0053].) Rast discloses that the stored sound may comprise specific commands or tone sequences, such as "the sound of a doorbell, or a siren." (*Id.*, ¶[0055]; *see also id.*, ¶[0018], [0021], [0059], [0062], Table 1.) Rast also explains that a "siren" may comprise "the siren of an emergency vehicle." (*Id.*, ¶[0008].) A person having ordinary skill in the art would have understood that Rast's reference to an emergency vehicle encompasses a police car or an ambulance because both are types of

emergency vehicles that have sirens. Rast also describes the processing unit identifying a horn or screeching brakes. (*Id.*, ¶[0021].)

197. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest identifying the siren of an emergency vehicle such a police car or an ambulance or a horn ("wherein the siren is at least one of a police siren or an ambulance siren or a car honking ….").

9. Claim 8

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected."

198. Nemirovski in combination with Rast discloses or suggests this feature. As discussed in Section X.B.3.b, Rast discloses analyzing the signal received from the external microphone to determine whether a sound in the external microphone signal is a word or phrase uttered by the user in order to trigger "hearthrough" mode.

199. Rast explains that the mode change of the headset comprises altering the level of acoustical isolation accorded the wearer by the headset, which may involve "temporarily routing external sound to earpieces (a 'hearthrough mode')," "attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation ('block matched sound')," "amplifying the ambient sounds after extracting spurious noises," "accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants)," or "combinations of the preceding

modes." (Rast (PA-B), ¶[0014].) Rast further discloses that "the audio amplitude of the programmed audio source may be temporarily attenuated, or disconnected, such that the external sound may be more easily discerned by the wearer." (*Id.*)

200. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest attenuating the broadcast of received sounds and amplifying the derived signal of ambient sounds provided to the user when word or phrase spoken by the user is detected ("adjusting the mixed audio signal when the user's voice is detected").

10. Claim 9

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal."

201. Nemirovski in combination with Rast discloses or suggests this feature. Indeed, for the same reasons discussed above in Section X.B.9, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest attenuating the broadcast of received sounds and amplifying the derived signal of ambient sounds provided to the user when word or phrase spoken by the user is detected ("where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal.").

11. Claim 10

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal."

202. Nemirovski in combination with Rast discloses or suggests this feature. As explained above in Section X.B.2.f, Nemirovski discloses or suggests filtering or mathematically removing the noise detected by internal microphone 34 from the external signal from external microphone 54, and to thereby derive a signal of externally generated sounds. As explained in Section 154, in the Nemirovski-Rast device, a person having ordinary skill in the art would have considered it obvious to amplify and mix Nemirovski's derived signal of externally generated sounds for the purpose of implementing Rast's hearthrough mode.

203. A person having ordinary skill in the art would have understood that mathematically removing noise detected by internal microphone 34 from the external signal to comprise creating an inverse signal of the noise detected by internal microphone 34 and combining that inverse signal with the external signal to "derive a signal of externally generated sounds" that is free of unwanted noise. Indeed, a person of ordinary skill in the art would have understood that combining a signal with an inverted noise signal was a widely known and common technique for canceling noise in one signal using another signal. For example, the Beasley reference describes utilizing such an inverse or complementary audio signal to

cancel out noise. (Beasley (PA-F), ¶¶[0031], [0050]-[0051].)[13] As such, a person of ordinary skill in the art would have understood Nemirovski's derived signal of externally generated sounds to include the external signal from external microphone 54 and an inverse signal of the noise detected by internal microphone 34.

204. A person having ordinary skill in the art would have understood that the broadest reasonable interpretation of "a noise reduction signal" includes any signal that cancels or otherwise reduces noise in the output signal. Therefore, a person of ordinary skill in the art would have understood the inverse signal of the noise detected by internal microphone 34 to comprise "a noise reduction signal."

205. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest a mixed audio signal ("where the mixed audio signal") comprising a derived signal of externally generated sounds that includes an inverse signal of the noise detected by internal microphone 34 ("includes a noise reduction signal derived from the second microphone signal").

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[13] Here, I am relying on Beasley (PA-F) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 1.

12. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

206. Nemirovski in combination with Rast discloses or suggests this feature. Nemirovski discloses that the housing 150 includes an interior portion 154 made from a "pliable material, such as rubber, and is sized and shaped to snugly fit within the ear" and may be made from "sound absorbing (or sound proofing) material." (Nemirovski (PA-A), 14:19-25.) Nemirovski further discloses that the housing 150 "at least partially blocks externally generated sound waves before reaching the [internal] microphone 54." (*Id.*, 13:3-7; *see also id.*, 12:16-18 ("Such arrangement at least partially insulates the interior microphone 34 from externally generated noise and air pressure changes.").) The '666 patent explains that a "a seal is typically achieved by means of a soft and compliant housing of assembly 113" and that the seal "serves to significantly reduce the sound pressure level at the user's eardrum 133 resulting from the sound field at the entrance to the ear canal." (PAT-A, 5:9-19.)

207. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast device to disclose or suggest that housing 150 includes an interior portion 154 made from a pliable material that snugly fits within

the ear to at least partially block externally generated sound waves ("wherein the earphone includes a sealing section").

C. Ground 2: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Dijkstra

208. As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and persons having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground further addresses the circumstance in which Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in the housing of a device in which the microphone is mounted.

1. Independent Claim 1

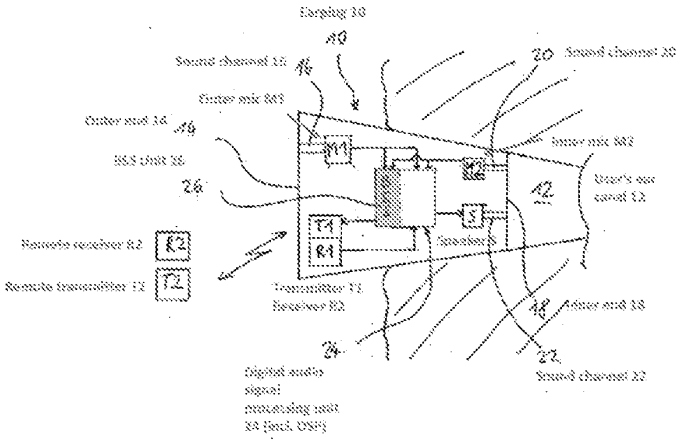
209. With respect to 1[pre] and 1[c]-1[k], Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section X.B, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section X.B.2.a.
1[c]	*See supra* Section X.B.2.c.
1[d]	*See supra* Section X.B.2.d.
1[e]	*See supra* Section X.B.2.d.
1[f]	*See supra* Section X.B.2.e.
1[g]	*See supra* Section X.B.2.e.
1[h]	*See supra* Section X.B.2.f.
1[i]	*See supra* Section 154.
1[j]	*See supra* Section X.B.2.h.
1[k]	*See supra* Section X.B.2.i.

210. With respect to 1[a] and 1[b], Nemirovski in combination with Rast discloses or suggests "a first microphone configured to measure an ambient acoustic environment" and "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone" in the same way as discussed above in Section X.B.2.b. To the extent that Nemirovski and Rast are determined to not teach or render obvious "wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted" and "wherein the second microphone has a

second microphone port that is configured to face toward the user when the earphone is inserted," as recited in independent clam 1, Dijkstra teaches these limitations within their broadest reasonable interpretation. Furthermore, as explained below, a person having ordinary skill in the art would have been motivated to combine Dijkstra with Nemirovski and Rast with a reasonable expectation of success.

211. Dijkstra is directed to a hearing protection device comprising an earplug to be worn at a user's ear or partly in the user's ear canal. (Dijkstra (PA-C), Abstract, ¶¶[0001], [0013], [0026], [0029].) Dijkstra describes earplug 10 having an outer microphone M1 oriented outwardly toward the environment and an inner microphone M2 oriented inwardly towards the user's ear canal. (*Id.*, Abstract, ¶¶[0010], [0013]-[0015], [0029], [0031], [0034].) FIG. 1, which depicts earplug 10 having outer microphone M1 and inner microphone M2, is reproduced below:



Dijkstra, FIG. 1 (annotated)

212. Dijkstra discloses that outer microphone M1 "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." (*Id.*, ¶[0029].) Dijkstra also discloses that inner microphone M2 "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." (*Id.*)

213. A person having ordinary skill in the art would have understood Dijkstra's outer microphone M1 which is open to the outer end 14 of the earplug 10 via sound channel 16 and inner microphone M2 which is open to the inner end of the earplug 10 via a sound channel 20 to each include microphone ports. Indeed, sound channel 16 and sound channel 20 comprise openings associated with outer microphone M1 and inner microphone M2, respectively, through which measurements are taken. As such, under the broadest reasonable interpretation of claim 1, Dijkstra's sound channel 16 and sound channel 20 comprise microphone ports within the meaning of the '666 patent.

214. Notwithstanding the foregoing, and to the extent Patent Owner asserts that a "microphone port" refers specifically to an opening in the housing of a device in which the microphone is mounted, Dijkstra's sound channel 16 and sound channel 20 disclose or suggest microphone ports for at least the reason that a person having ordinary skill in the art would have understood that sound channel 16 and sound channel 20 terminate at openings in earplug 10. Indeed, the prior art uses the term

"port" to refer to an opening in the housing of a device. For example, Ngia describes how "[s]ounds from the ear canal 110 may be conveyed to the microphone 18 through an air medium via a channel 40" and the opening is referred to as a "port 38," as illustrated in Figure 5 reproduced below. (Ngia (PA-K), 9:18-27.)[14]

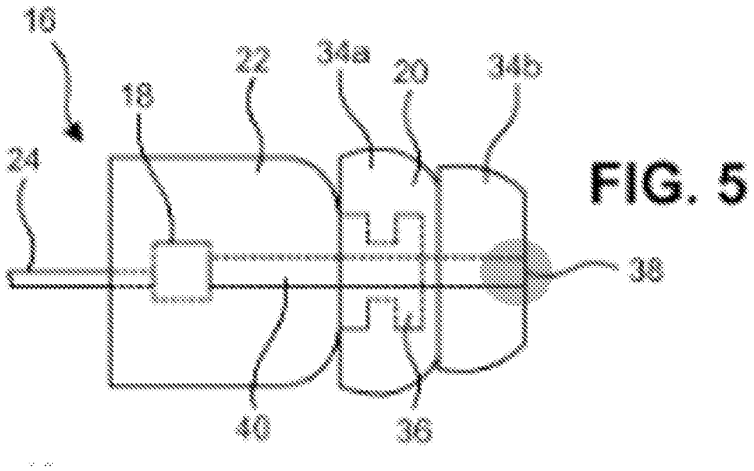


FIG. 5

Ngia, FIG. 5 (annotated)

215. Moreover, the prosecution history of the '666 patent itself supports a conclusion that Dijkstra discloses or suggests "wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone

[14] Here, I am relying on Ngia (PA-K) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 2.

is inserted" and "wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted." *See supra* Section VII.A. During prosecution of the '666 patent, the Office rejected claim 1 relying on Dijkstra to teach or suggest the "first microphone port" and "second microphone port" limitations. (PAT-B, Aug. 16, 2021 Final Office Action, 3-5.) In response, the applicant did not dispute the Office's determination that Dijkstra taught the "first microphone port" and "second microphone port" limitations, instead overcoming the rejection on other grounds. (*See generally* PAT-B, Oct. 25, 2021 Response to Office Action.)

216. A person having ordinary skill in the art would have found it obvious to modify the Nemirovski-Rast device to dispose Nemirovski's external microphone 54 and internal microphone 34 within housing 150 based on the teachings of Dijkstra with respective sound channels providing fluid communication between external microphone 54 and the air external to sensor 12 and between internal microphone 34 and the air inside the ear canal.

217. A person having ordinary skill in the art would have recognized that disposing external microphone 54 and internal microphone 34 within housing 150 would better protect the microphones from physical damage. A person having ordinary skill in the art would also have recognized that disposing external microphone 54 and internal microphone 34 within housing 150 would shield the

microphones from influences that would impact their performance and ability to detect a word spoken by the user or a specific sound in the environment in the combined Nemirovski-Rast device.

218. For example, U.S. Patent No. 8,150,084 ("Jessen") (PA-R) discloses a hearing aid with a microphone including a "a port in the surface of the housing and a canal from the port to an electronic part of the microphone within that housing. By the use of a canal the wind noise is damped." *Jessen*, 2:55-61.

219. A person having ordinary skill in the art would have considered the combination of Nemirovski, Rast, and Dijkstra to be predictable and would have had a reasonable expectation of success in disposing Nemirovski's external microphone 54 and internal microphone 34 within housing 150 based on the teachings of Dijkstra in the combined Nemirovski-Rast device. A person having ordinary skill in the art would have readily understood that the teachings of Dijkstra could be readily applied to a device already having substantially the same structure, such as the Nemirovski-Rast device.

220. Nemirovski also already describes an embodiment in which internal microphone 34 is disposed within a hollow cavity defined by the housing 150. (Nemirovski (PA-A), 12:64-13:2.) A person having ordinary skill in the art would have thus found it obvious to incorporate Dijkstra's teachings into the combined Nemirovski-Rast device because it would have amounted to merely applying a

known technique (Dijkstra's disposition of internal and external microphones within housing) to improve a similar device (the combined Nemirovski-Rast device) in the same way (to protect the microphones from physical damage and shield them from influences that would impact performance).

221. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast-Dijkstra device to disclose or suggest an external microphone 54 disposed within housing 150 and in fluid communication with air external to sensor 12 via a channel ("wherein the first microphone has a first microphone port") to allow external microphone 54 to detect sounds outside the ear ("that is configured to face away from a user when the earphone is inserted"), and an internal microphone 34 disposed within housing 150 and in fluid communication with air inside the ear canal via a channel ("wherein the second microphone has a second microphone port") to allow internal microphone 34 to detect sounds within the ear canal ("that is configured to face toward the user when the earphone is inserted").

2. Dependent Claims 2-11

222. With respect to dependent claims 2-11, Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section X.B, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section X.B.3.
3	*See supra* Section X.B.4.
4	*See supra* Section X.B.5.
5	*See supra* Section X.B.6.
6	*See supra* Section X.B.7.
7	*See supra* Section X.B.8.
8	*See supra* Section X.B.9.
9	*See supra* Section X.B.10.
10	*See supra* Section X.B.11.
11	*See supra* Section X.B.12.

223. Accordingly, for at least the same reasons, these claims would have been obvious over Nemirovski, Rast, and Dijkstra.

D. Ground 3: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Van Doorn

224. As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and persons having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground addresses the circumstance in which

Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in a microphone package itself. Under such circumstances, it is my opinion that that Nemirovski in combination with Rast and Van Doorn would have rendered obvious claims 1-11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. Independent Claim 1

225. With respect to 1[pre] and 1[c]-1[k], Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section X.B, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section X.B.2.a.
1[c]	*See supra* Section X.B.2.c.
1[d]	*See supra* Section X.B.2.d.
1[e]	*See supra* Section X.B.2.d.
1[f]	*See supra* Section X.B.2.e.
1[g]	*See supra* Section X.B.2.e.
1[h]	*See supra* Section X.B.2.f.
1[i]	*See supra* Section 154.
1[j]	*See supra* Section X.B.2.h.
1[k]	*See supra* Section X.B.2.i.

226. With respect to 1[a] and 1[b], Nemirovski in combination with Rast discloses or suggests "a first microphone configured to measure an ambient acoustic environment" and "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone" in the same way as discussed above in Section X.B.2.b. To the extent that Nemirovski and Rast are determined to not teach or render obvious "wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted" and "wherein the second microphone has a

second microphone port that is configured to face toward the user when the earphone is inserted," as recited in independent clam 1, Van Doorn teaches these limitations. Furthermore, as explained below, a person having ordinary skill in the art would have been motivated to combine Van Doorn with Nemirovski and Rast with a reasonable expectation of success.

227. Van Doorn is directed to "a microphone or listening device with an improved sound inlet port." (Van Doorn (PA-D), ¶[0001].) Van Doorn discloses a microphone 110 with a cup-shaped sound input plate 50 that is designed for miniature microphones. (*Id.*, ¶[0020].) Van Doorn discloses that sound input plate 50 includes an outer wall 52 and a sound port 56 comprising an opening in outer wall 52 that leads to an interior recess 54. (*Id.*) FIG. 3, reproduced and annotated below, depicts a microphone 110 with sound inlet plate 50:

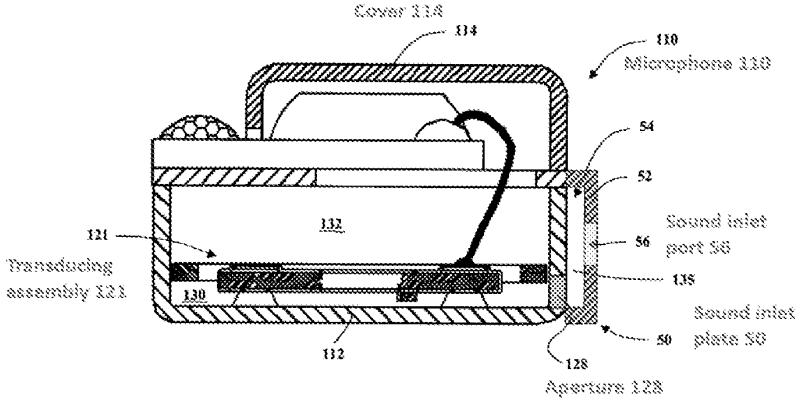


FIG. 3

Van Doorn, FIG. 3 (annotated)

228. Van Doorn explains that, "[i]n operation, the sound inlet plate 50 receives sound through the sound inlet port 56 in its exterior wall 52. The sound propagates through a passageway 135 that is defined by the interior recess 54 of the plate 50 and the exterior wall of the housing 112 adjacent to the aperture 128. Eventually, the sound is transmitted through the aperture 128 and acts upon the transducing assembly 121." (*Id.*, ¶[0022].)

229. Van Doorn explains that, unlike typical prior art microphones, "the sound inlet plate 50 only protrudes slightly away from the housing 112. One design aspect leading to the minimal protrusion feature of the plate 50 is the fact that the sound port 56 is simply an opening or hole in the outer wall 52." (*Id.*, ¶[0023]; *see also id.*, ¶[0006] ("The sound inlet plate is made very thin so that it does not extend substantially away from the housing.").) Additionally, "the location of the opening of the sound inlet can be offset to a more desirable position (e.g., the midpoint of the microphone exterior surface) to ease installation of the microphone in the hearing aid." (*Id.*, ¶[0006]; *see also id.*, ¶[0024] (further explaining how another benefit of the design is that "sound port 56 can be vertically and/or horizontally offset from the aperture 128 in the housing 112").)

230. A person having ordinary skill in the art would have understood Van Doorn's sound port 56 to comprise an opening in microphone 110 through which measurements are taken. As such, to the extent Patent Owner asserts or the Office

determines that a "microphone port" refers specifically to an opening in the microphone itself, a person having ordinary skill in the art would have understood Van Doorn's sound port 56 to comprise a "microphone port."

231. As discussed above in Section X.B.2.b, Nemirovski discloses a sensor 12 including an external microphone 54 that detects air pressure changes located outside the ear and an internal microphone 34 that detects air pressure changes within the ear. However, Nemirovski is silent as to the type or make up of external microphone 54 and internal microphone 34. As such, a person having ordinary skill in the art would have been motivated for look for microphones for use in sensor 12 as external microphone 54 and internal microphone 34. In so looking, a person having ordinary skill in the art would have readily found Van Doorn.

232. A person having ordinary skill in the art would have found it obvious to modify the Nemirovski-Rast device to use Van Doorn's microphone 110 with a sound inlet plate 50 comprising sound port 56 for external microphone 54 and internal microphone 34. Indeed, a person having ordinary skill in the art would have been motivated to use Van Doorn's microphone 110 because it provides "a more compact design" as compared with other prior art microphones (Van Doorn (PA-D), ¶¶[0006], [0023]), and because sound port 56 can be offset from aperture 128 to ease installation of external microphone 54 and internal microphone 34 in the combined Nemirovski-Rast device (*Id.*, ¶¶[0006], [0024]).

233. A person having ordinary skill in the art would also have recognized Van Doorn's microphone 110 was suitable for inclusion the Nemirovski-Rast device as Van Doorn discloses that its sound inlet plate 50 is designed for use on miniature microphones in hearing aids. Indeed, Van Doorn explains that its microphone package will "ease installation of the microphone in the hearing aid." (*Id.*, ¶[0006].)

234. A person having ordinary skill in the art would have thus found it obvious to incorporate Van Doorn's microphone 110 into the combined Nemirovski-Rast device because it would have amounted to merely a simple substation of one known element (Van Doorn's microphone 110) for another (Nemirovski's external microphone 54 and internal microphone 34) to obtain predictable results (an ear sensor system with external and internal microphones having a more compact design).

235. Accordingly, a person having ordinary skill in the art would have understood the Nemirovski-Rast-Van Doorn device to disclose or suggest an external microphone having a sound port as taught by Van Doorn ("wherein the first microphone has a first microphone port") that is configured to detect sounds outside the ear ("that is configured to face away from a user when the earphone is inserted"), and an internal microphone having a sound port as taught by Van Doorn ("wherein the second microphone has a second microphone port") that is configured to detect

sounds within the ear canal ("that is configured to face toward the user when the earphone is inserted").

2. Dependent Claims 2-11

236. With respect to dependent claims 2-11, Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section X.B, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section X.B.3.
3	*See supra* Section X.B.4.
4	*See supra* Section X.B.5.
5	*See supra* Section X.B.6.
6	*See supra* Section X.B.7.
7	*See supra* Section X.B.8.
8	*See supra* Section X.B.9.
9	*See supra* Section X.B.10.
10	*See supra* Section X.B.11.
11	*See supra* Section X.B.12.

237. Accordingly, for at least the same reasons, these claims would have been obvious over Nemirovski, Rast, and Van Doorn.

E. Ground 4: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Dijkstra and Beasley

As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and person having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground addresses the circumstance in which Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in the housing of a device in which the microphone is mounted. Under such circumstances, it is my opinion that Victorian in combination with Beasley and Dijkstra would have rendered obvious claims 1-2, 8-9, and 11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. A person having ordinary skill in the art Would Have Been Motivated to Combine Victorian and Beasley

238. As explained further below, a person having ordinary skill in the art would have been motivated to combine the teachings of Victorian and Beasley and would have had a reasonable expectation of success in doing so.

239. Victorian discloses an ear-level communication system 100 comprising a full-duplex ear-level device 110 designed to allow two-way simultaneous conversation with a remote device. (Victorian (PA-E), ¶¶[0018]-[0019].) Victorian discloses ear-level device 110 including an "external microphone 420 [that] picks

up ambient sound such that a hearing impaired person wearing ear-level device 110 is not 'isolated' when communicating to remote device 190." (*Id.*, ¶[0030].) Victorian also discloses ear-level device 110 including an "[e]ar canal microphone 440 [that] detects sound from the ear canal," that is primarily the wearer's speech. (*Id.*) Victorian discloses implementing ear-level device 110 with an earmold that fully occludes the ear to maximize audio efficiency. (*Id.*, Abstract, ¶[0027].)

240. Victorian further discloses that ear-level device 110 includes a VOX Circuit with gating modules and a controller that together determine which sounds are provided to the ear canal speaker. (*Id.*, ¶¶[0031]-[0035] (describing ambient sound gating module 421, occluded sound gating module 441, and remote sound gating module 451.) When the amplitude of sound detected by each gating module is over a predetermined threshold, and when permitted by a controller, the modules are turned "on," i.e., "[sound] is allowed to pass and be processed." (*Id.*, ¶¶[0032]-[0035].) When the gating modules are "off," sound is "is blanked or substantially attenuated." (*Id.*, ¶[0032].) In one embodiment, the controller coordinates the gating rules so that sounds are processed "in an orderly fashion," e.g., "if both sounds A and B are detected, whichever sound is detected earlier is gated on." (*Id.*, ¶[0037].) In one embodiment, the controller coordinates the on/off states "according to predetermined or programmed gating coordination rules." (*Id.*, ¶[0038].) One such rule allows the transmission of both the remote sound and the ambient sound to the

ear canal. (*Id.*, ¶[0038].) Victorian expressly discloses that "[o]ther rules are applied as a person skilled in the art should see fit based on an understanding after reading this entire document." (*Id.*)

241. Beasley discloses an audio bypass mode to "to adjust the level of electrical background signal relative to the audio signal" before combining the signals and providing the combined signal to a speaker. (Beasley (PA-F), ¶¶[0010], [0012], [0014], Claims 7, 20.) For example, Beasley discloses that when audio bypass mode is initiated, audio signals may be attenuated and background audio signals may be amplified such that "the signal level, and thus ultimate volume, of background audio signal 312 is increased relative to that of audio signal 314, thus enabling the user to better hear the background audio signal 312 over the audio program content 314 from the audio source of the electronic device." (*Id.*, ¶[0046]; *see also id.*, ¶[0054]). Beasley further discloses automatically engaging audio bypass mode by using one or more of various techniques to detect the occurrence of the one or more events, including voice or speech recognition and background audio level detection. (Beasley (PA-F), ¶¶[0037], [0061], claims 18, 19, 30, 31.)

242. A person having ordinary skill in the art would have been motivated to modify Victorian in view of Beasley. More particularly, a person having ordinary skill in the art would have recognized that implementing Beasley's automatic and programmable audio bypass mode would have provided enhanced and convenient

control over the simultaneous reproduction of Victorian's ambient sound and remote sound together. In the combination, Beasley's automatic bypass mode would be used to amplify the ambient sound relative to the remote sound in response to the user's voice or speech command so that the user could better hear the ambient sound.

243. Accordingly, a person having ordinary skill in the art would have been motivated to combine Victorian and Beasley because the combination would have amounted to applying a known technique (Beasley's technique for triggering ambient sound passthrough) to a known device (Victorian's ear-level communication system) ready for improvement to yield predictable results (enabling a user to programmatically control the provision and relative level of remote sound and ambient sound in Victorian's ear-level device).

244. The combination would also have been obvious to a person having ordinary skill in the art because it would have amounted to the use of a known technique (Beasley's technique for controlling sound isolation) to improve a similar device (Victorian's ear-level device) in the same way (to make them both responsive to the detection of specific words or phrases). The combination would have also amounted to using known work in one field of endeavor (Beasley's programmable audio bypass feature) for use in either the same field or a different one (Victorian's ear-level device) based on design incentives or other market forces (i.e., the ability to automatically adjust Victorian's mixing ratios when certain speech is detected).

245. In the combined Victorian-Beasley system, Beasley's teachings related to implementing automatic ambient sound passthrough would be implemented in Victorian's system. More particularly, the signal generated by Victorian's external microphone 420 would be, according to Beasley's technique, analyzed to detect a particular word or phrase by Victorian's processor 460, and upon detecting a match, Victorian's processor 460 would automatically adjust the ratios of the mixed signals and provide the mixed signal to the speaker at the headset.

246. There is ample evidence that a person having ordinary skill in the art would have had a reasonable expectation of success in making the proposed combination. For example, Victorian and Beasley share a very similar hardware and software configuration, that is, operably coupled microphone(s) and speaker(s) to a processor to conduct signal processing. Moreover, Victorian already discloses processor 460 including "a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110." (Victorian (PA-E), ¶[0030]).

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

247. To the extent the preamble is limiting, Victorian discloses or suggests this feature. Victorian discloses an ear-level communication system 100 comprising an ear-level device 110. (Victorian (PA-E), ¶[0018].) Ear-level device 110 is

depicted in the user's ear in FIG. 1 of Victorian, which also provides that this device can be implemented in an "in-the-ear" or "in-the-canal" configuration as shown in Figures 3A and 3B. (*Id.*, ¶[0024]-[0025].) The ear-level device 110 can be used as an ear piece:

> 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.

(*Id.*, ¶[0008]; *see also id.*, Abstract.)

248. Accordingly, Victorian discloses or suggests an ear-level device that can be inserted in-the-ear or in-the-canal and can be used as an ear piece ("an earphone").

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;"

1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;"

249. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses ear-level device 110 including an "external microphone

420 [that] picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not 'isolated' when communicating to remote device 190." (Victorian (PA-E), ¶[0030].) Victorian also discloses ear-level device 110 including an "[e]ar canal microphone 440 detects sound from the ear canal." (*Id.*)

250. As such, a person having ordinary skill in the art would have understood Victorian to disclose or suggest an external microphone 420 configured to pick up ambient sounds ("a first microphone configured to measure an ambient acoustic environment"), and an ear canal microphone 440 configured to detect sound from the ear canal ("a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone").

251. To the extent Patent Owner argues that Victorian's external microphone 420 capturing sound from the ambient environment does not encompass "*measur[ing]* an ambient acoustic environment," such an argument would be incorrect because a person having ordinary skill in the art would have understood that ambient sound captured by external microphone 420 necessarily comprises a measure of the sound reaching that microphone.

252. Additionally, Victorian discloses an ambient sound gating module 421 that "compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound," and "[w]hen the amplitude of the detected ambient

sound exceeds the threshold for the ambient sound, … ambient sound gating module 421 is turned on." (Victorian (PA-E), ¶[0033].)

253. Accordingly, a person having ordinary skill in the art would have understood Victorian discloses or suggests obtaining an amplitude of ambient sound reaching external microphone 420 and that this amplitude comprises a measure of that sound at least because it can be compared to a threshold.

254. A person having ordinary skill in the art would have understood that sound from the ear canal captured by ear canal microphone 440 necessarily comprises a measure of the sound reaching that microphone. Additionally, Victorian discloses an occluded sound gating module 441 that compares the amplitude of a detected occluded sound with a predetermined threshold for the occluded sound, and that "when the amplitude of the detected occluded sound exceeds the threshold for the occluded sound, … occluded sound gating module 441 is turned on." (Victorian (PA-E), ¶¶[0034], [0031].)

255. Accordingly, a person having ordinary skill in the art would have understood Victorian discloses or suggests obtaining an amplitude of ear canal sound reaching ear canal microphone 440 and that this amplitude comprises a measure of that sound at least because it can be compared to a threshold.

256. Although Victorian describes an external microphone 420 and ear canal microphone 440 capturing sounds from the ambient environment and from inside

the ear canal, respectively, it does not specifically describe the openings through which sound travels to reach external microphone 420 and ear canal microphone 440. To the extent such openings are not inherent, it would have been obvious to a person having ordinary skill in the art to modify Victorian based on the teachings of Dijkstra to dispose external microphone 420 and ear canal microphone 440 within ear-level device 110 with a sound channel providing fluid communication between external microphone 420 and air external to ear-level device 110 and between ear canal microphone 440 and air inside the ear canal, for the same reasons provided above in Section X.C.1.

257. A person having ordinary skill in the art would have considered modifying Victorian based on Dijkstra to be predictable and would have had a reasonable expectation of success in disposing Victorian's external microphone 420 and ear canal microphone 440 within ear-level device 110 based on the teachings of Dijkstra. Indeed, a person having ordinary skill in the art would have readily understood that the teachings of Dijkstra could be readily applied to a device already having substantially the same structure, such as Victorian's ear-level device. A person having ordinary skill in the art would have thus found it obvious to incorporate Dijkstra's teachings into Victorian because it would have amounted to merely applying a known technique (Dijkstra's disposition of internal and external microphones within housing) to improve a similar device (the Victorian ear-level

device) in the same way (to protect the microphones from physical damage and shield them from influences that would impact performance).

258. Accordingly, the Victorian-Dijkstra device discloses or suggests an external microphone 420 configured to pick up sound ambient external sounds ("a first microphone configured to measure an ambient acoustic environment"), wherein external microphone 420 is disposed within ear-level device 110 and in fluid communication with air external to ear-level device 110 via a channel ("wherein the first microphone has a first microphone port") to allow external microphone 420 to pick up sound ambient external sounds ("that is configured to face away from a user when the earphone is inserted"), as well as an ear canal microphone 440 configured to detect sound from the ear canal ("a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone"), wherein ear canal microphone 440 is disposed within ear-level device 110 and in fluid communication with air inside the ear canal via a channel ("wherein the second microphone has a second microphone port") to allow ear canal microphone 440 to detect sounds in the ear canal ("that is configured to face away from a user when the earphone is inserted").

c. 1[c] "a speaker configured to play an audio signal;"

259. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses that ear-level device 110 includes an "ear canal speaker

470" disposed in the ear canal that "transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal." (Victorian (PA-E), ¶[0030], [0051].)

260. Accordingly, the Victorian-Dijkstra device discloses or suggests an ear canal speaker 470 ("a speaker") that transmits sound received from a remote device 190 to the user's ear canal ("configured to play an audio signal").

d. 1[d] "a memory that stores instructions; and"

1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:"

261. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses ear-level device 110 including "a voice operated exchange (VOX) circuit 430" and "a processor 460." (Victorian (PA-E), ¶[0030].) Victorian's FIG. 4A shows processor 460 coupled to ear canal speaker 470 and, through VOX circuit 430, to external microphone 420 and ear canal microphone 440.

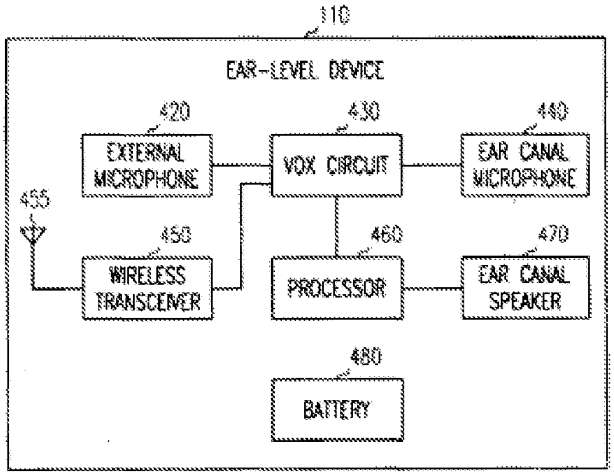


FIG. 4A

Victorian, FIG. 4A

262. A person having ordinary skill in the art would have found it obvious to implement Victorian's processor 460 using a digital signal processor because such digital processors were widely available and commonly used to perform signal processing. For example, Svean discloses using "a processing unit E3 that may be a digital signal processor (DSP), a microprocessor (µP) or a combination of both" to process signals from various microphones. (Svean (PA-I), 6:35-51).[15] In fact, a

[15] Here, I am relying on Svean (PA-I) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 4.

person having ordinary skill in the art would be motivated to use a digital signal processor (DSP) when in need of processing a signal because a DSP "has all of the same basic characteristics and components; a CPU, memory, instruction set, buses, etc.;" as a microprocessor, but "each of these components is customized slightly to perform certain operations more efficiently," therefore DSPs "consume less time, energy and power than a general-purpose microprocessor when carrying out signal processing tasks." (Oshana, 12).

263. A person having ordinary skill in the art would have known that a digital processor would operate by executing instructions and that these instructions would be stored in a memory. At the very least, it would have been obvious in view of the knowledge of a person having ordinary skill in the art that a digital processor would execute instructions and that these instructions would be stored in a memory, as this configuration was a well-known and common way of implementing a digital processor-based system to perform operations. For example, Svean discloses that "[t]he processing unit E3 is connected to memory elements RAM (Random access memory) E8, ROM (read only memory) E9, and EEPROM (electrically erasable programmable read only memory) E10," where these memory elements are "used for storing computer programs, filter coefficients, analysis data and other relevant data," all of which are necessary for proper operation of the processor. (Svean (PA-

I), 6:52-58).[16] And whether a DSP or other processor, the memory is an integral component. *See* Oshana, 12 (explaining that a DSP "has all of the same basic characteristics and components; a CPU, memory, instruction set, buses, etc.;" as a microprocessor but is "customized slightly to perform certain operations more efficiently.").

264. Accordingly, the Victorian-Dijkstra device discloses or suggests a processor 460 with a memory that stores instructions executed by the processor ("a memory that stores instructions; and a processor that is configured to execute the instructions to perform operations") where the processor is coupled to ear canal speaker 470 ("the speaker is coupled to the processor") and, through VOX circuit 430, to external microphone 420 and ear canal microphone 440 ("the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone"). Victorian discloses or suggests processor 460 performing the claimed operations, as discussed below.

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[16] Here, I am relying on Svean (PA-I) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 4.

e. 1[f] "receiving a first microphone signal from the first microphone;"

265. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses that processor 460 "converts the signals received from … external microphone 420 to a sound audible to the person wearing ear-level device 110." (Victorian (PA-E), ¶[0030].)

266. Accordingly, the Victorian-Dijkstra device discloses or suggests a processor configured to receive sound from an external microphone ("receiving a first microphone signal from the first microphone") and convert the sound into a sound audible to the user.

f. 1[g] "receiving a second microphone signal from the second microphone;"

267. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses processor 460 converting "the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190." (Victorian (PA-E), ¶[0030].)

268. Accordingly, the Victorian-Dijkstra device discloses or suggests a processor configured to receive sound from an ear canal microphone ("receiving a second microphone signal from the second microphone") and convert the sound into a sound to be transmitted to a remote device.

g. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;"

269. Victorian in combination with Dijkstra discloses or suggests this feature. Victorian discloses that "[p]rocessor 460" "converts" the signals received from external microphone 420 to a sound audible to the user and that the processor includes "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." (PA-E, ¶[0030].) Victorian also discloses that processor 460 includes a "speech recognition module" that "enhances" the converted ambient sound signal to "improve[] the signal-to-noise ratio." (*Id.*, ¶¶[0030], [0052].)

270. As explained above in Section X.B.2.f, a person having ordinary skill in the art would have understood that the broadest reasonable interpretation of the step of "generating an ambient sound signal" includes any manipulation of the signal received from the ambient sound microphone.

271. A person having ordinary skill in the art would have understood that Victorian's disclosure relating to converting the ambient sound signal (e.g., with amplification) or enhancing the signal with an increased signal-to-noise ratio would require manipulating the signal.

272. Thus, a person having ordinary skill in the art would have understood that "converting" and "enhancing" signals received from an external microphone received from an external microphone "to a sound audible to the person wearing the ear-level device" entails generating a signal from the external microphone signal to be played by the ear canal speaker 470. A person having ordinary skill in the art would also have understood the signal generated to be an ambient sound signal because it has been generated from the ambient sound received from the external microphone 420.

273. Accordingly, the Victorian-Dijkstra device discloses or suggests a processor configured to convert and enhance a signal from an external microphone to a sound audible to the user ("generating an ambient sound signal from at least one of the first microphone signal …").

h. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

274. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. Beasley discloses implementing an automatic audio bypass module utilizing a digital signal processor (DSP). (Beasley (PA-F), ¶[0054].) An embodiment utilizing DSP technology is depicted in Beasley's FIG. 10, which is reproduced and annotated below:

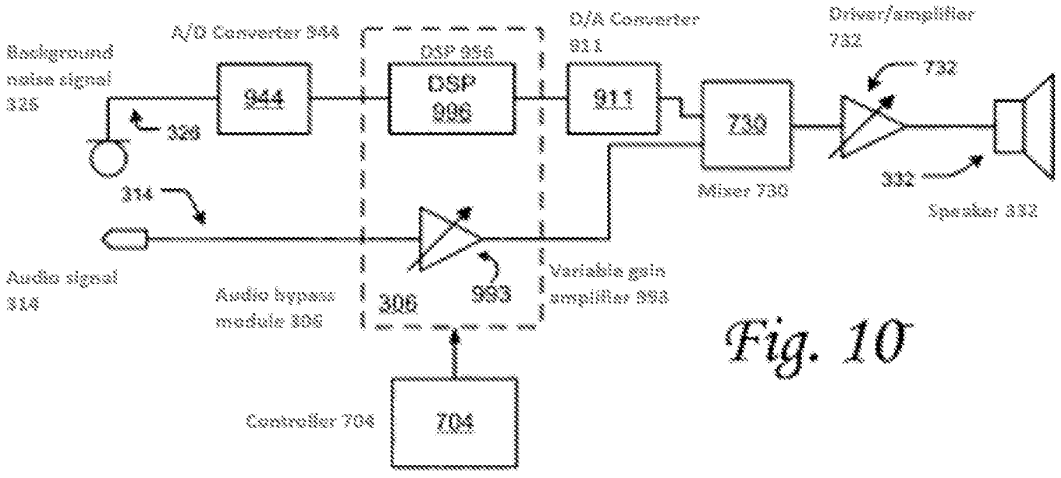


Fig. 10

Beasley, FIG. 10 (annotated)

275. With respect to at least this embodiment, Beasley discloses digitizing a background noise signal and providing the digitized signal to DSP 996, which is utilized to "enhance the background audio signal such that the conversation or other audio that the user desires to hear in the bypass mode is better heard via speaker 332." (*Id.*) A person having ordinary skill in the art would have understood Beasley's disclosure of "enhancing" the signal so that it can be "better heard" to encompass amplifying the signal. (*See id.*, ¶[0031] (describing enhancing a signal using "an amplifier such as pre-amp 202"); ¶[0051] (disclosing that "the gain of amplifiers 704 or amplifier 732 (or both) can be increased in the audio bypass mode to allow the background audio signal to be amplified to a greater extent so that it can be heard by the user"); ¶[0052] ("Additionally, circuitry components or instrumentality can be

utilized in signal paths 744A and 744B to enhance or amplify signals falling within the frequency range of normal human speech. … Thus, in this embodiment, amplifying the range of frequencies covered by human speech can enhance the ability of the user to hear a conversation").) Beasley also discloses utilizing a digital signal processor to amplify the speech signal. (Beasley (PA-F), ¶[0054] ("Once digitized, these signals can be processed utilizing a digital signal processor to, for example, amplify the speech signal, diminish the background noise, and inhibit the audio signal 314 from passing through to speaker 332.").) A person having ordinary skill in the art would have understood utilizing a digital signal processor to amplify the speech signal to comprise DSP 996 amplifying background noise signal 326.

276. A person having ordinary skill in the art would thus have understood that Beasley discloses or suggests amplifying the ambient sound so that it can be better heard by the user. A person having ordinary skill in the art would also have understood that amplifying the ambient sound signal by Victorian's processor 460, based on the teachings in Beasley, would comprise applying an ambient sound gain. Indeed, Beasley discloses adjusting "the gain of amplifiers" so that the background audio signal can be "amplified to a greater extent so that it can be heard by the user." (Beasley (PA-F), ¶[0051].) A person having ordinary skill in the art would also further understood that a widely known and common technique for amplifying an audio signal in the digital domain was to adjust (e.g., increase or decrease) its gain.

277. Accordingly, the Victorian-Dijkstra-Beasley device discloses or suggests processor 460 amplifying the ambient sound signal by applying an ambient sound gain ("applying an ambient sound gain to the ambient sound signal") to produce an amplified signal that can be heard by the user to a greater extent ("to generate a modified ambient sound signal").

i. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

278. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. For example, Victorian discloses a gating rule that allows the transmission of both the remote sound and the ambient sound to the ear canal. (Victorian (PA-E), ¶[0038] ("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 ….").) The gating rule determines which sounds reach the processor 460, but a person having ordinary skill in the art would have understood that Victorian's processor 460 mixes the ambient sound signal from the external microphone 420 and the audio content from the remote device 190 to provide a mixed signal to the ear canal speaker 470 for playback. (*Id.*, ¶[0030] ("Processor 460 … 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."). Because Victorian discloses that both the ambient sound and remote sound are provided to the speaker, a person having ordinary skill in the art would have understood that the processor would have to mix the two signals.

279. For the reasons discussed in Section X.E.1, it would have been obvious to a person having ordinary skill in the art to modify Victorian in view of Beasley to implement Beasley's automatic bypass mode, which would include processor 460 amplifying the background audio signal so that it can be better heard by the user, and then mix this enhanced background audio signal with the audio content from the remote device for playback by the ear canal speaker 470.

280. Accordingly, a person having ordinary skill in the art would have understood the combined Victorian-Dijkstra-Beasley device to disclose or suggest processor 460 mixing the enhanced ambient sound signal from external microphone 420 ("mixing the modified ambient sound signal") with audio content from remote device 190 ("with an audio content signal") to generate a combined output signal ("to generate a mixed audio signal").

j. 1[k] "sending the mixed audio signal to the speaker."

281. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. For example, Victorian discloses a gating rule that allows the transmission of both the remote sound and the ambient sound to the ear canal. (Victorian (PA-E), ¶[0038] ("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 ….").) As discussed in Section X.E.2.i, a person having ordinary skill in the art would have understood that Victorian's processor 460 mixes the ambient sound signal from the external microphone 420 and the audio content from the remote device 190 to provide a mixed signal to the ear canal speaker 470 for playback.

282. Accordingly, a person having ordinary skill in the art would have understood the combined Victorian-Dijkstra-Beasley device to disclose or suggest providing the mixed signal comprising the enhanced ambient sound signal from external microphone 420 and audio content from remote device 190 to ear canal speaker 470 ("sending the mixed audio signal to the speaker") for playback to the user.

3. Claim 2

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and"

283. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. Beasley describes an "automatic mode" in which "the audio bypass mode is controlled automatically based on the occurrence of one or more

events." Beasley describes using one of various techniques to detect the occurrence of the one or more events, including voice or speech recognition and background audio level detection. (Beasley (PA-F), ¶¶[0037], [0061].) Beasley further explains that "voice or speech recognition techniques can be utilized to allow the headset to enter the audio bypass mode in response to a command that is spoken by the user." (*Id.*, ¶[0061].) And with respect to "background audio level detection," Beasley explains that "the device may be programmed to enter automatically the audio bypass mode when a background audio level is sensed that is louder or at a higher level than the ambient background noise or, for example, when a particular threshold is reached." (*Id.*) As an example, Beasley explains that, "when the device senses that the user has spoken (as words spoken by the user may be sensed by the device at a level louder than the background noise), the device could automatically enter the audio bypass mode." *Id.*, ¶[0061].

284. A person having ordinary skill in the art would have understood Beasley to disclose using several techniques to detect events that automatically initiate the audio bypass mode—e.g., at least voice or speech recognition and background audio level detection. A person having ordinary skill in the art would also have understood that each of these techniques is configured to detect acute sounds—i.e., the presence of a specific word or phrase or a sound exceeding a threshold. In the combined Victorian-Dijkstra-Beasley device, a person having

ordinary skill in the art would have understood that one or more of these techniques would be performed on signals picked up by Victorian's external microphone 420.

285. This is consistent with how the '666 patent describes this feature. The '666 patent does not define the term "acute sounds," but it does identify abrupt high energy sounds such as warning sounds (e.g., bell, security system), sirens (e.g., police car), voice (e.g., 'help', 'stop', 'police', etc.), or specific noise types (e.g., breaking glass) as examples of acute sounds.

286. Accordingly, a person having ordinary skill in the art would have understood the combined Victorian-Dijkstra-Beasley device to disclose or suggest detecting a specific word or phrase or a sound exceeding a threshold ("detecting an acute sound") by performing voice or speech recognition or background audio level detection on signals picked up by Victorian's external microphone 420 ("by analyzing … the first microphone signal ….").

b. 2[b] "determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals."

287. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. As discussed in Section X.E.3.a, Beasley discloses using voice or speech recognition or background audio level detection to detect the occurrence of one or more events that are programmed to automatically initiate audio bypass

mode. A person having ordinary skill in the art would have understood that each of the events detected by these techniques comprise the user's voice. For example, to utilize voice or speech recognition, Beasley explains that "the user can program the device to enter the audio bypass mode when a particular word or phrase is spoken." *Id.*, ¶[0061]. And when utilizing background audio level detection, Beasley explains that the device is programmed to automatically enter audio bypass mode "when the device senses that the user has spoken (as words spoken by the user may be sensed by the device at a level louder than the background noise)." (*Id.*) As noted in Section X.E.3.a, a person having ordinary skill in the art would have understood that, in the combined Victorian-Dijkstra-Beasley device, one or more of these techniques would be performed on signals picked up by Victorian's external microphone 420.

288. Accordingly, a person having ordinary skill in the art would have understood the Victorian-Dijkstra-Beasley device to disclose or suggest determining whether a sound detected is the user's voice by detecting a particular word or phrase or determining that an audio level is louder than the background noise ("determining whether the acute sound is a user's voice") by performing voice or speech recognition or background audio level detection on signals picked up by Victorian's external microphone 420 ("by analyzing … the first microphone signal ….").

289. To the extent Patent Owner asserts or the Office finds that Beasley determining whether a sound detected is the user's voice (e.g., by detecting a

particular word or phrase or determining that an audio level is louder than the background noise) does not disclose or suggest both the "detecting an acute sound" and "determining whether the acute sound is a user's voice" limitations, it is my opinion that a person having ordinary skill in the art would have nonetheless found each of these features to be obvious in view of the teachings of Beasley. For example, Beasley describes using speech recognition techniques to respond to a phrase "Could you repeat that?", thus "enabling the audio bypass mode when the user asks another person to repeat what they have just said." (Beasley (PA-F), ¶[0061].) However, a person having ordinary skill in the art would have understood that performing speech recognition alone would not distinguish the headset wearer uttering the phrase "Could you repeat that?" from another person uttering the phrase. As such, a person having ordinary skill in the art would have found it obvious to perform speech recognition to detect a specific word or phrase and then use background audio level detection to determine that the user spoke the word or phrase. Indeed, Beasley's claims contemplate using both speech recognition and background audio level threshold detection. (Beasley (PA-F), claims 19 ("[W]herein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection." (Beasley (PA-F), claims 19 ("[W]herein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection."

290. Accordingly, a person having ordinary skill in the art would have understood the Victorian-Dijkstra-Beasley device to disclose or suggest detecting a specific word or phrase ("detecting an acute sound") by performing speech recognition on signals picked up by Victorian's external microphone 420 ("by analyzing … the first microphone signal ….") and determining whether the specific word or phrase was spoken by the headset wearer by determining that an audio level is louder than the background noise ("determining whether the acute sound is a user's voice") by performing background audio level detection on signals picked up by Victorian's external microphone 420 ("by analyzing … the first microphone signal ….").

4. Claim 8

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected."

291. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. As discussed in Section X.E.3, Beasley discloses entering audio bypass mode when the user's voice is detected. Beasley also discloses that, "when the audio bypass mode is initiated, the gain of variable gain amplifier 993 can be decreased to minimize the level of audio signal 314." (Beasley (PA-F), ¶[0054].) In the same mode, Beasley explains that "the DSP 996 can be utilized to actually enhance the background audio signal such that the conversation or other audio that

the user desires to hear in the bypass mode is better heard via speaker 332." (*Id.*) As discussed in Section X.E.2.f, a person having ordinary skill in the art would have understood Beasley to disclose that "enhancing" the signal so that it can be "better heard" includes amplification of the signal. As discussed in Section X.E.2.i, Beasley discloses or at a minimum renders obvious implementing the functionality of variable gain amplifier 993 in software executed by DSP 996.

292. Accordingly, a person having ordinary skill in the art would have understood the Victorian-Dijkstra-Beasley device to disclose or suggest decreasing the gain of audio content from remote device 190 and amplifying the ambient sound signal ("adjusting the mixed audio signal") when audio bypass mode is initiated in response to the user's voice being detected ("when the user's voice is detected").

5. Claim 9

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal."

293. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. Indeed, for the same reasons discussed above in Section X.E.4, a person having ordinary skill in the art would have understood the Victorian-Dijkstra-Beasley device to disclose or suggest decreasing the gain of audio content from remote device 190 ("where the operation of adjusting the mixed audio signal

is to reduce the audio content signal") and amplifying the ambient sound signal ("and increase the ambient sound signal") when audio bypass mode is initiated in response to the user's voice being detected.

6. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

294. Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. Victorian discloses embodiments in which an earmold is used with the ear-level device. (Victorian (PA-E), Abstract.) "In one embodiment, the earmold is fully occluded to maximize audio efficiency." (*Id.*, ¶[0027].) The earmold has a sealing section, as depicted, for example, in FIGSs. 3A, 3B, and 3C of Victorian, reproduced below:

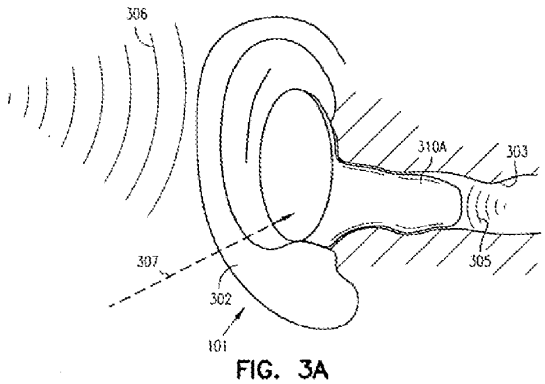


FIG. 3A

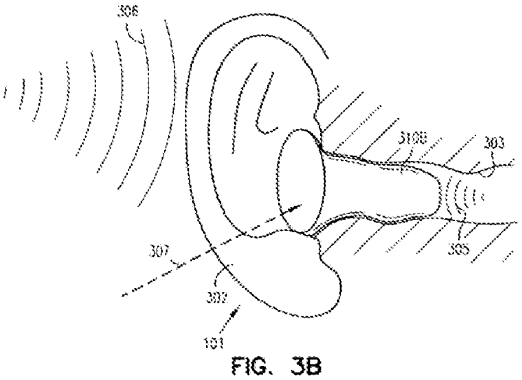


FIG. 3B

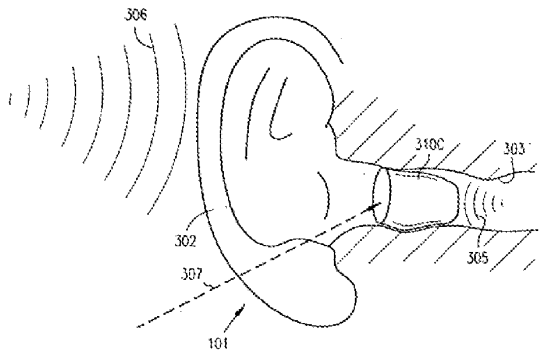


FIG. 3C

Victorian, FIGs. 3A-C

295. These embodiments include the same "ear-level device 110" discussed above with respect to claim 1. (*See id.*, [0029].) A person having ordinary skill in the art would have understood that an ear-level device having "a custom earmold" and designed to "fully occlude" the ear (*id.*, [0027]), would have been capable of achieving noise reduction in the range of up to 22-26dB, which is the example of "sealing" provided in the '666 patent (PAT-A., 6:47-54.). At a minimum a person having ordinary skill in the art would have found it obvious to configure the ear-level device and its earmold to achieve noise reduction of at least 22-26 dB to achieve Victorian's goal of "full occlusion."

296. Accordingly, a person having ordinary skill in the art would have understood the Victorian-Dijkstra-Beasley to disclose or suggest an ear-level device having an earmold designed to fully occlude a user's ear canal to maximize audio efficiency ("the earphone includes a sealing section").

F. Ground 5: Claim 10 is Obvious in View of Victorian in Combination with Dijkstra, Beasley, and Andrea

297. It is my opinion that that Victorian in combination with Dijkstra, Beasley, and Andrea would have rendered obvious claim 10 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. Claim 10

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal."

298. Victorian in combination with Dijkstra, Beasley, and Andrea discloses or suggests this feature. Victorian discloses that processor 460 of ear-level device 100 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. (Victorian (PA-E), ¶[0030].) For example, Victorian discloses processor 460 including "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." (*Id.*) Neither Victorian, Beasley, nor Dijkstra expressly disclose, however, "where the mixed audio signal includes a noise reduction signal derived from the second microphone signal." Andrea, however, when combined with Victorian, Beasley, and Dijkstra, would have rendered this limitation obvious.

299. Andrea is directed to an adaptive noise cancellation (ANC) and speech enhancement system. (Andrea (PA-G), Title, 5:29-34). Andrea discloses "a headset apparatus for use in an intercommunications system," such as an intercom, telephone, or radio. (*Id.*, Abstract, 5:29-34.) Specifically, Andrea discloses an in-ear headset 40 that includes "a speaker 50, a reference sensor or microphone 60 which is adapted to detect external background signals, and a feedback/voice transmission sensor or microphone 70 which are mounted in a single ear piece 42." (*Id.*, 6:45-49.) Andrea explains that microphone 60 is "adapted to detect external background signals" such as "sirens, engine noises and crowd noises." (*Id.*, 6:46-47, 7:11-15.) Andrea's FIG. 4 depicting headset 40 is reproduced below:

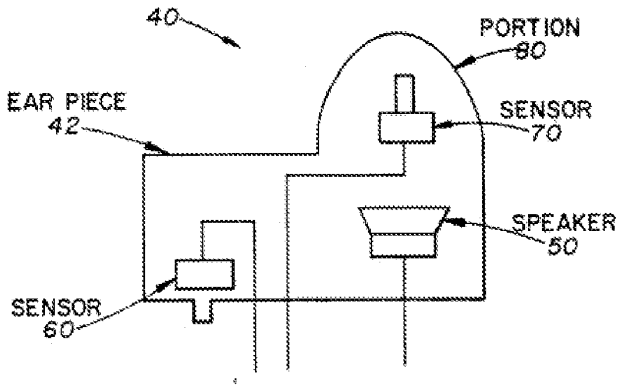
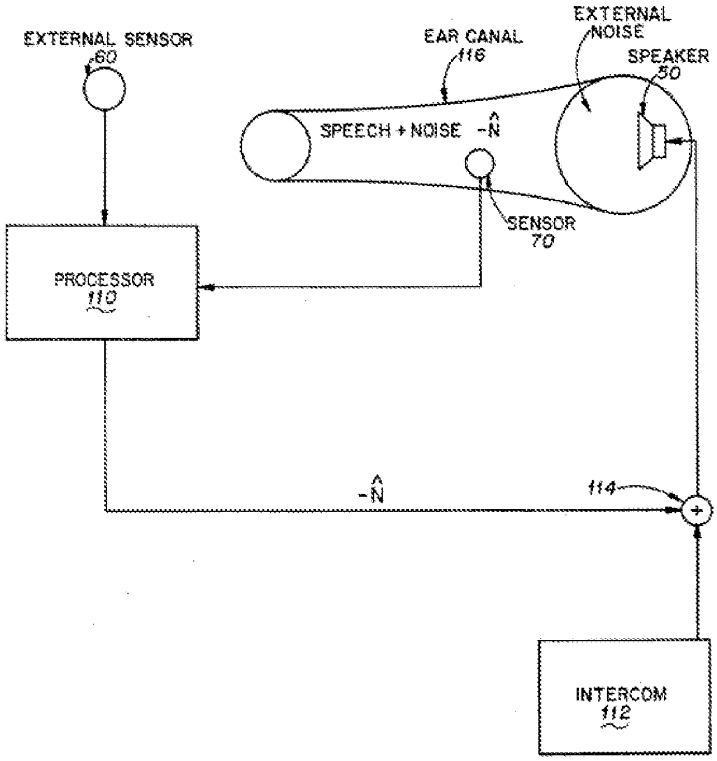


Fig. 4

Andrea, FIG. 4

300. Andrea further explains that, in the ANC headphone depicted in FIG. 4, microphone 70 comprises "an in ear sensor 70 acoustically isolated with the use of plastic tubing," speaker 50 is configured "to play the antinoise and speech," and microphone 60 (or "external sensor 60") is configured to sense ambient acoustic noise. (*Id.*, 14:35-39.)

301. Andrea discloses headset 40 operating in receive and transmit modes. (*Id.*, 7:8-9.) In "receive mode," Andrea discloses that "external background or reference acoustic sounds (which may be referred to as noise), for example, sirens, engine noises and crowd noises, are picked-up by the sensor 60 and supplied to a processor 110 along with a feedback or error signal from the sensor 70." (*Id.*, 7:10-15.) Andrea's "receive mode" is depicted in FIG. 5, reproduced below. (*See id.*, 4:67-68 ("FIG. 5 illustrates a receive mode of the system according to an embodiment of the invention; ….").



Andrea, FIG. 5

302. Andrea discloses processor 110 supplying the received signals—i.e., the sound picked up by sensor 60 and a feedback or error signal from sensor 70—through a finite impulse response (FIR) filter. (*Id.*, 7:15-22.) The resulting output "represent[ing] anti-noise" is "supplied to a combining circuit 114 along with a signal from an intercom 112." (*Id.*, 7:15-25.) Andrea further discloses that "[t]he combined signal from the combining circuit 114 is supplied to the speaker 50, contained within the headset 40 which is located in the ear canal 116 of the operator, so as to be heard by the operator." (*Id.*, 7:25-29.)

303. A person having ordinary skill in the art would have understood the output of Andrea's processor 110 representing anti-noise to comprise a noise reduction signal. A person having ordinary skill in the art would also have understood the output of Andrea's processor 110 generated from a received signal that includes sound picked up by sensor 60 and a feedback or error signal from sensor 70 to comprise an output derived from a signal from sensor 70. A person having ordinary skill in the art would also have understood Andrea's microphone 70 to comprise an internal microphone configured to detect sound from the ear canal. Accordingly, a person having ordinary skill in the art would have understood Andrea to disclose a combined signal from combining circuit 114 that includes an anti-noise signal ("where the mixed audio signal includes a noise reduction signal") generated from a received signal that includes a feedback or error signal from microphone 70 ("derived from the second microphone signal").

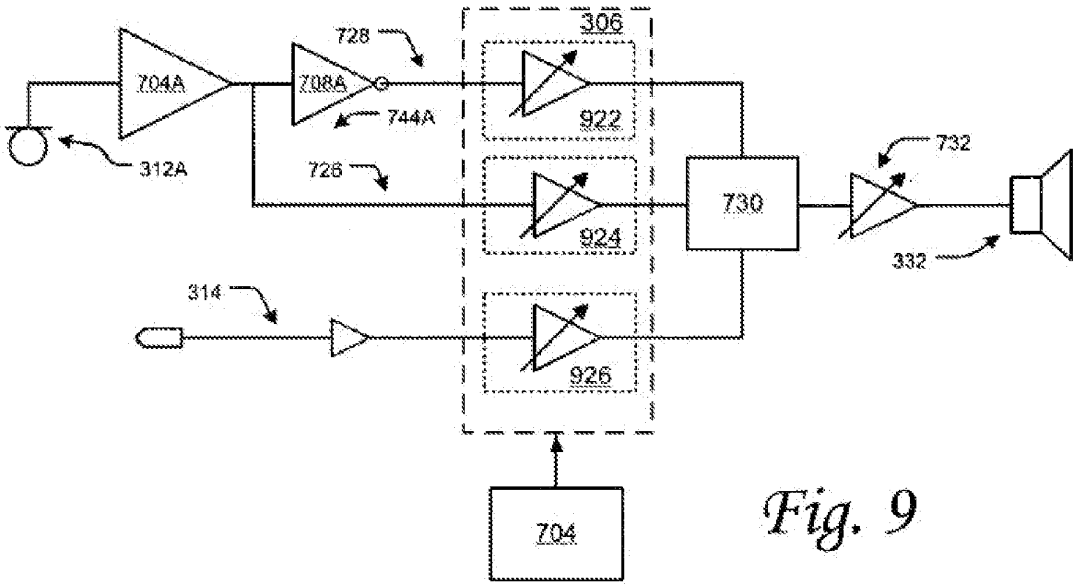
304. In view of Victorian's invitation to apply noise reduction circuitry in processor 460 configured to convert signals received from an external microphone to a sound audible to the person wearing ear-level device 110 (Victorian (PA-E), ¶[0030]), a person having ordinary skill in the art would have been motivated to look for ways to implement such noise reduction circuitry in a similar device having an external microphone for picking up ambient sound and ear canal microphone that detects sound from the ear canal. In so looking, a person having ordinary skill in the

art would have readily found Andrea. Indeed, a person having ordinary skill in the art would have recognized that Andrea provides implementation details for using an ear canal microphone to produce an anti-noise signal that is provided to a user as part of a mixed signal.

305. A person having ordinary skill in the art would have considered the combination of Victorian, Dijkstra, Beasley, and Andrea to be predictable and would have had a reasonable expectation of success in implementing Andrea's noise reduction functionality in Victorian's processor. A person having ordinary skill in the art would have readily understood that the teachings of Andrea could be readily applied to a device already having substantially the same structure (i.e., an external and internal microphone), such as the Victorian-Dijkstra-Beasley device. A person having ordinary skill in the art would also have understood that Andrea describes implementing its noise reduction functionality with digital signal processing technology. (Andrea (PA-G), 5:35-37 ("The system uses advanced digital signal processing technology to implement adaptive noise canceling and a digital voice operated switch (VOX).").)

306. A person having ordinary skill in the art would also have recognized that it was known to selectively provide to a user a noise cancellation signal along with a background audio signal and an audio signal. For example, Beasley describes an embodiment in which a noise cancellation signal 728, a background audio signal

726, and an audio signal 314 are selectively provided to a user. (Beasley (PA-F), ¶[0053].) The foregoing embodiment is depicted in Beasley's FIG. 9, which is reproduced below:



Beasley, FIG. 9

307. Beasley explains that, with the noise cancellation feature turned on, microphone 312A is used to transduce acoustical pressure generated by background noise that a user wishes to hear." (*Id.*, ¶¶[0050], [0055].) The background noise signal is conditioned by an optional driver/pre-amp 704A and inverted via inverter 708A to create a noise cancellation signal 728. (*Id.*, ¶[0050].) Beasley discloses that "controller 704 can be utilized to control the gain of amplifiers 922, 924, and 926 to selectively pass cancellation signal 728, background audio signal 726, or the audio

signal 314." (*Id.*, ¶[0053].) For example, "when the audio bypass mode is initiated, the gain of variable gain amplifiers 922 and 926 is attenuated such that noise cancellation signal 728 and audio signal 314 are attenuated and the gain of variable gain amplifier 924 is increased such that the background audio signal 726 is amplified and readily heard by the user through speaker 332." (*Id.*) Beasley also discloses a mixer 730 receiving noise cancellation signal 728, background audio signal 726, and audio signal 314 and providing the signals as a single mixed signal to speaker 332 via a driver 732. (*Id.*, FIG. 9; *see also id.*, ¶[0049] (describing components 730A and 730B in FIG. 8 as "mixers").)

308. A person having ordinary skill in the art would have understood Beasley to disclose, in audio bypass mode (discussed above, for example, in Section X.E.1), providing a mixed signal that includes a background audio signal, an audio signal, and a noise cancellation signal. Notably, Beasley also discloses that "inverter 708 can be implemented utilizing a number of different techniques to provide an inverse or complementary audio signal utilized to cancel out the background audio noise." (Beasley (PA-F), ¶[0051].) For example, "[t]his can be accomplished using inverters, compensation filters, and other known techniques." (*Id.*) As such, a person having ordinary skill in the art would have understood that the functionality of Beasley's inverter 708 could be implemented instead by Andrea's noise reduction functionality. Thus, a person having ordinary skill in the art would have recognized

that Andrea's noise reduction functionality could be implemented to produce an anti-noise signal that is included in a mixed signal with a background audio signal and an audio signal.

309. Accordingly, when Beasley's "audio bypass mode" is implemented in the combined Victorian-Dijkstra-Beasley device, a person having ordinary skill in the art would have considered it predictable to modify Victorian's VOX circuit 430 and processor 460 to implement Andrea's noise reduction functionality to produce a noise-cancellation signal based on the signal from Victorian's external microphone 420 and provide that noise-cancellation signal in the mixed background and audio content signal, and would have had a reasonable expectation of success in doing so.

G. Ground 6: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Van Doorn and Beasley

310. As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and person having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground addresses the circumstance in which Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in a microphone package itself. Under such circumstances, it is my opinion that Victorian in combination with Van Doorn and

Beasley would have rendered obvious claims 1-2, 8-9, and 11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. Independent Claim 1

311. With respect to 1[pre] and 1[c]-1[k], Victorian in combination with Beasley meets the limitations in the same way discussed above in Section X.E, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section X.E.2.a.
1[c]	*See supra* Section X.E.2.c.
1[d]	*See supra* Section X.E.2.d.
1[e]	*See supra* Section X.E.2.d.
1[f]	*See supra* Section X.E.2.e.
1[g]	*See supra* Section X.E.2.f.
1[h]	*See supra* Section X.E.2.g.
1[i]	*See supra* Section X.E.2.h.
1[j]	*See supra* Section X.E.2.i.
1[k]	*See supra* Section X.E.2.j.

312. With respect to 1[a] and 1[b], Victorian discloses or suggests "a first microphone configured to measure an ambient acoustic environment" and "a second

microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone" in the same way as discussed above in Section X.E.2.b. Additionally, for the reasons discussed in Section X.D.1 a person having ordinary skill in the art would have understood Van Doorn's microphone having a sound port 56 to disclose a "microphone port."

313. A person having ordinary skill in the art would have found it obvious to modify Victorian's ear-level device 110 to use Van Doorn's microphone 110 with a sound inlet plate 50 comprising sound port 56 for external microphone 420 and ear canal microphone 440. Indeed, a person having ordinary skill in the art would have been motivated to use Van Doorn's microphone 110 because it provides "a more compact design" as compared with other prior art microphones (Van Doorn (PA-D), ¶¶[0006], [0023]), and because sound port 56 can be offset from aperture 128 to ease installation of external microphone 420 and ear canal microphone 440 in Victorian's ear-level device 110 (*Id.*, ¶¶[0006], [0024]).

314. A person having ordinary skill in the art would also have recognized Van Doorn's microphone 110 was suitable for inclusion Victorian's ear-level device as Van Doorn discloses that its sound inlet plate 50 is designed for use on miniature microphones in hearing aids. Indeed, Van Doorn explains that its microphone package will "ease installation of the microphone in the hearing aid." Van Doorn, ¶[0006].

315. A person having ordinary skill in the art would have thus found it obvious to incorporate Van Doorn's microphone 110 into Victorian's ear-level device because it would have amounted to merely a simple substation of one known element (Van Doorn's microphone 110) for another (Victorian's external microphone 420 and ear canal microphone 440) to obtain predictable results (an ear-level device with external and internal microphones having a more compact design).

316. Accordingly, a person having ordinary skill in the art would have understood the Victorian-Van Doorn-Beasley device to disclose or suggest an external microphone having a sound port as taught by Van Doorn ("wherein the first microphone has a first microphone port") that is configured to detect sounds outside the ear ("that is configured to face away from a user when the earphone is inserted"), and an ear canal microphone having a sound port as taught by Van Doorn ("wherein the second microphone has a second microphone port") that is configured to detect sounds within the ear canal ("that is configured to face toward the user when the earphone is inserted").

2. Dependent Claims 2, 8-9, and 11

317. With respect to dependent claims 2, 8-9, and 11, Victorian in combination with Beasley meets the limitations in the same way discussed above in Section X.E, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section X.E.3.
8	*See supra* Section X.E.4.
9	*See supra* Section X.E.5.
11	*See supra* Section X.E.6.

318. Accordingly, for at least the same reasons, these claims would have been obvious over Victorian, Van Doorn, and Beasley.

H. Ground 7: Claim 10 is Obvious in View of Victorian in Combination with Van Doorn, Beasley, and Andrea

319. It is my opinion that Victorian in combination with Van Doorn, Beasley, and Andrea would have rendered obvious claim 10 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. Claim 10

a. "10. The earphone according to claim 9, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal."

320. It would have been obvious for a person having ordinary skill in the art to incorporate Andrea with the Victorian-Van Doorn-Beasley device for the same reasons and in the same way as discussed above in Section X.F. Accordingly, for at least the same reasons, claim 10 would have been obvious over Victorian, Van Doorn, Beasley, and Andrea.

I. Ground 8: Claims 1 and 11 are Obvious in view of Dijkstra in combination with Boersma

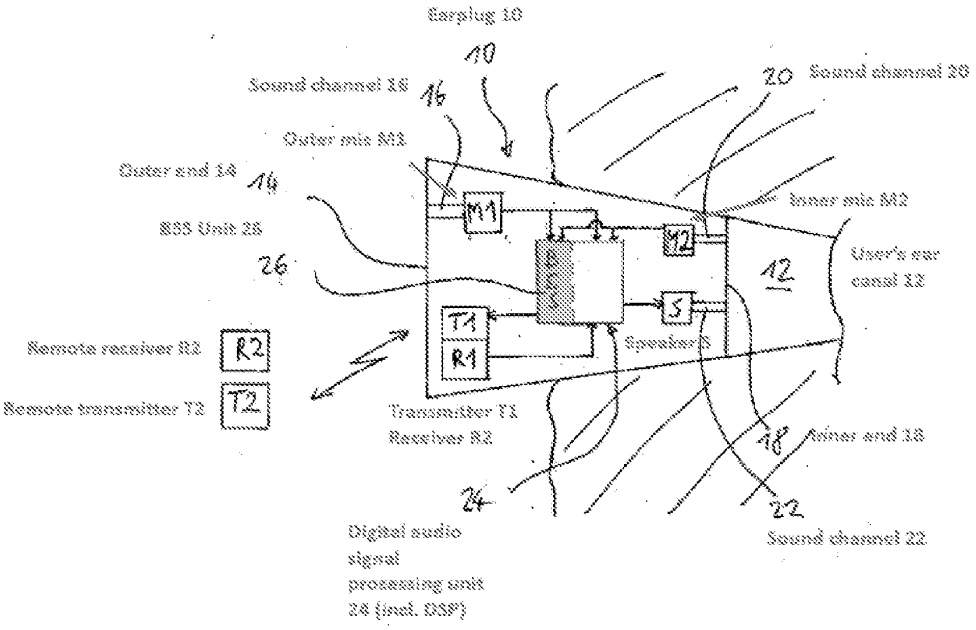
321. As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and person having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground addresses the circumstance in which Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in the housing of a device in which the microphone is mounted. Under such circumstances, it is my opinion that Dijkstra in combination with Boersma would have rendered obvious claims 1 and 11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. A person having ordinary skill in the art Would Have Been Motivated to Combine Dijkstra and Boersma

322. As explained further below, a person having ordinary skill in the art would have been motivated to combine the teachings of Dijkstra and Boersma and would have had a reasonable expectation of success in doing so.

323. Dijkstra discloses a "hearing protection device" comprising an earplug inserted into the user's ear and that implements a blind source separation ("BSS") algorithm to obtain an intelligible speech signal for use in wireless communications. (Dijkstra (PA-C), ¶¶[0010], [0013].) The earplug includes an outer microphone M1,

an inner microphone M2, and an audio signal processing unit 24 including a BSS unit 26. (*Id.*, ¶[0029.) The outer microphone M1 is open to the outer end 14 of the earplug via a sound channel 16 and primarily picks up ambient sound. (*Id.*, ¶¶[0029], [0031].) The inner microphone M2 is open to the inner end 18 of the earplug via sound channel 20 and primarily picks up the user's voice. (*Id.*) The audio signal processing unit includes a digital signal processor and program and data memory components. (*Id.*, ¶[0029]). The processing unit also includes a transceiver and receiver for communication with a mobile telephone. (*Id.*, ¶¶[0030], [0032], [0033].)



Dijkstra, FIG. 1 (annotated)

324. The signals from both microphones are passed to the BSS unit 26. (*Id.*) The output signal of the BSS unit 26 is transmitted to another user with a similar

earplug or a mobile telephone. (*Id.*, ¶[0032].) Having undergone BSS, the output signal contains the user's voice separated from background noise, resulting in "good intelligibility of speech." (*Id.*, ¶¶[0007]-[0008]; *see also id.*, ¶¶[0009]-[0012].) "Such a system could be used by any persons who need to communicate in a noisy environment, such as workers, soldiers, firemen, etc." (*Id.*, ¶[0032].)

325. Dijkstra also discloses passing the signals from both microphones to the audio signal processing unit 24 for other audio signal processing besides BSS. (*Id.*, ¶[0034].) For example, the ambient sound picked up by microphone M1 may be provided to the speaker "for enabling perception of desired ambient sound during times when the noise level is low so that the hearing protection function provided by the earplug 10 is not needed." (*Id.*)

326. Boersma relates to "a headset for electronic equipment" such as media players or mobile phones. (PA-H, 1:6-7, 4:38-56.) Boersma identifies several disadvantages of conventional headsets, including that they "can completely block out all other sounds from the environment" which "can be undesirable or even dangerous." (*Id.*, 1:37-46.) Boersma notes that prior efforts have been made to reproduce ambient sounds in a headset (*id.*, 1:51-56), but those attempts had drawbacks, including that they did not provide sufficient control to the user. (*Id.*, 1:57-67.)

327. To address those drawbacks, Boersma discloses a headset 30 that provides a user with the ability to listen to reproduced audio yet still be able to hear ambient sound, including "the ability to control the respective levels of the reproduced audio and ambient sound, thereby allowing the user to adjust the listening experience to his or her preference." (*Id.*, 5:51-65.) The device can be used in a variety of environments with different levels of noise, for example in traffic, a crowded train, or in one's living room. (*Id.*, 7:36-46.)

328. In one embodiment, the headset includes a pair of earpieces 38. (*Id.*, 6:14-16.) Each earpiece includes a microphone 50 "for picking up ambient sound" (*id.*, 6:32-33), and a loudspeaker 56 (*id.*, 6:50-51).

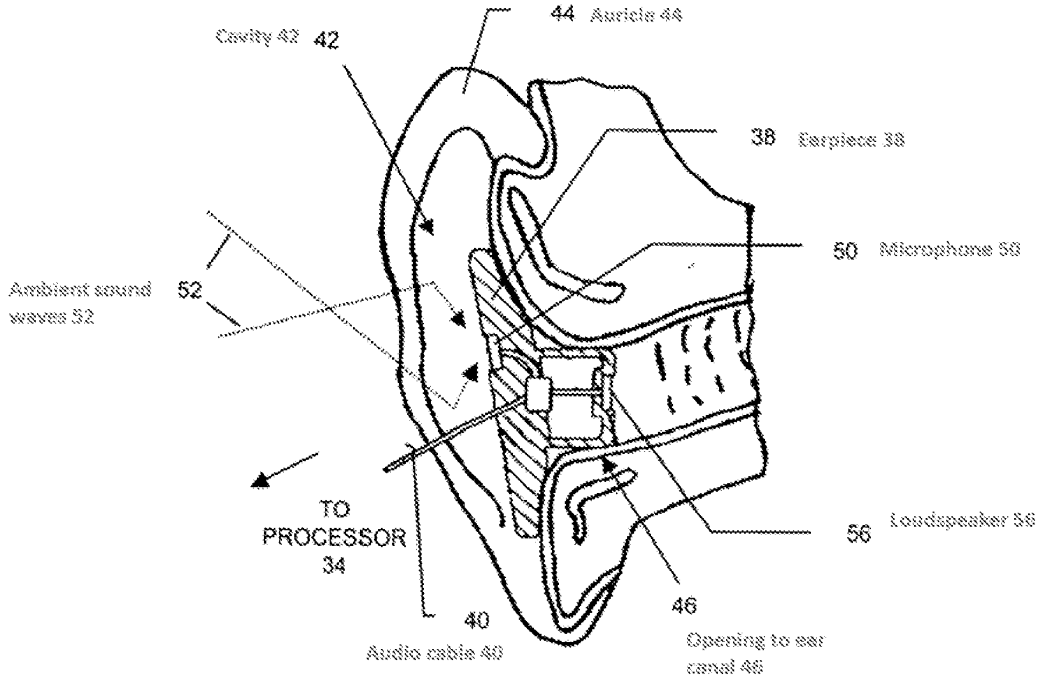


FIG. 2

Boersma, FIG. 2 (annotated)

329. The headset also includes a processor 34. (*Id.*, 6:13). The processor 34 "functions to combine the audio signal received from the mobile phone 10 with the ambient sound signal provided by the respective microphones 50." (*Id.*, 6:58-62). "The processor 34 then provides the combined signals to the loudspeaker 56 in each earpiece 38." (*Id.*, 6:62-63). The processor also has a microphone 102 "for picking up the voice of the user during hands-free communication with the mobile phone." (*Id*, 8:63-67.) Boersma discloses implementing the processor 34 with "digital circuitry" using a "digital design" (*id.*, 7:54-61) and implementing the processor 34

"within the housing of the earpiece(s) 38)" or "within a small housing designed to fit behind the ear of the user," (*id.*, 9:13-18).

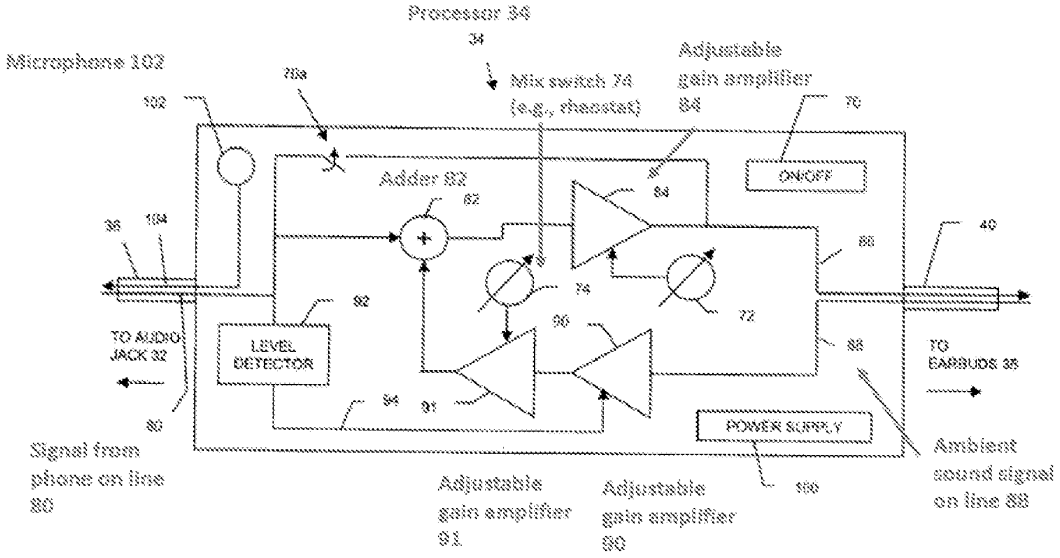


FIG. 4

Boersma, FIG. 4 (annotated)

330. A person having ordinary skill in the art would have recognized that Boersma's and Dijkstra's teachings complement each other in a manner that would have motivated a person having ordinary skill in the art to implement the functionality of Boersma's processor 34 within Dijkstra's digital audio signal processing unit 24 and would have had a reasonable expectation of success in doing so.

331. For example, a person having ordinary skill in the art would have understood that implementing Boersma's technique for providing user control over

the amount of ambient sound reproduced at the user's ear canal would enhance and extend the usability of Dijkstra's earplug. Dijkstra discloses providing ambient sound picked up by the outer microphone M1 to the user, and a person having ordinary skill in the art would have understood that implementing Boersma's controllable mixing functionality would have provided the user additional control without negatively impacting Dijkstra's hearing protection objective. A person having ordinary skill in the art also would have understood that adding Boersma's technique to Dijkstra's earpiece would have made it safer to use. (*Id.*). For example, it was well-known that noise-attenuating headsets can be dangerous for the user. Boersma itself explains that headsets which block out sound can be dangerous. (PA-H, 1:43-50.)

332. Accordingly, a person having ordinary skill in the art would have understood the proposed combination to amount to applying a known technique (Boersma's user-controlled ambient sound mixing technique) to a known device (Dijkstra's ear plug) ready for improvement to yield predictable results (an earplug with controllable ambient sound mixing).

333. There is ample evidence that a person having ordinary skill in the art would have had a reasonable expectation of success in making the proposed combination. For example, Boersma itself suggests implementing processor 34 with digital circuitry in a digital design. (PA-H, 7:47-61.) Moreover, it was well-known

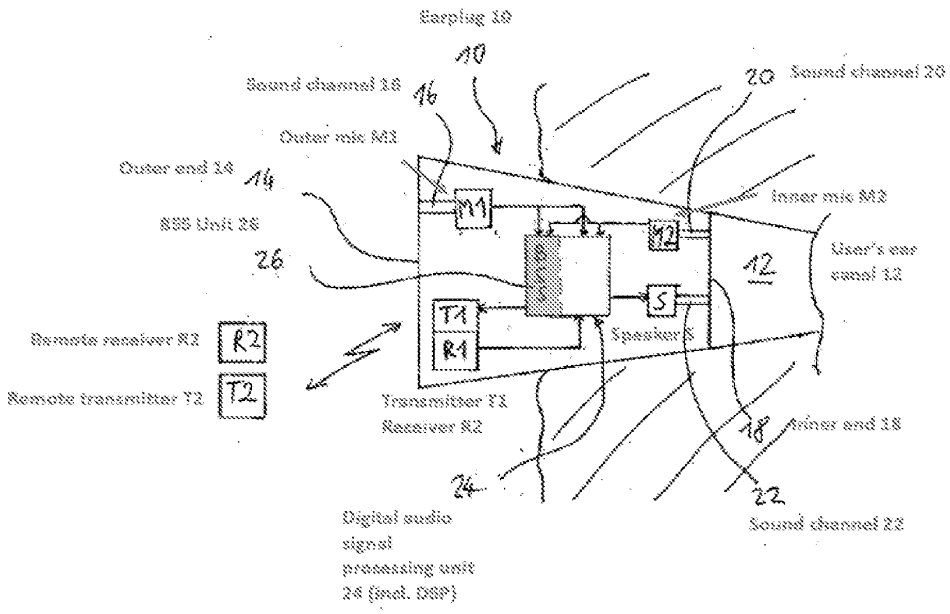
to use DSPs for mixing audio signals with ambient sound. (discussing U.S. Patent Application Publication No. 2003/0035551A1 ("Light")).[17] Furthermore, Boersma discloses that a person having ordinary skill in the art would have readily appreciate that mix switch 74 and volume switch 72 with devices to program Dijkstra's DSP implement the functionality of those switches.

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

334. To the extent the preamble is limiting, Dijkstra discloses or suggests an earphone. For example, Dijkstra discloses earplug 10 that is inserted into the user's ear. (Dijkstra (PA-C), ¶¶[0013], [0026]-[0030], [0032], [0034]-[0043], FIGs. 1-6.

[17] Here, I am relying on Light (PA-S) as evidence of a person having ordinary skill in the art's knowledge and the state of the art; it is not part of the combination in Ground 8.



Dijkstra, FIG. 1 (annotated)

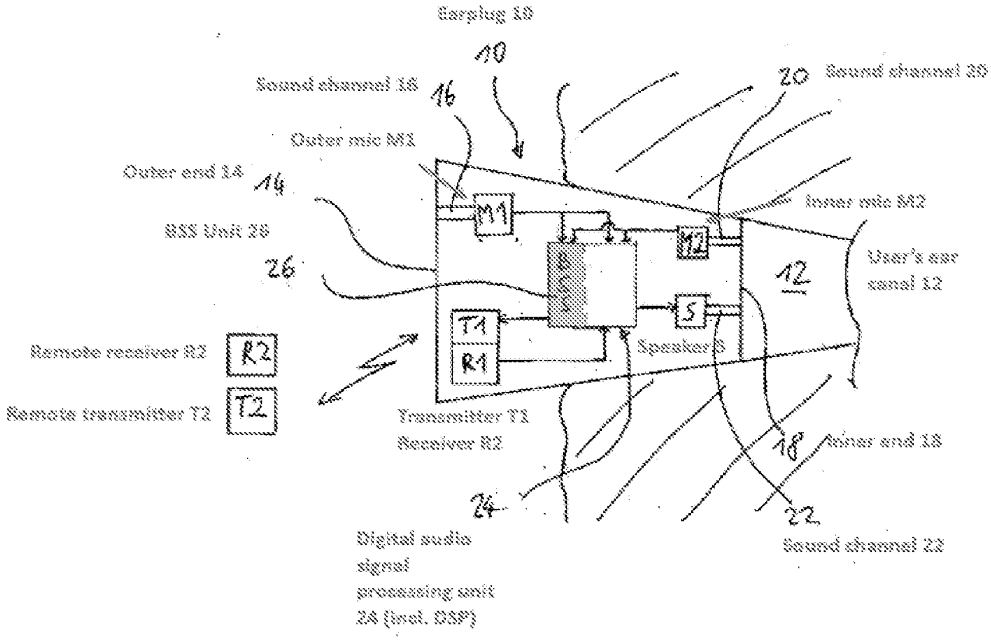
335. A person having ordinary skill in the art would have understood Dijkstra's earplug 10 to comprise "[a]n earphone" at least because it is configured to be inserted at least part way into the ear of a person and because it includes a speaker 38 configured to transmit sounds to the individual. Dijkstra's earplug 10 is also of the same general construction as the "earphone" (or "earpiece") described in the '666 patent. (PAT-A, FIG. 1, 4:52-65.)

336. Thus, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest an earplug ("an earphone").

b. 1[a] "a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;"

1[b] "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;"

337. Dijkstra discloses or suggests these features. For example, Dijkstra describes earplug 10 having an outer microphone M1 oriented outwardly toward the environment and an inner microphone M2 oriented inwardly towards the user's ear canal. (*Id.*, Abstract, ¶¶[0010], [0013]-[0015], [0029], [0031], [0034].) Figure 1, which depicts earplug 10 having outer microphone M1 and inner microphone M2, is reproduced below:



Dijkstra, FIG. 1 (annotated)

338. Dijkstra discloses that outer microphone M1 "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." (*Id.*, ¶[0029].) Dijkstra also discloses that inner microphone M2 "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." (*Id.*)

339. A person having ordinary skill in the art would have understood Dijkstra's outer microphone M1 which is open to the outer end 14 of the earplug 10 via sound channel 16 and inner microphone M2 which is open to the inner end of the earplug 10 via a sound channel 20 to each include the claimed "microphone ports" for the same reasons discussed above in Section X.C.1.

340. Accordingly, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest an outer microphone M1 ("a first microphone configured to measure an ambient acoustic environment") which is open to the outer end 14 of the earplug 10 via a sound channel 16 ("wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted"). Furthermore, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest an inner microphone M2 ("a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone,") which is open to the inner end 18 of the earplug 10 via sound channel 20 ("wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted").

c. 1[c] "a speaker configured to play an audio signal;"

341. Dijkstra discloses or suggests this feature. Dijkstra discloses that the earplug includes a speaker to provide an external or remote audio signal, for example, from a mobile telephone, to the user to provide bidirectional communication. (Dijkstra (PA-C), ¶¶[0017], [0029]-[0030], [0032], [0034].) A person having ordinary skill in the art would have understood that a remote audio signal is an "audio signal." This is consistent with how the specification describes an "audio signal". (PAT-A, 12:26-30 ("The method 900 also receives as input an

Audio Content signal 902, e.g. a music audio signal from a portable Media Player or mobile-phone."), FIG. 9 (step 902).)

342. Accordingly, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest a speaker to provide a remote audio signal from a mobile telephone ("a speaker configured to play an audio signal.")

d. 1[d] "a memory that stores instructions; and"

343. Dijkstra discloses or suggests this feature. Dijkstra discloses "[i]t is evident to the skilled person that the earplug 10 includes additional electrical components" including "program and data memory components for the audio signal processing unit 24." (Dijkstra (PA-C), ¶[0029].) A person having ordinary skill in the art would have understood that a "program" refers to software instructions.

344. Accordingly, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest that the digital signal processor includes a program and data memory component ("a memory that stores instructions").

e. 1[e] "a processor that is configured to execute the instructions to perform operations, wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone, wherein the speaker is coupled to the processor, and the operations comprising:"

345. Dijkstra discloses or suggests this feature. Dijkstra discloses that the earplug 10 includes an audio signal processing unit including a digital signal

processor that has a program memory component. (Dijkstra (PA-C), ¶[0029].) A person having ordinary skill in the art would have understood that a processor is a component specifically designed to execute instructions stored in memory. Dijkstra explains that the audio signal processing unit receives signals from microphones M1 and M2 and provides them to the speaker. (Dijkstra (PA-C), ¶[0034].) This is also schematically illustrated in FIG. 1, which shows arrows from the microphones into the processing unit and from the processing unit to the speaker. (*Id.*, FIG. 1.)

346. Accordingly, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest an audio signal processing unit that executes instructions stored in a program memory component ("processor that is configured to execute the instructions to perform operations") wherein the audio signal processing unit receives signals from the inner microphone and outer microphone ("wherein the processor is coupled to the first microphone, wherein the processor is coupled to the second microphone") and wherein a speaker receives signals from the processor ("wherein the speaker is coupled to the processor").

f. 1[f] "receiving a first microphone signal from the first microphone;"

347. Dijkstra discloses or suggests this feature. Dijkstra explains that the audio signal processing unit receives signals from outer microphone M1. (Dijkstra (PA-C), ¶¶[0029], [0031], [0034], FIG. 1.) Thus, a person having ordinary skill in

the art would have understood Dijkstra to disclose or suggest the audio signal processing unit receives a signal from the outer microphone ("receiving a first microphone signal from the first microphone").

g. 1[g] "receiving a second microphone signal from the second microphone;"

348. Dijkstra discloses or suggests this feature. Dijkstra explains that the audio signal processing unit receives signals from inner microphone M2. (Dijkstra (PA-C), ¶¶[0029], [0031], [0034], FIG. 1.) Thus, a person having ordinary skill in the art would have understood Dijkstra to disclose or suggest the audio signal processing unit receives a signal from the inner microphone ("receiving a second microphone signal from the second microphone").

h. 1[h] "generating an ambient sound signal from at least one of the first microphone signal or the second microphone signal or a combination of both signals;"

349. Dijkstra in combination with Boersma discloses or suggests this feature.

350. Boersma discloses that the mix switch 74 provides the user with control over the mix of ambient sound signal and the audio signal. (Boersma (PA-H), 7:30-46, 8:12-26.) Boersma also discloses that the processor includes a level detector 92 that, together with adjustable gain amplifier 92, is used to automatically adjust the level of an ambient sound signal in proportion to changes in the level of the received

audio signal. (*Id.*, 8:33-42.) "In this manner, even if the user adjusts the audio output volume of the mobile phone 10 or other electronic equipment, the mix ratio of the audio signal and the ambient sound signal will remain substantially constant." (*Id.*, 8:42-46.) Thus, the ambient sound signal from the microphone 50 is amplified by a variable gain amplifier 90 in response to a signal from level detector 92 to ensure that the selected ratio of ambient sound to audio will be kept constant. (Boersma (PA-H), 8:33-46.)

351. As reflected in the annotated Figure 4 below, a person having ordinary skill in the art would have understood that applying a gain to the signal received from microphone 50 generates an ambient sound signal.

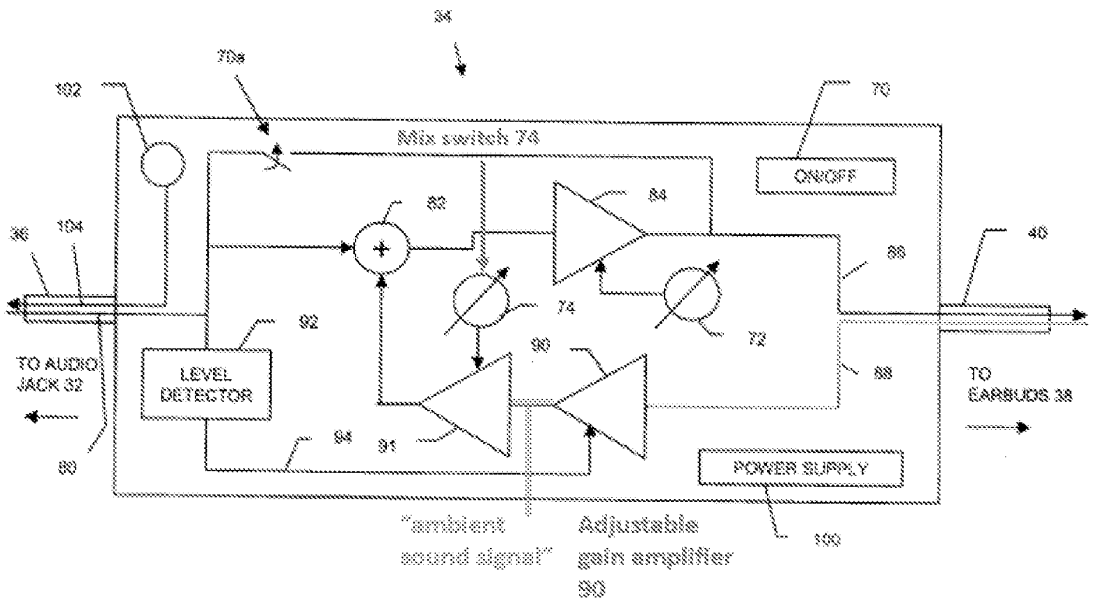


FIG. 4

Boersma, FIG. 4 (annotated)

352. This is consistent with how the '666 patent discloses this feature. As explained above in §X.B.2.f, a person having ordinary skill in the art would have understood that the broadest reasonable interpretation of the step of "generating an ambient sound signal" includes any manipulation of the signal received from the ambient sound microphone. Importantly, this interpretation includes applying a gain to the signal, as the '666 patent expressly describes applying a gain to an ambient sound signal two times, i.e., that "an input signal is captured from the ASM 111 and processed at step 910 (e.g. ADC, EQ, ***gain***)" and then subsequently amplified or attenuated in step 936 and mixed with audio content in step 940. (PAT-A, 12:22-61). Thus, a person having ordinary skill in the art would have understood that applying a gain to a signal from the ambient sound signal encompasses "generating an ambient sound signal."

353. As discussed above in Section X.I.1, in the Dijkstra-Boersma combination, microphone 50 is replaced by Dijkstra's outer microphone M1 and the functionality of Boersma's processor 34 is implemented in a digital design by Dijkstra's audio signal processor. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma device to disclose or suggest an audio signal processor implementing the function of Boersma's adjustable gain amplifier 90 and digitally applying a gain to the signal received from outer

microphone M1 ("generating an ambient sound signal from at least one of the first microphone signal …").

i. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

354. Dijkstra in combination with Boersma discloses or suggests this feature. Boersma discloses a mix switch 74 that controls the adjustable gain amplifier 91 and gives the user control over the "ratio by which the processor 34 combines the audio signal from the mobile telephone 10 and the ambient sound signal from the microphones 50." (*Id.*, 7:30-33, 8:15-18.) The switch gives the user control over whether the earpieces provide more or less ambient sound in relation to an incoming audio signal. (*Id.*, 7:33-36.) The user can (1) block most or all of the environmental sound, (2) retain ambient sound at a lower level, or (3) retain ambient sound at a higher level. (*Id.*, 7:36-46).

355. As reflected in annotated Figure 4 below, a person having ordinary skill in the art would have understood that applying the gain of adjustable gain amplifier 91 results in a modified ambient sound signal. This is consistent with how the '666 patent describes this feature. (PAT-A, 12:22-25, FIG. 9 (step 936).)

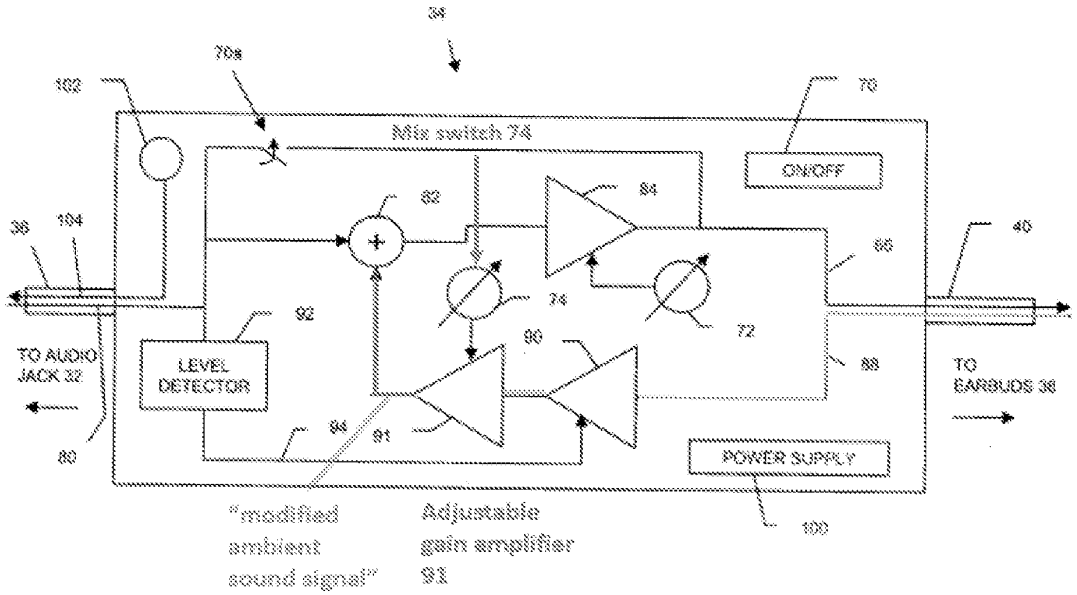


FIG. 4

Boersma, FIG. 4 (annotated)

356. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma device to disclose or suggest a digital signal processor that implements the function of adjustable gain amplifier 91 to generate an ambient sound signal in accordance with the ratio selected by the user ("applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal").

j. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

357. Dijkstra in combination with Boersma discloses or suggests this feature. Boersma discloses that adder 82 "functions to combine the ambient sound signal with the audio signal from the mobile phone 10." (Boersma (PA-H, 8:19-21). As reflected in the annotated FIG. 4 below, a person having ordinary skill in the art would have understood that adder 82 generates a signal that is a mix of the signal from adjustable gain amplifier 91 and the signal from the mobile phone. This is consistent with how the '666 patent describes this feature. (PAT-A, 12:59-61, FIG. 9 (step 940).)

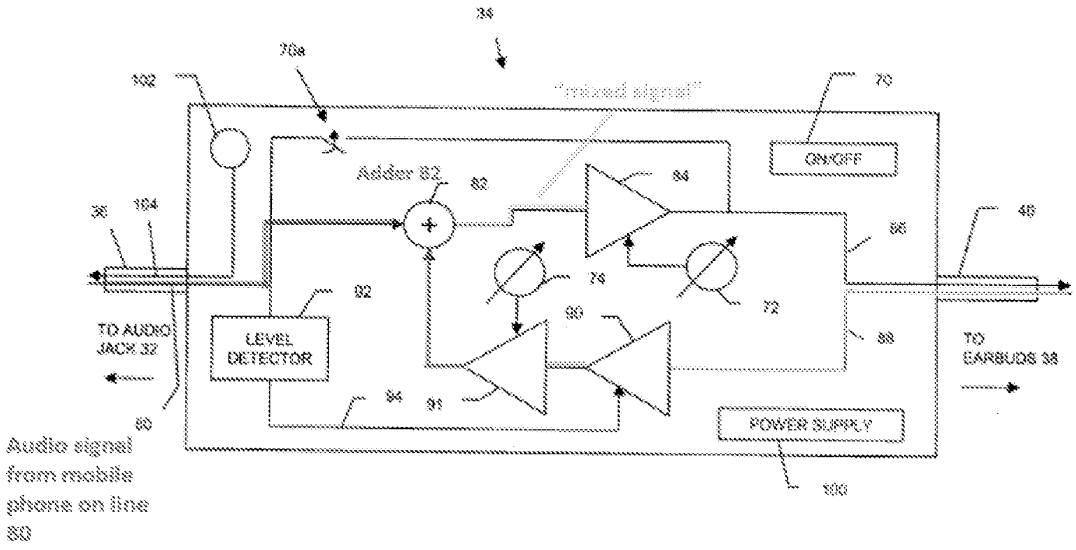


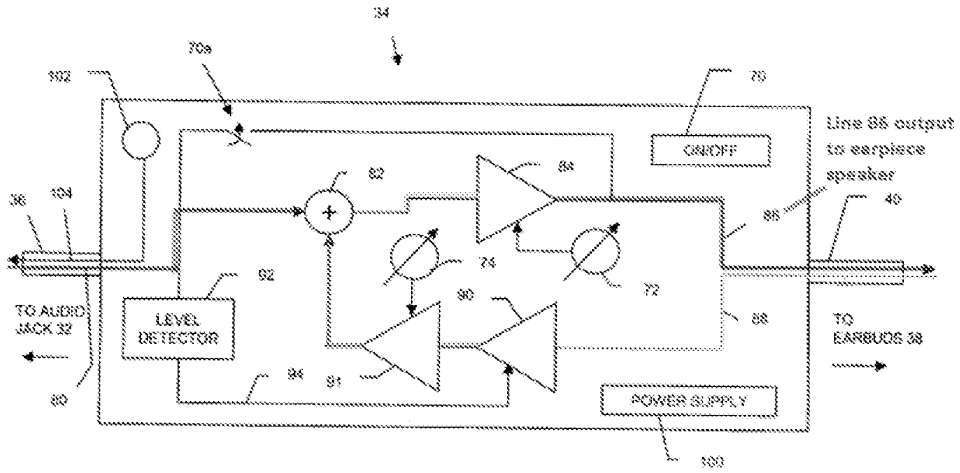
FIG. 4

Boersma, FIG. 4 (annotated)

358. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma device to disclose or suggest a digital signal processor that implements the function of adder 82 to mix the ambient sound signal after it has passed through adjustable gain amplifier 91 ("mixing the modified ambient sound signal") with the audio signal from the mobile phone 10 ("with an audio content signal") to generate a mixed signal ("to generate a mixed audio signal").

k. 1[k] "sending the mixed audio signal to the speaker."

359. Dijkstra in combination with Boersma discloses or suggests this feature. Boersma discloses that the output of the amplifier 84 is provided to the earpieces 38 on line 86 to be output through the speaker. (Boersma (PA-H, 8:6-7, 2:21-26, 2:65-3:5, 6:62-63).



Boersma, FIG. 4 (annotated)

360. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma device to disclose or suggest a digital signal processor that implements the function of providing the mixed signal to Dijkstra's speaker ("sending the mixed audio signal to the speaker").

3. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

361. Dijkstra in combination with Boersma discloses or suggests this feature. Dijkstra discloses a "hearing protection device" that attenuates the ambient sound that reaches the inner microphone. (Dijkstra (PA-C), ¶[0013]). The device comprises an "earplug" comprises "a shell which is to be inserted at least partially into the user's ear canal." (*Id.*) Dijkstra discloses implementing the shell with 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." (*id.*, ¶[0013].) "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." (*Id.*, ¶[0027].) Dijkstra further discloses implementing the shell as a "generic soft shell capable of adapting to the shape of the user's outer ear and ear canal." (*Id.*, ¶[0013].) The '666 patent explains that a "a seal is typically achieved by means of a soft and compliant housing of assembly

113" and that the seal "serves to significantly reduce the sound pressure level at the user's eardrum 133 resulting from the sound field at the entrance to the ear canal." (PAT-A, 5:9-19.)

362. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma device to disclose or suggest a hearing protection earplug that includes a custom hard shell or conforming soft shell that attenuates the ambient sound that reaches the inner microphone ("wherein the earphone includes a sealing section").

J. Ground 9: Claims 2, 8-9 are Obvious in view of Dijkstra in combination with Boersma and Beasley

1. Claim 2

a. 2[a] "2. The earphone according to claim 1, where the operations further comprise detecting an acute sound by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals; and

363. Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. Boersma discloses a mix switch 74 that "allows the user to adjust a ratio by which the processor 34 combines the audio signal from the mobile telephone 10 and the ambient sound signal from the microphones 50. By controlling the mix, the user can choose whether the earpieces 38 provide more or less of the ambient sound in relation to the audio signal." (PA-H, 7:30-33.) Thus, "[d]epending

on the situation," such as "in a crowded train," "while in traffic," or "in one's living room," the user can choose to receive "more or less of the ambient sound in relation to the audio signal." (*Id.*, 7:33-46.)

364. Beasley discloses an audio bypass mode to "to adjust the level of electrical background signal relative to the audio signal" before combining the signals and providing the combined signal to a speaker. (Beasley (PA-F), ¶¶[0010], [0012], [0014], Claims 7, 20.) For example, Beasley discloses that when audio bypass mode is initiated, audio signals may be attenuated and background audio signals may be amplified such that "the signal level, and thus ultimate volume, of background audio signal 312 is increased relative to that of audio signal 314, thus enabling the user to better hear the background audio signal 312 over the audio program content 314 from the audio source of the electronic device." (*Id.*, ¶[0046]; *see also id.*, ¶[0054]).

365. Beasley further discloses automatically engaging audio bypass mode by using one or more of various techniques to detect the occurrence of the one or more events, including voice or speech recognition and background audio level detection. (Beasley (PA-F), ¶¶[0037], [0061], claims 18, 19, 30, 31.)

366. Beasley explains that "voice or speech recognition techniques can be utilized to allow the headset to enter the audio bypass mode in response to a command that is spoken by the user." (*Id.*, ¶[0061].) And with respect to

"background audio level detection," Beasley explains that "the device may be programmed to enter automatically the audio bypass mode when a background audio level is sensed that is louder or at a higher level than the ambient background noise or, for example, when a particular threshold is reached." (*Id.*) As an example, Beasley explains that, "when the device senses that the user has spoken (as words spoken by the user may be sensed by the device at a level louder than the background noise), the device could automatically enter the audio bypass mode." (*Id.*, ¶[0061].)

367. A person having ordinary skill in the art would have understood Beasley to disclose using several techniques to detect events that automatically initiate the audio bypass mode—e.g., at least voice or speech recognition and background audio level detection. A person having ordinary skill in the art would also have understood that each of these techniques is configured to detect acute sounds—i.e., the presence of a specific word or phrase or a sound exceeding a threshold.

368. A person having ordinary skill in the art would have been motivated to combine Beasley's teachings related to automatically activating audio bypass mode in the Dijkstra-Boersma device and would have had a reasonable expectation of success. In the proposed combination, the Dijkstra-Boersma device would implement Beasley's technique to automatically engage an audio bypass mode to adjust the mix provided by Boersma's mix switch 74 in response to Beasley's

automatic speech recognition and/or background audio level detection processing on the signal from Dijkstra's outer or inner microphone. A person having ordinary skill in the art would have understood that such a modification would have improved the Dijkstra-Boersma device because it would have made the Dijkstra-Boersma device more convenient. For example, Beasley's technique would allow the user to automatically adjust the mixed audio signal in response to pre-programmed words or phrases without having to manually operate mix switch 74 to adjust the mix.

369. Thus, a person having ordinary skill in the art would have considered the combination to be no more than applying a known technique (Beasley's automatic speech recognition technique for triggering audio bypass mode) to a known device (the Dijkstra-Boersma device) ready for improvement to yield predictable results (the Dijkstra-Boersma device that adjusts the mix of ambient audio and an audio signal in response to a speech command).

370. There is ample evidence that a person having ordinary skill in the art would have had a reasonable expectation of success in making the combination. For example, Beasley itself discloses implementing the audio bypass feature using a DSP and programming it to use speech recognition. (Beasley (PA-F), ¶¶[0054], [0061].)

371. As noted, in the proposed combination, the Dijkstra-Boersma device would implement Beasley's technique to automatically engage an audio bypass mode to adjust the mix provided by Boersma's mix switch 74 in response to

Beasley's automatic speech recognition and/or background audio level detection processing on the signal from Dijkstra's outer or inner microphone. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Beasley device to disclose or suggest a digital signal processor detecting a speech command and/or background audio level detection ("detecting an acute sound") by processing a signal from Dijkstra's outer microphone M1 or inner microphone M2 ("by analyzing at least one of the first microphone signal or the second microphone signal...")

b. 2[b] determining whether the acute sound is a user's voice by analyzing at least one of the first microphone signal or the second microphone signal or a combination of both signals."

372. As discussed in Section X.J.1.a, Beasley discloses using voice or speech recognition or background audio level detection to detect the occurrence of one or more events that are programmed to automatically initiate audio bypass mode. A person having ordinary skill in the art would have understood that each of the events detected by these techniques comprise the user's voice. For example, to utilize voice or speech recognition, Beasley explains that "the user can program the device to enter the audio bypass mode when a particular word or phrase is spoken." *Id.*, ¶[0061]. And when utilizing background audio level detection, Beasley explains that the device is programmed to automatically enter audio bypass mode "when the

device senses that the user has spoken (as words spoken by the user may be sensed by the device at a level louder than the background noise)." (*Id.*) As noted in Section X.J.1.a, in the combination, the Dijkstra-Boersma device would implement Beasley's technique to automatically engage an audio bypass mode to adjust the mix provided by Boersma's mix switch 74 in response to Beasley's automatic speech recognition and/or background audio level detection processing on the signal from Dijkstra's outer or inner microphone.

373. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Beasley device to disclose or suggest determining whether a sound detected is the user's voice by detecting a particular word or phrase or determining that an audio level is louder than the background noise ("determining whether the acute sound is a user's voice") by processing a signal from Dijkstra's outer microphone M1 or inner microphone M2 ("analyzing at least one of the first microphone signal or the second microphone signal…").

374. To the extent Patent Owner asserts or the Office finds that Beasley determining whether a sound detected is the user's voice (e.g., by detecting a particular word or phrase or determining that an audio level is louder than the background noise) does not disclose or suggest both the "detecting an acute sound" and "determining whether the acute sound is a user's voice" limitations, it is my opinion that a person having ordinary skill in the art would have nonetheless found

each of these features to be obvious in view of the teachings of Beasley. For example, Beasley describes using speech recognition techniques to respond to a phrase "Could you repeat that?", thus "enabling the audio bypass mode when the user asks another person to repeat what they have just said." (Beasley (PA-F), ¶[0061].) However, a person having ordinary skill in the art would have understood that performing speech recognition alone would not distinguish the headset wearer uttering the phrase "Could you repeat that?" from another person uttering the phrase. As such, a person having ordinary skill in the art would have found it obvious to perform speech recognition to detect a specific word or phrase and then use background audio level detection to determine that the user spoke the word or phrase. Indeed, while Beasley's detailed description does not describe performing both speech recognition and background audio level detection, Beasley's clams contemplate using both speech recognition and background audio level threshold detection. (Beasley (PA-F), claims 19 ("[W]herein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection."

375. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Beasley device to disclose or suggest detecting a specific word or phrase ("detecting an acute sound") by processing a signal from Dijkstra's outer microphone M1 or inner microphone M2 ("by analyzing … the first microphone signal ….") and determining whether the specific word or phrase was

spoken by the headset wearer by determining that an audio level is louder than the background noise ("determining whether the acute sound is a user's voice") by performing background audio level detection on signals picked up by Victorian's external microphone 420 ("by analyzing … the first microphone signal ….").

2. Claim 8

a. "8. The earphone according to claim 2, where the operations further comprise: adjusting the mixed audio signal when the user's voice is detected."

376. Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. As explained above in §§ X.J.1.a and X.J.1.b, Beasley discloses an audio bypass mode to "to adjust the level of electrical background signal relative to the audio signal" before combining the signals and providing the combined signal to a speaker, and that Beasley discloses triggering audio bypass mode detecting a particular word or phrase or determining that an audio level is louder than the background noise.

377. As noted, in the proposed combination, the Dijkstra-Boersma device would implement Beasley's technique to automatically engage an audio bypass mode to adjust the mix provided by Boersma's mix switch 74 in response to Beasley's automatic speech recognition and/or background audio level detection processing on the signal from Dijkstra's outer or inner microphone.

378. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Beasley device to disclose or suggest adjusting the level of an ambient sound signal relative to an audio signal ("adjusting the mixed audio signal") when detecting a particular word or phrase or determining that an audio level is louder than the background noise ("when the user's voice is detected").

3. Claim 9

a. "9. The earphone according to claim 8, where the operation of adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal."

379. Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. As explained above in §§ X.J.1.a and X.J.1.b, Beasley discloses an audio bypass mode to "to adjust the level of electrical background signal relative to the audio signal" before combining the signals and providing the combined signal to a speaker. (Beasley (PA-F), ¶¶[0010], [0012], [0014], Claims 7, 20.) For example, Beasley discloses that when audio bypass mode is initiated, audio signals may be attenuated and background audio signals may be amplified such that "the signal level, and thus ultimate volume, of background audio signal 312 is increased relative to that of audio signal 314, thus enabling the user to better hear the background audio signal 312 over the audio program content 314 from the audio source of the electronic device." (*Id.*, ¶[0046]; *see also id.*, ¶[0054]).

380. As noted, in the proposed combination, the Dijkstra-Boersma device would implement Beasley's technique to automatically engage an audio bypass mode to adjust the mix provided by Boersma's mix switch 74 in response to Beasley's automatic speech recognition and/or background audio level detection processing on the signal from Dijkstra's outer or inner microphone.

381. Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Beasley device to disclose or suggest adjusting the level of the modified ambient sound signal and audio signal so that the audio signal was attenuated and the modified ambient sound signal was amplified ("adjusting the mixed audio signal is to reduce the audio content signal and increase the ambient sound signal.")

K. Ground 10: Claims 1 and 11 are Obvious in view of Dijkstra, Boersma, and Van Doorn

382. As discussed in Section X.B.2.b, the '666 patent does not define the term "microphone port," and person having ordinary skill in the art were known to use the term "port"—and "microphone port" specifically—to refer to an opening in the housing of a device in which the microphone is mounted, or to refer to an opening in a microphone package itself. This Ground addresses the circumstance in which Patent Owner argues or the Office finds that the broadest reasonable interpretation of "microphone port" means an opening in a microphone package itself. Under such

circumstances, it is my opinion that Dijkstra in combination with Boersma and Van Doorn would have rendered obvious claims 1 and 11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. Independent Claim 1

383. With respect to 1[pre] and 1[c]-1[k], Dijkstra in combination with Boersma meets the limitations in the same way discussed above in Section X.I, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section X.I.2.a.
1[c]	*See supra* Section X.I.2.c.
1[d]	*See supra* Section X.I.2.d.
1[e]	*See supra* Section X.I.2.e.
1[f]	*See supra* Section X.I.2.f.
1[g]	*See supra* Section X.I.2.g.
1[h]	*See supra* Section X.I.2.h.
1[i]	*See supra* Section X.I.2.i.
1[j]	*See supra* Section X.I.2.j.
1[k]	*See supra* Section X.I.2.k.

384. With respect to 1[a] and 1[b], Dijkstra discloses or suggests "a first microphone configured to measure an ambient acoustic environment" and "a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone" in the same way as discussed above in Section X.I.2.b.

385. Additionally, for the reasons discussed in Section X.D.1 a person having ordinary skill in the art would have understood Van Doorn's microphone having a sound port 56 to disclose a "microphone port."

386. A person having ordinary skill in the art would have found it obvious to modify Dijkstra's earplug 10 to use Van Doorn's microphone 110 with a sound inlet plate 50 comprising sound port 56 for outer microphone M1 and inner microphone M2. Indeed, a person having ordinary skill in the art would have been motivated to use Van Doorn's microphone 110 because it provides "a more compact design" as compared with other prior art microphones (Van Doorn (PA-D), ¶¶[0006], [0023]), and because sound port 56 can be offset from aperture 128 to ease installation of outer microphone M1 and inner microphone M2 in Dijkstra's earplug 10 (*Id.*, ¶¶[0006], [0024]).

387. A person having ordinary skill in the art would also have recognized Van Doorn's microphone 110 was suitable for inclusion Dijkstra's earplug as Van Doorn discloses that its sound inlet plate 50 is designed for use on miniature

microphones in hearing aids. Indeed, Van Doorn explains that its microphone package will "ease installation of the microphone in the hearing aid." Van Doorn, ¶[0006].

388. A person having ordinary skill in the art would have thus found it obvious to incorporate Van Doorn's microphone 110 into Dijkstra's earplug because it would have amounted to merely a simple substitution of one known element (Van Doorn's microphone 110) for another (Dijkstra's outer microphone M1 and inner microphone M2) to obtain predictable results (Dijkstra's earplug with a compact microphone). Accordingly, a person having ordinary skill in the art would have understood the Dijkstra-Boersma-Van Doorn device to disclose or suggest an outer microphone having a sound port as taught by Van Doorn ("wherein the first microphone has a first microphone port") that is configured to detect sounds outside the ear ("that is configured to face away from a user when the earphone is inserted"), and an inner microphone having a sound port as taught by Van Doorn ("wherein the second microphone has a second microphone port") that is configured to detect sounds within the ear canal ("that is configured to face toward the user when the earphone is inserted").

2. Claim 11

389. With respect to dependent claim 11, Dijkstra in combination with Boersma meets it in the same way discussed above in Section X.I.3

390. Accordingly, for at least the same reasons, this claim would have been obvious over Dijkstra, Boersma, and Van Doorn.

L. Ground 11: Claims 2 and 8-9 are Obvious in view of Dijkstra, Boersma, Van Doorn, and Beasley

391. With respect to dependent claims 2 and 8-9, Dijkstra in combination with Boersma and Beasley meets the limitations in the same way discussed above in Section X.J, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section X.J.1.
8	*See supra* Section X.J.2
9	*See supra* Section X.J.3

392. Accordingly, for at least the same reasons, these claims would have been obvious over Dijkstra, Boersma, Van Doorn, and Beasley.

XI. CONCLUSION

393. I declare that all statements made herein of my knowledge are true, and that all statements made on information and belief are believed to be true, and that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Date: 2/24/23

Respectfully submitted,



Nathaniel Polish, Ph.D.

Exhibit PA-L



(12) **United States Patent**
Khenkin

(10) **Patent No.: US 8,351,634 B2**
(45) **Date of Patent: Jan. 8, 2013**

(54) **SIDE-PORTED MEMS MICROPHONE ASSEMBLY**

(75) Inventor: **Aleksey S. Khenkin**, Peterborough, NH (US)

(73) Assignee: **Analog Devices, Inc.**, Norwood, MA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 1007 days.

(21) Appl. No.: **12/323,703**

(22) Filed: **Nov. 26, 2008**

(65) **Prior Publication Data**

US 2010/0128914 A1 May 27, 2010

(51) **Int. Cl.**
H01L 23/495 (2006.01)

(52) **U.S. Cl.** **381/361**; 257/676; 257/415

(58) **Field of Classification Search** 257/676, 257/666, E23.145; 324/686, 658; 438/123, 438/299

See application file for complete search history.

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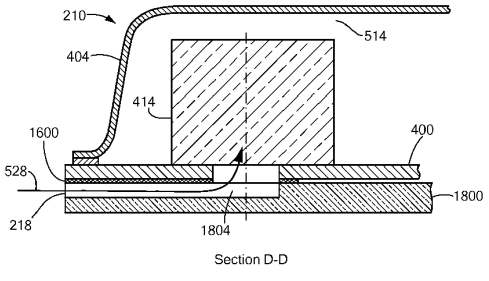
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Primary Examiner — Steven Loke
Assistant Examiner — David Goodwin
(74) *Attorney, Agent, or Firm* — Sunstein Kann Murphy & Timbers LLP

(57) **ABSTRACT**

A side-ported MEMS microphone package defines an acoustic path from a side of the package substrate to a microphone die disposed within a chamber defined by the substrate and a lid attached to the substrate. Optionally or alternatively, a circuit board, to which the microphone package is mounted, may define an acoustic path from an edge of the circuit board to a location under the microphone package, adjacent a bottom port on the microphone package. In either case, the acoustic path may be a hollow passage through at least a portion of the substrate or the circuit board. The passage may be defined by holes, channels, notches, etc. defined in each of several layers of a laminated substrate or circuit board, or the passage may be defined by holes drilled, molded or otherwise formed in a solid or laminated substrate or circuit board.

26 Claims, 18 Drawing Sheets



Section D-D

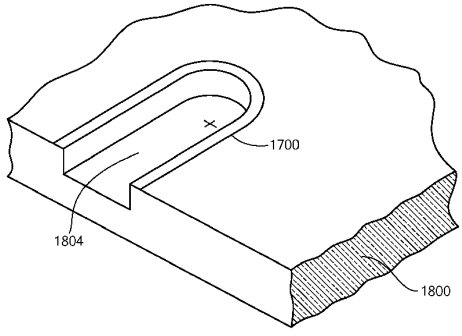


Exhibit PA-L

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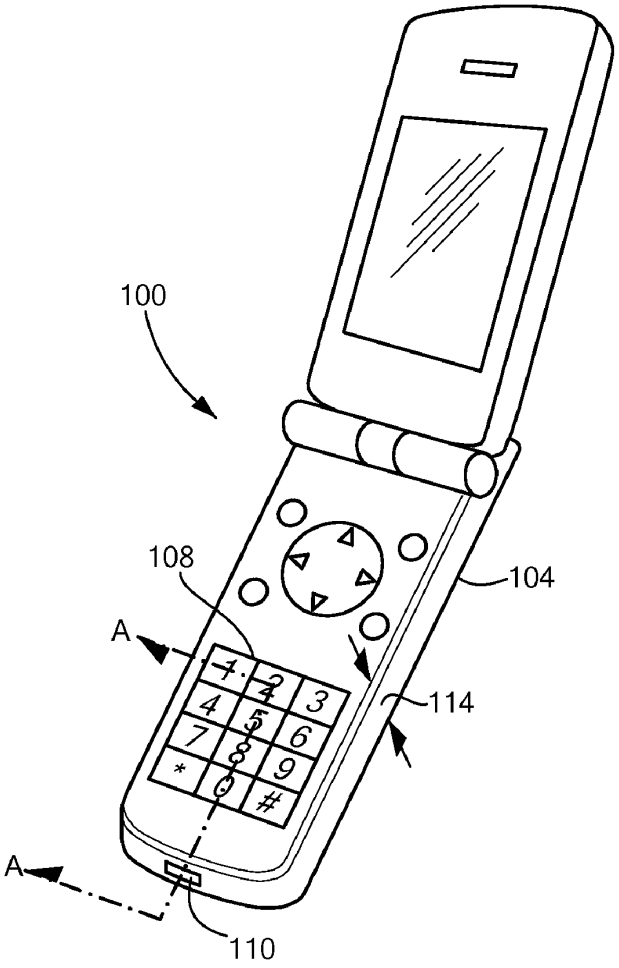
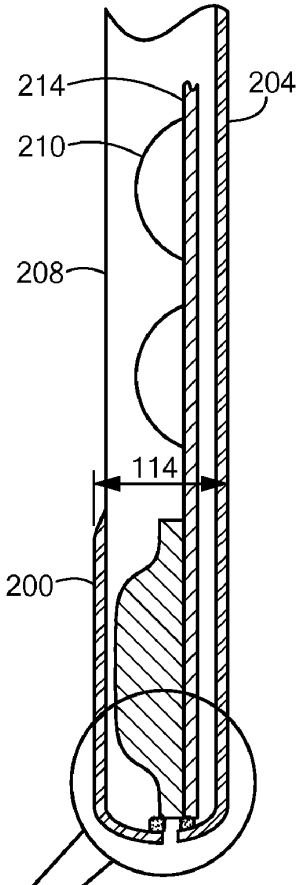
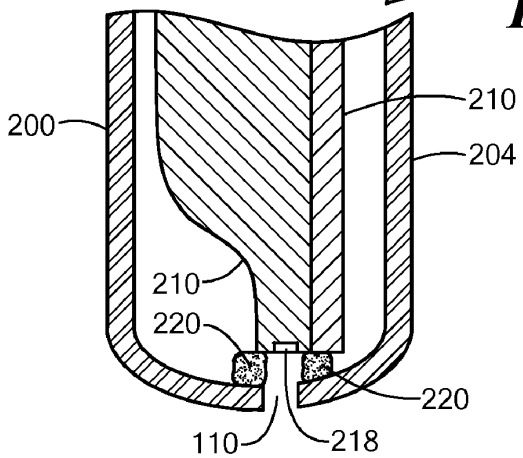


FIG. 1



Section A-A

FIG. 2



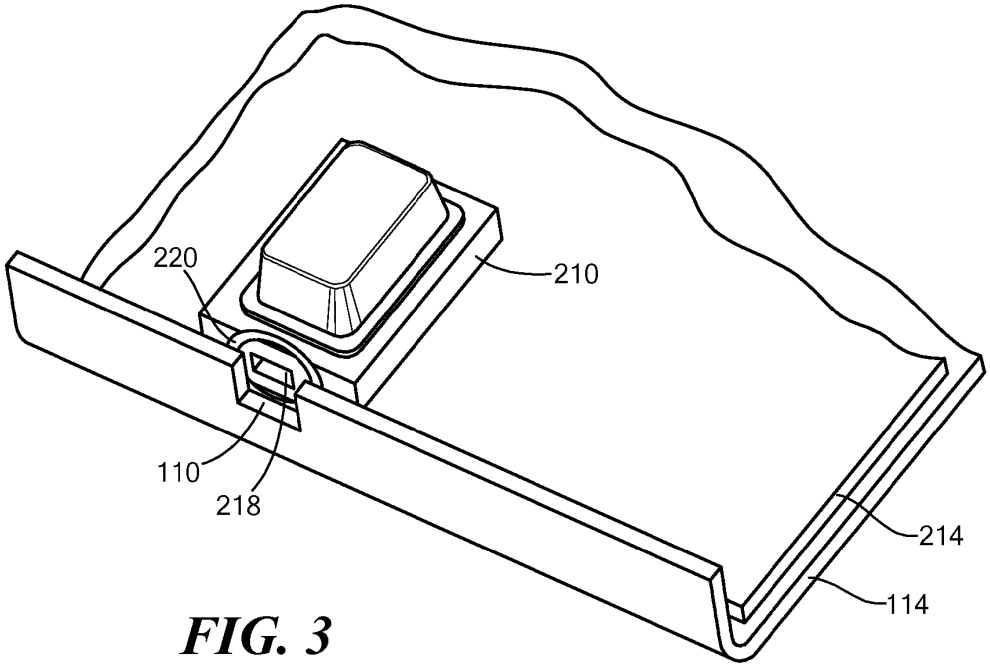

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FIG. 3

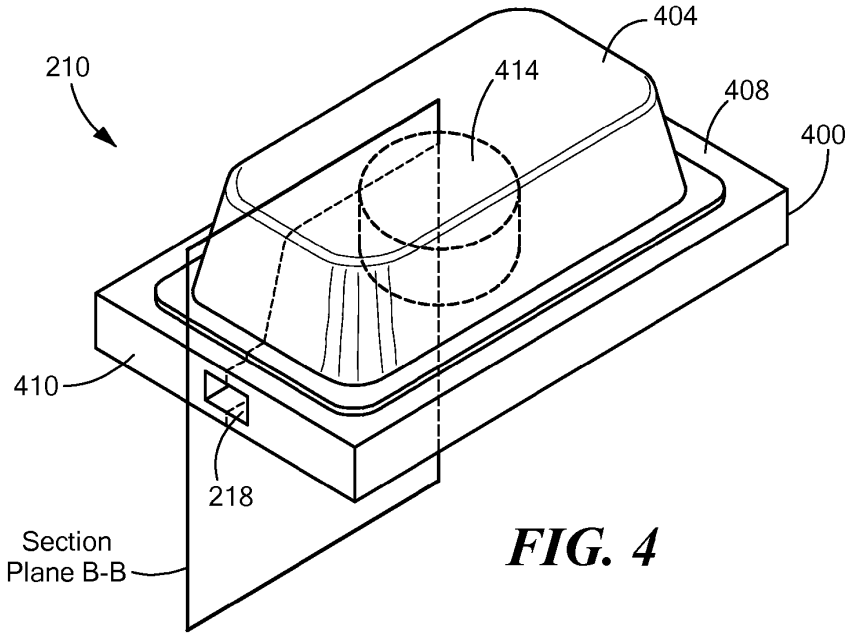

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FIG. 4

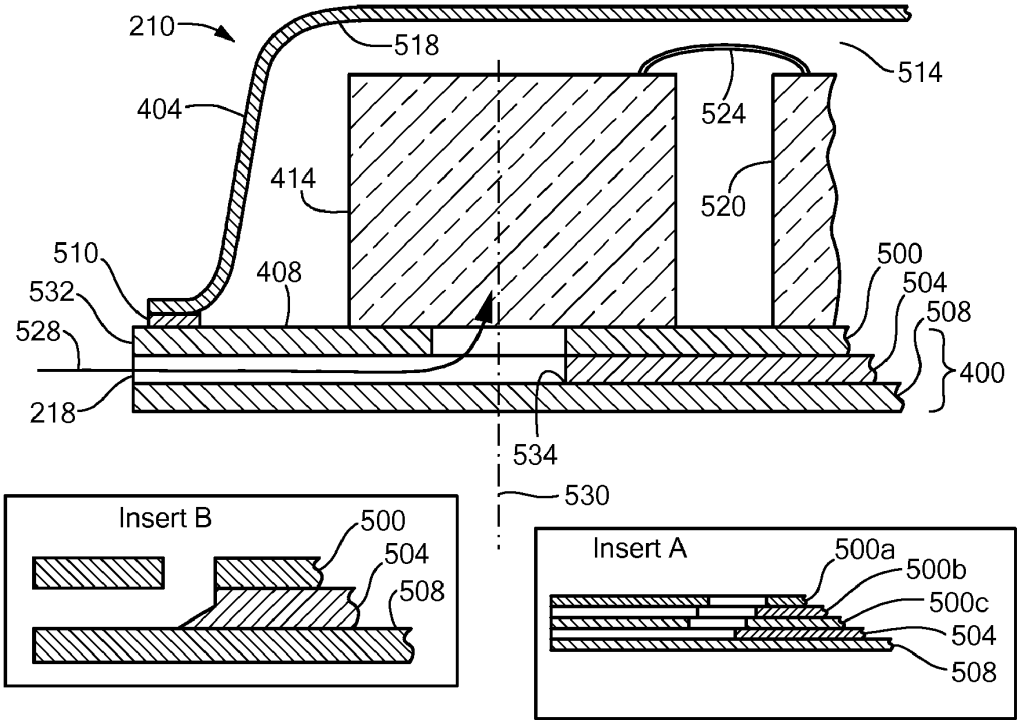

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Section B-B

FIG. 5

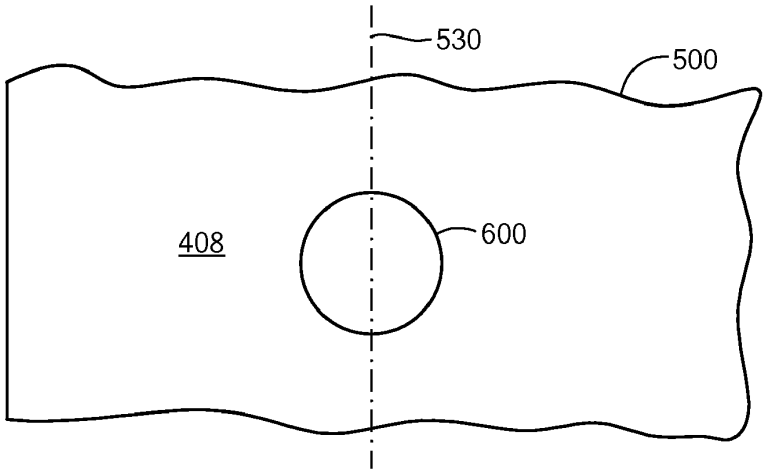

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FIG. 6

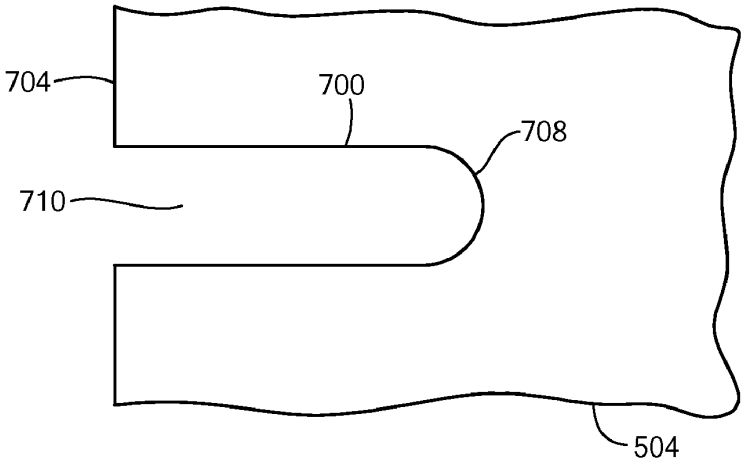

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FIG. 7

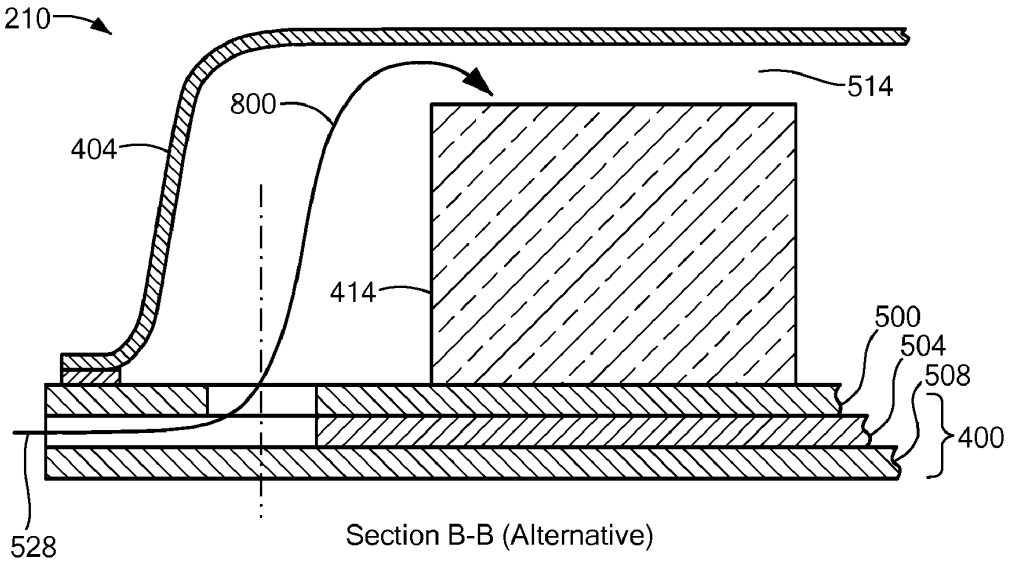

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Section B-B (Alternative)

FIG. 8

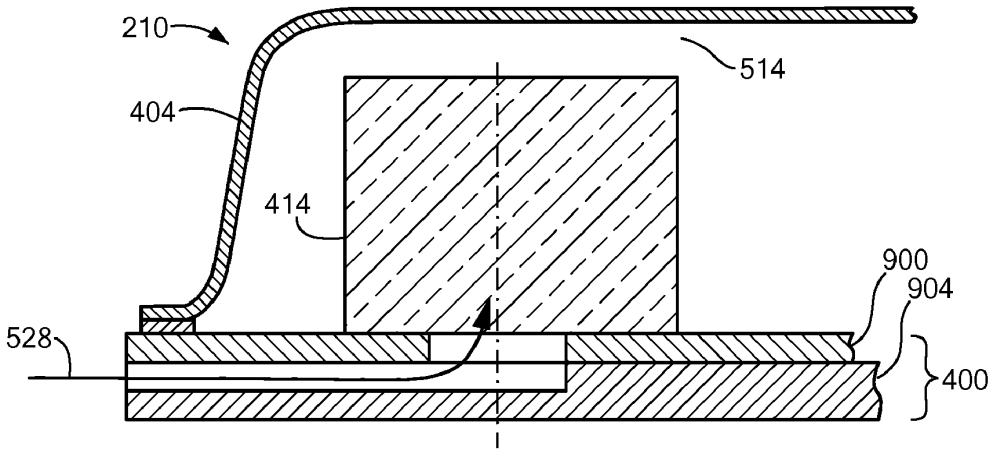

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Section B-B (Alternative)

FIG. 9

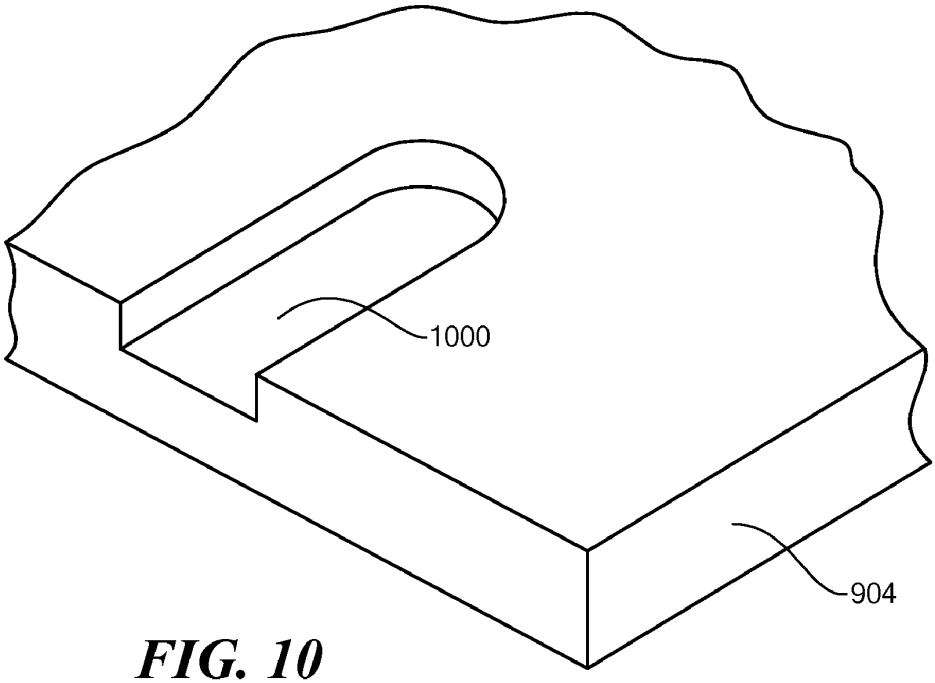
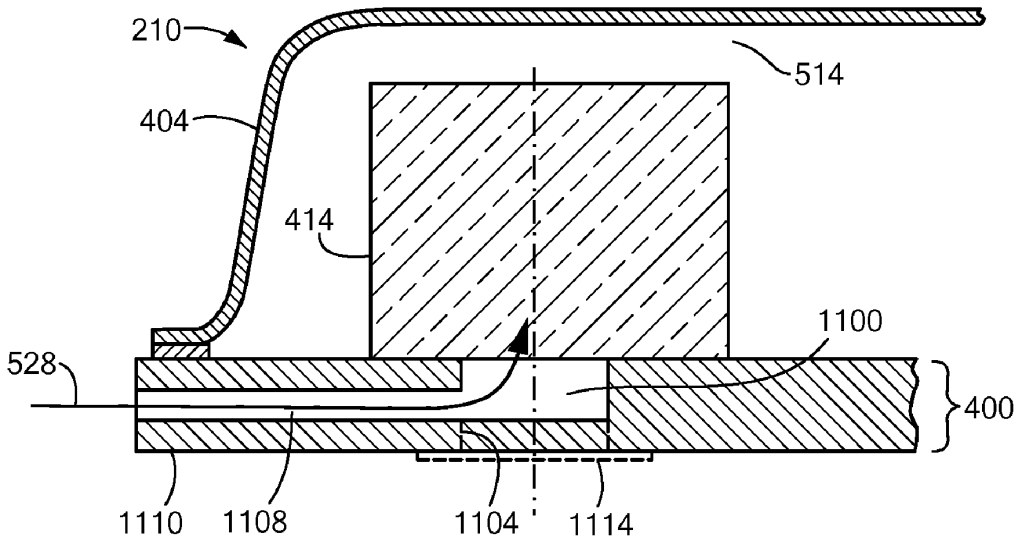

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FIG. 10



Section B-B (Alternative)

FIG. 11

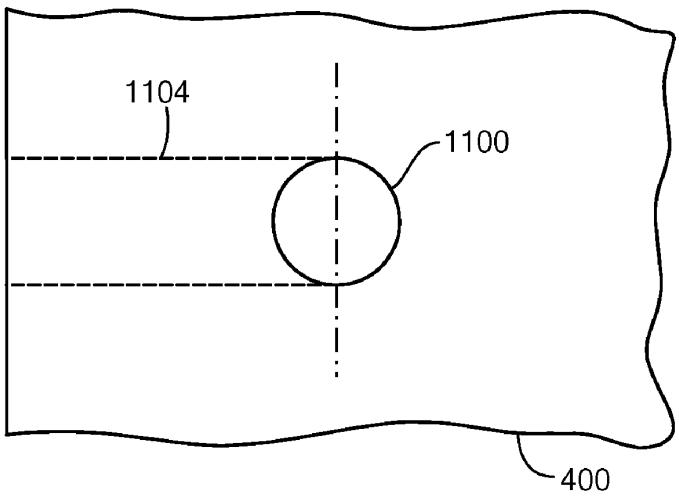


FIG. 12

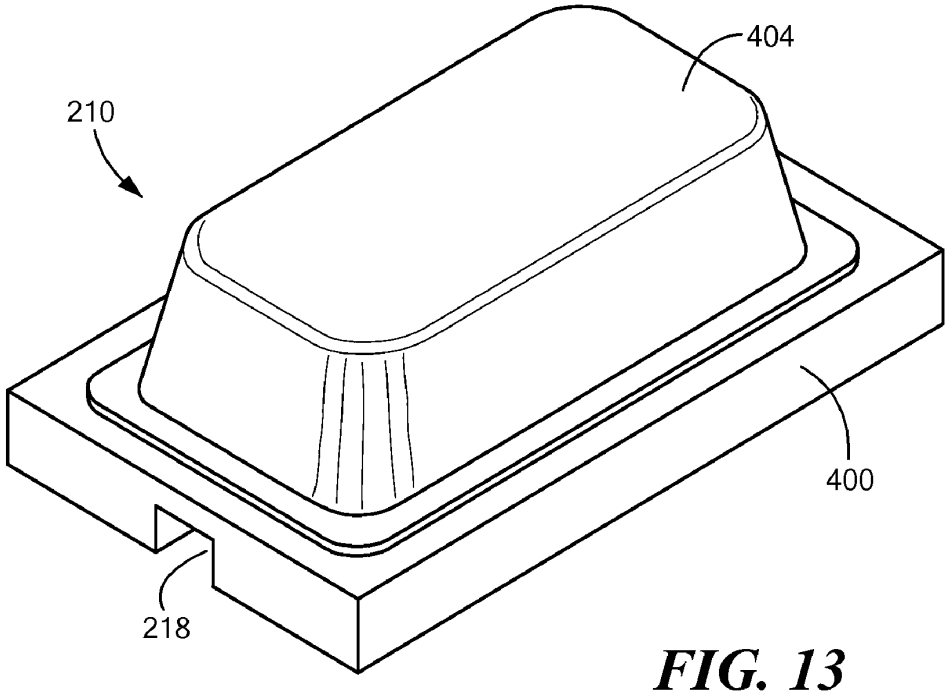

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FIG. 13

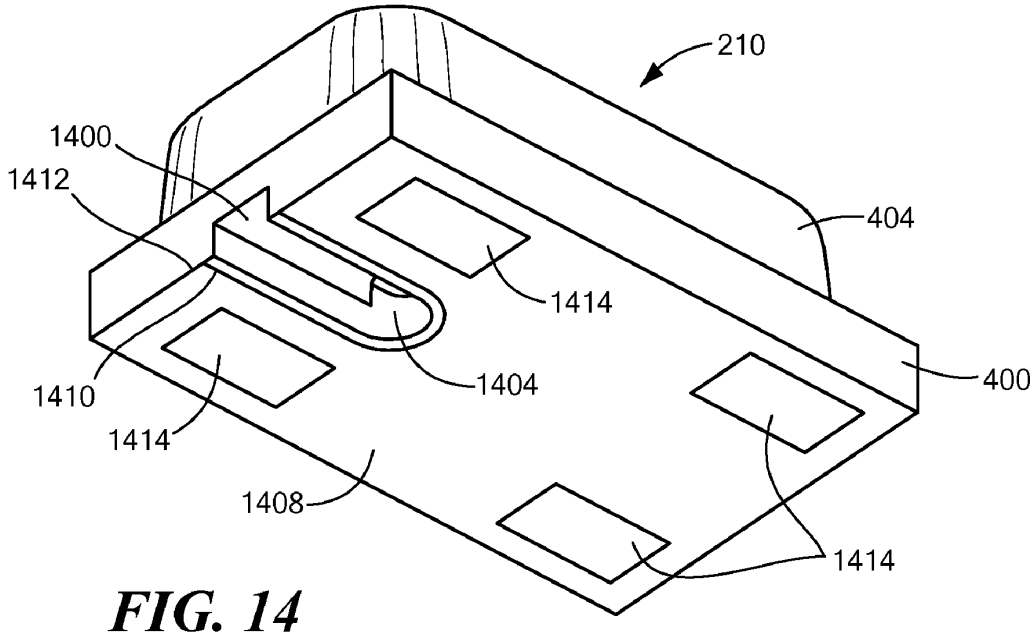

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FIG. 14

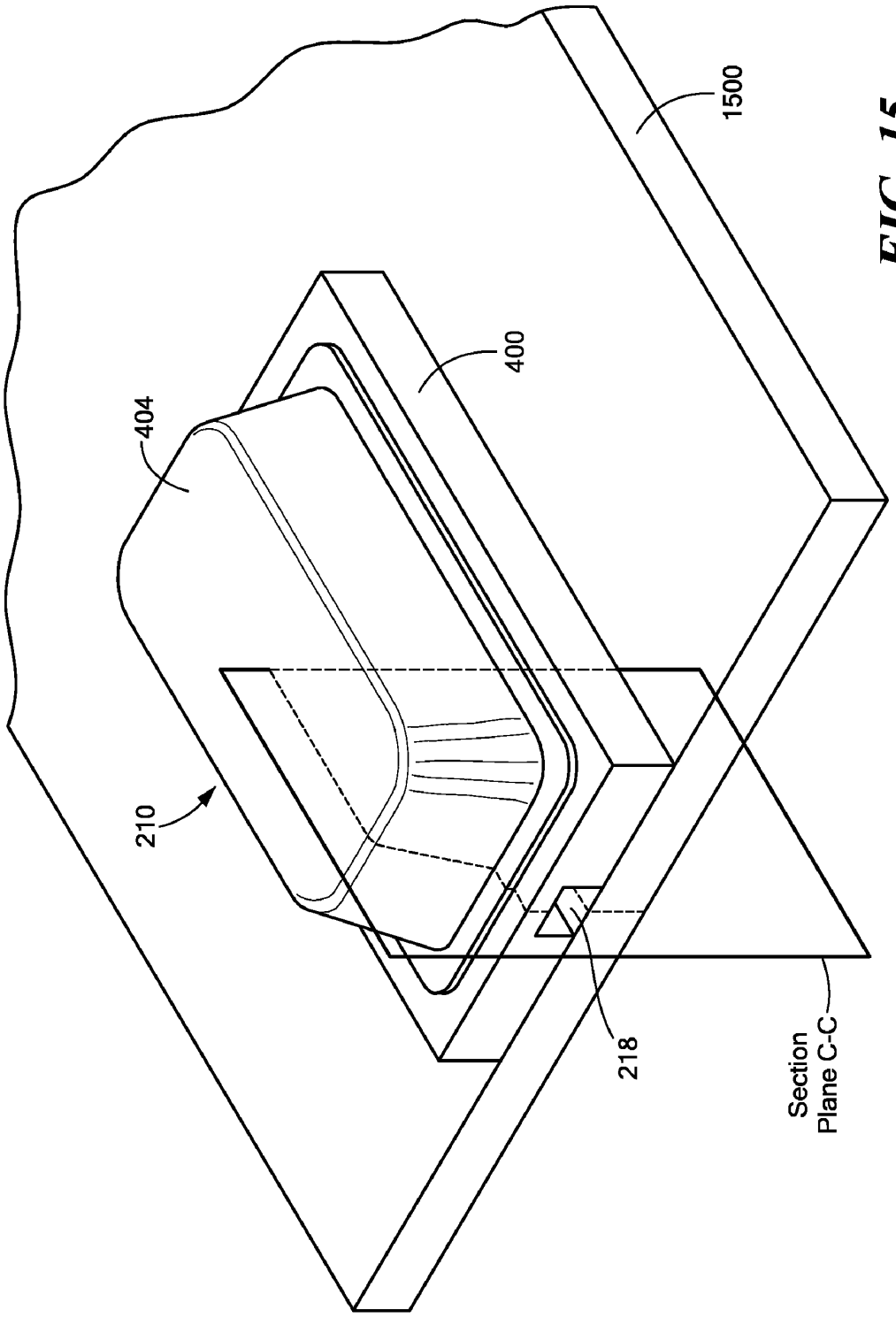


FIG. 15

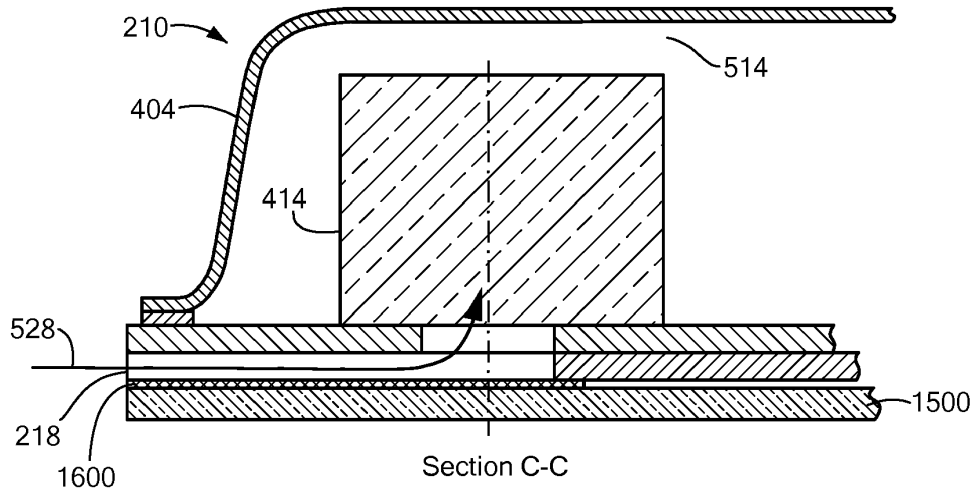


FIG. 16

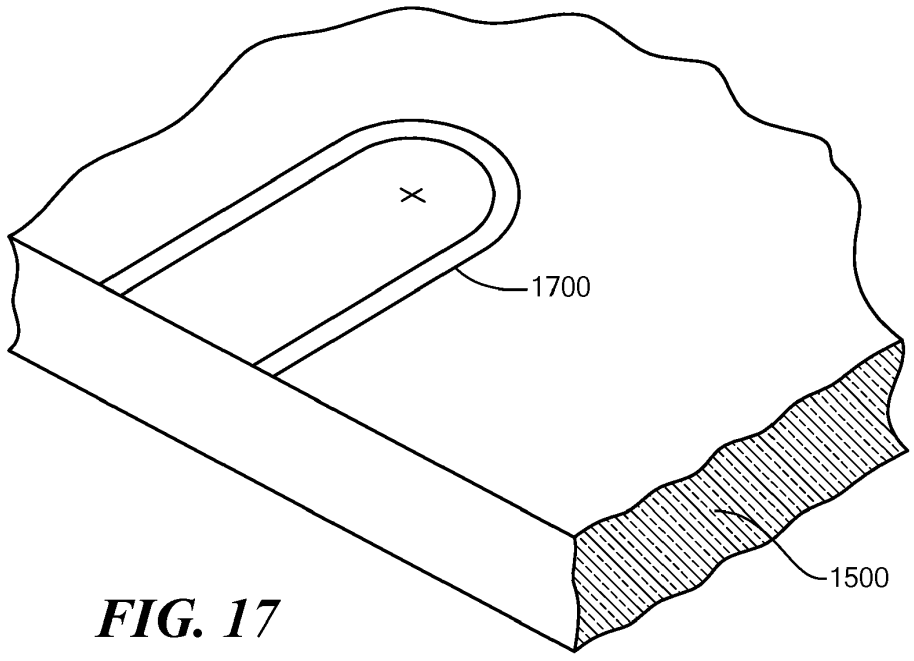


FIG. 17

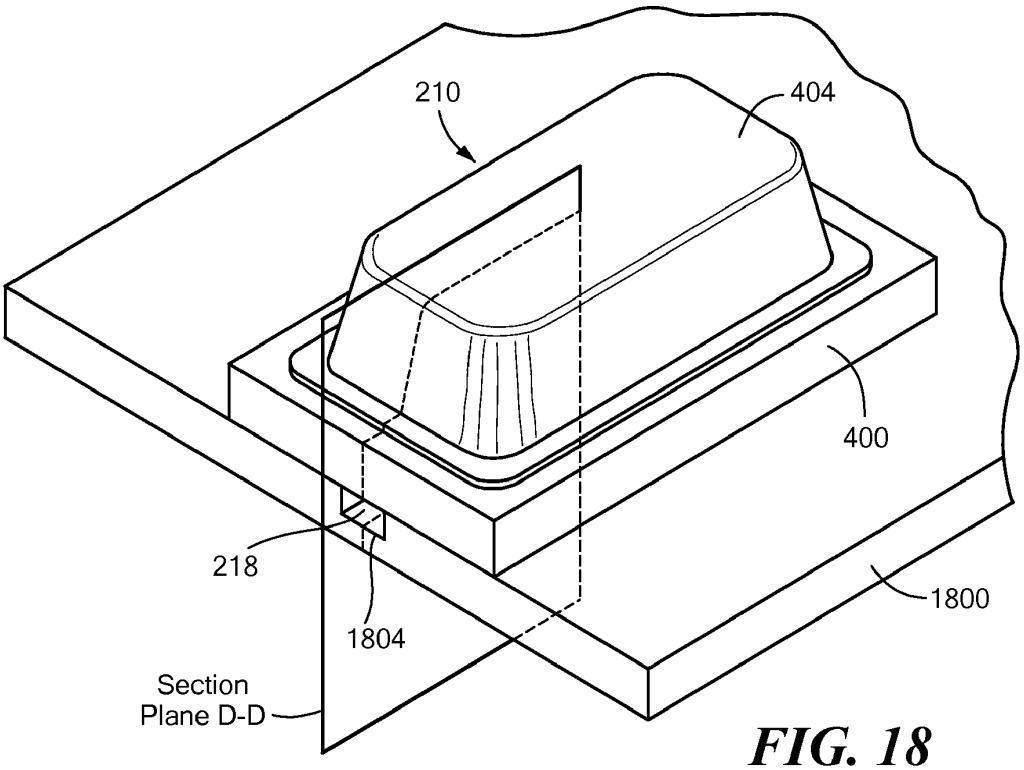

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FIG. 18

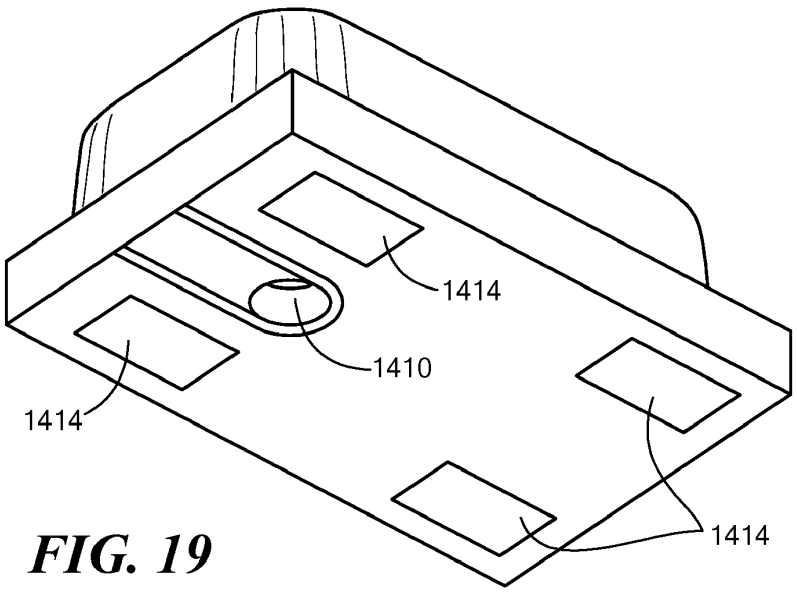

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FIG. 19

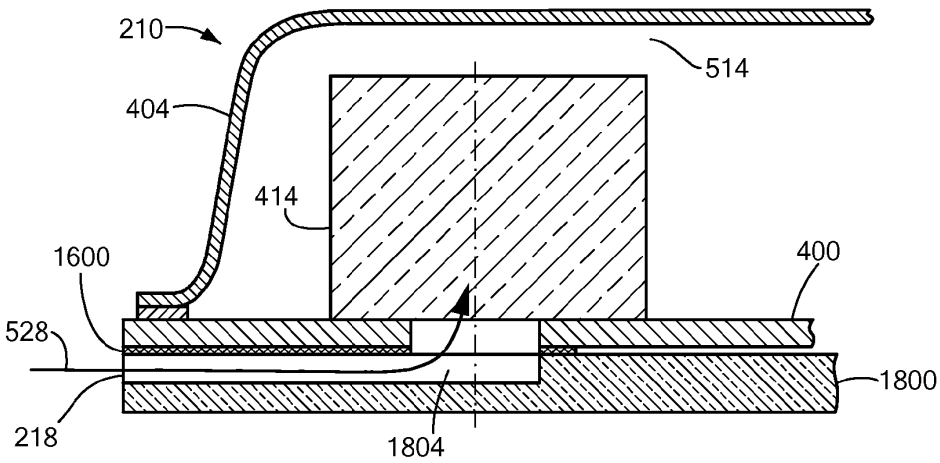

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Section D-D

FIG. 20

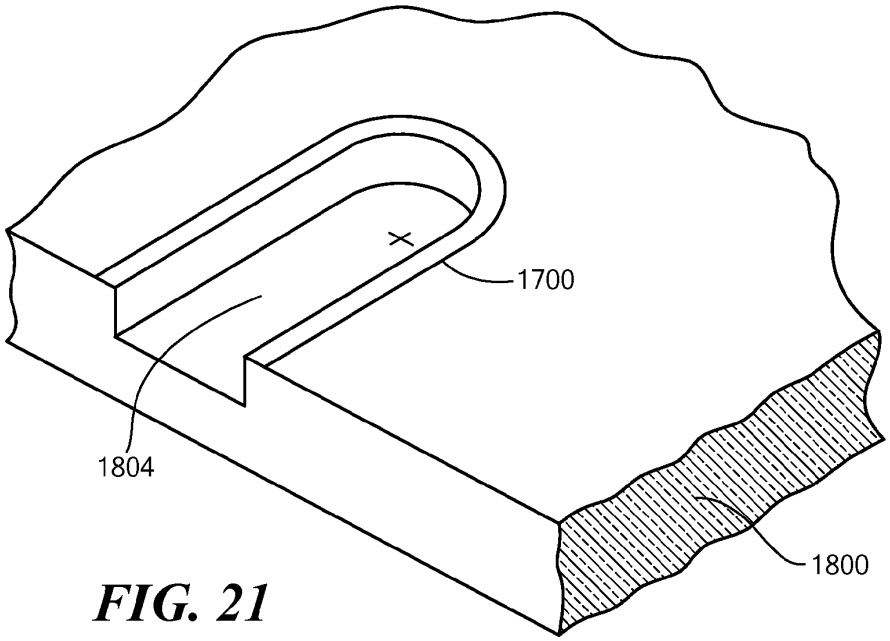

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FIG. 21

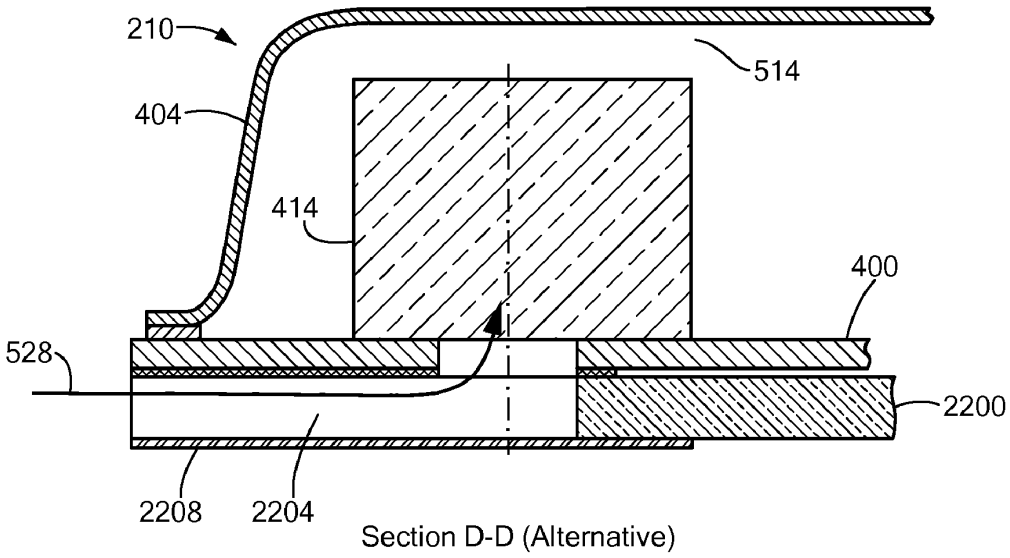

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Section D-D (Alternative)

FIG. 22

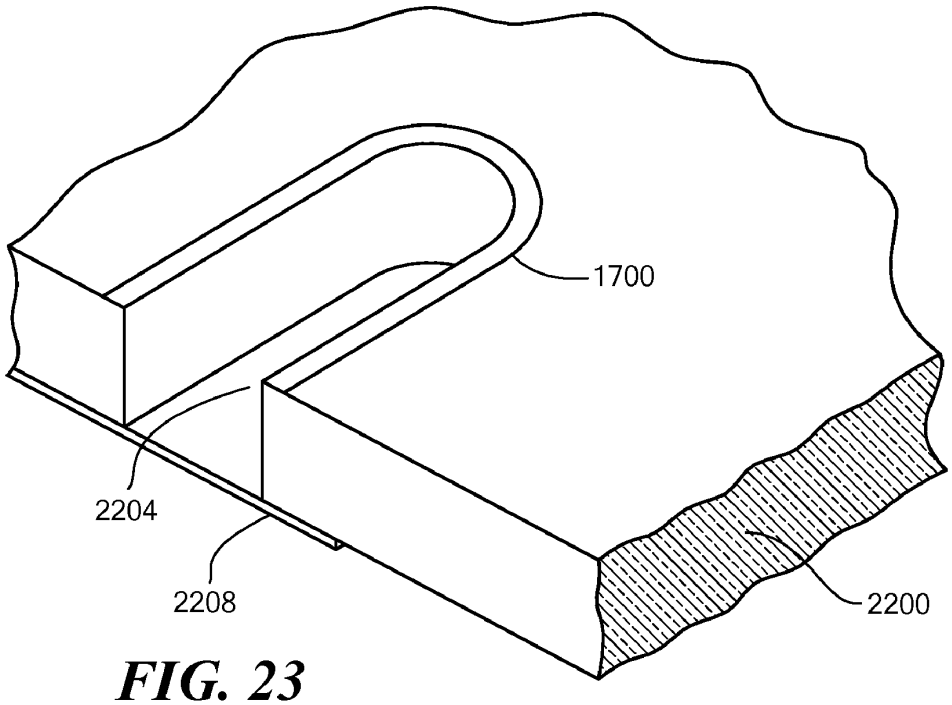

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FIG. 23

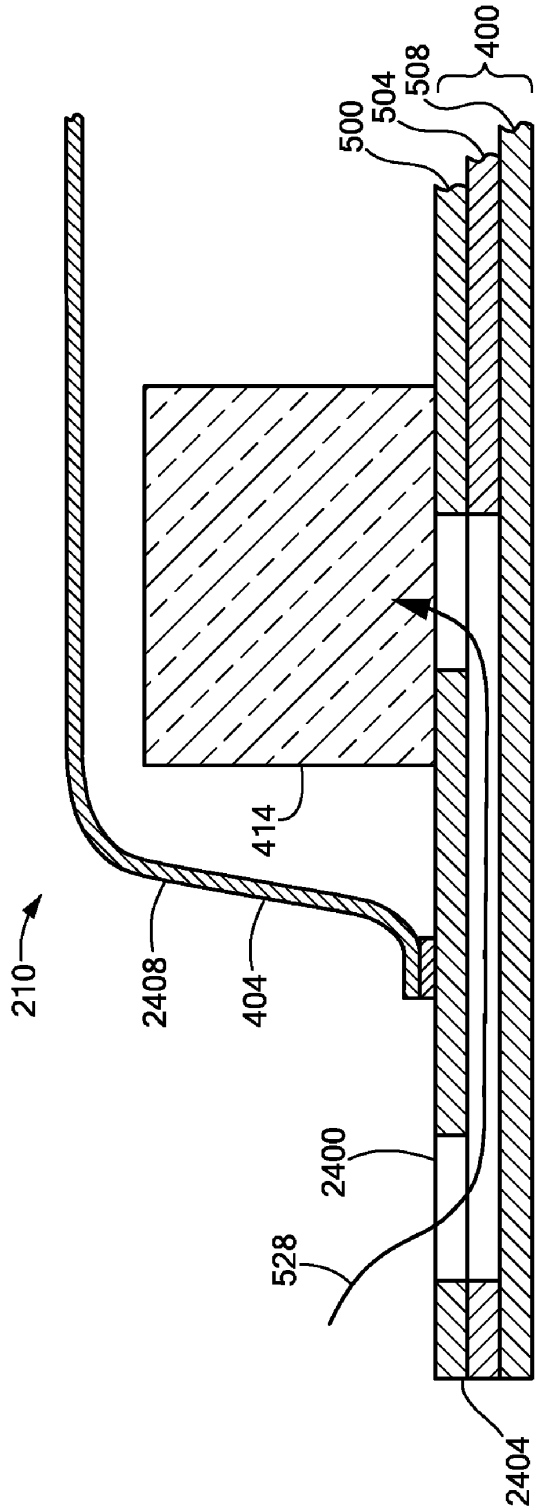


FIG. 24

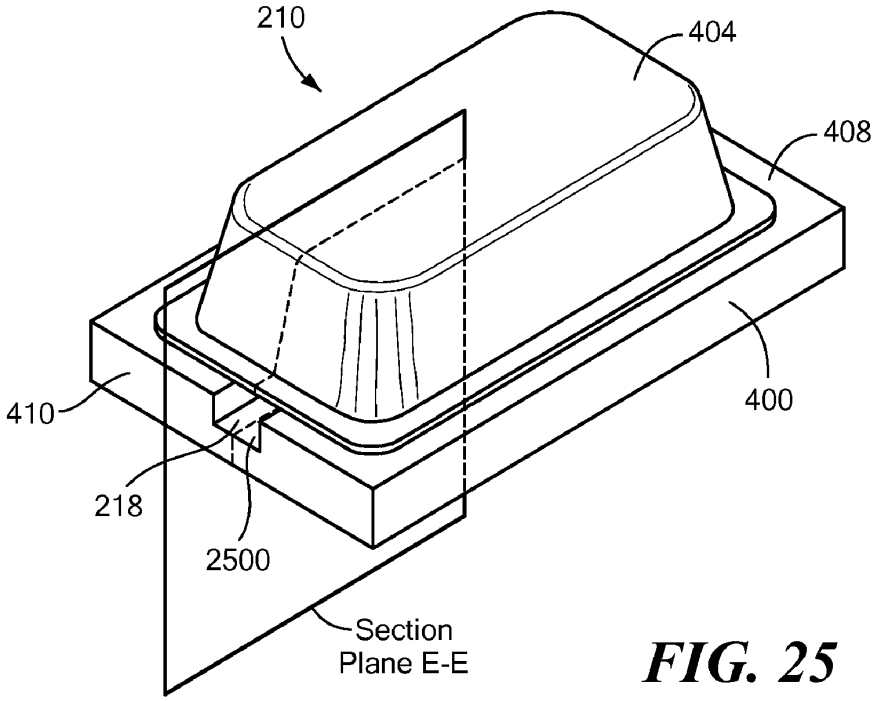
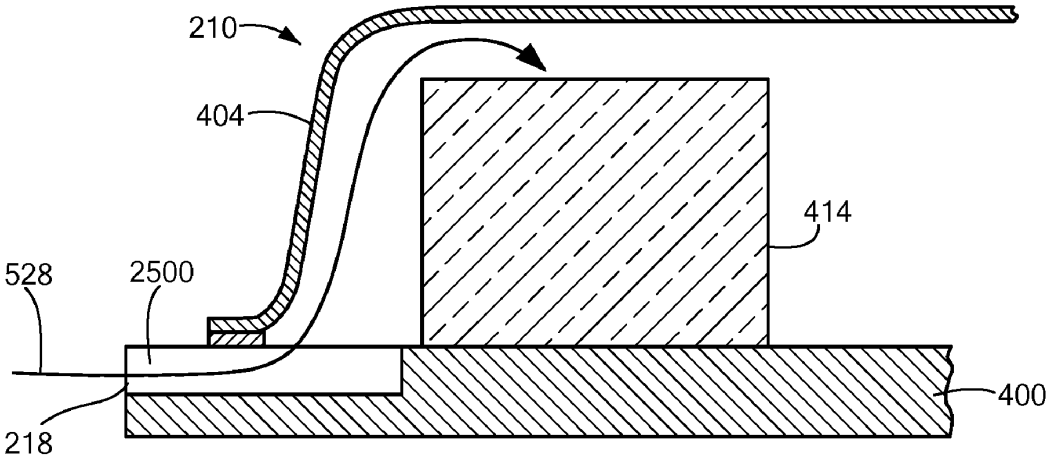


FIG. 25



Section E-E

FIG. 26

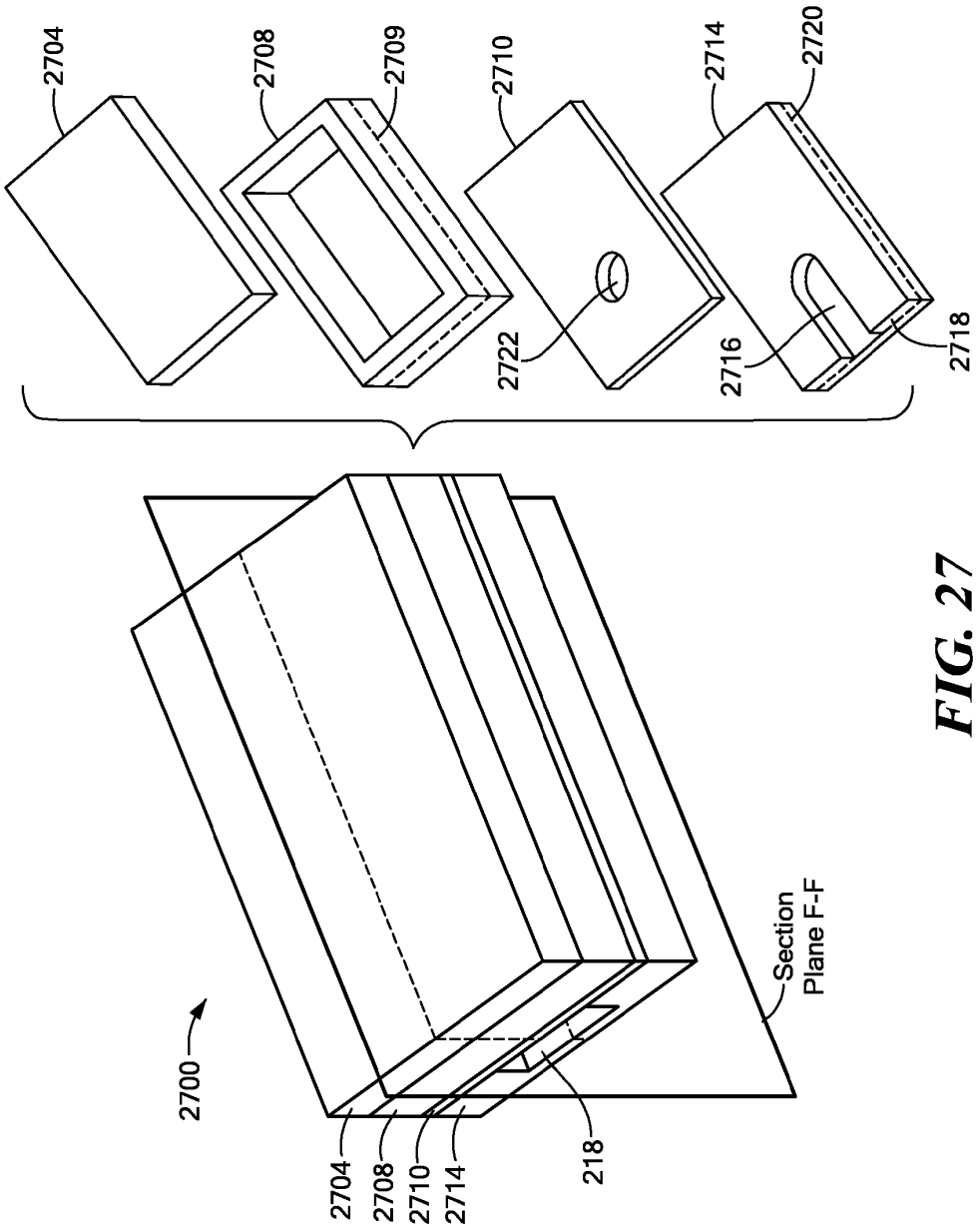

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FIG. 27

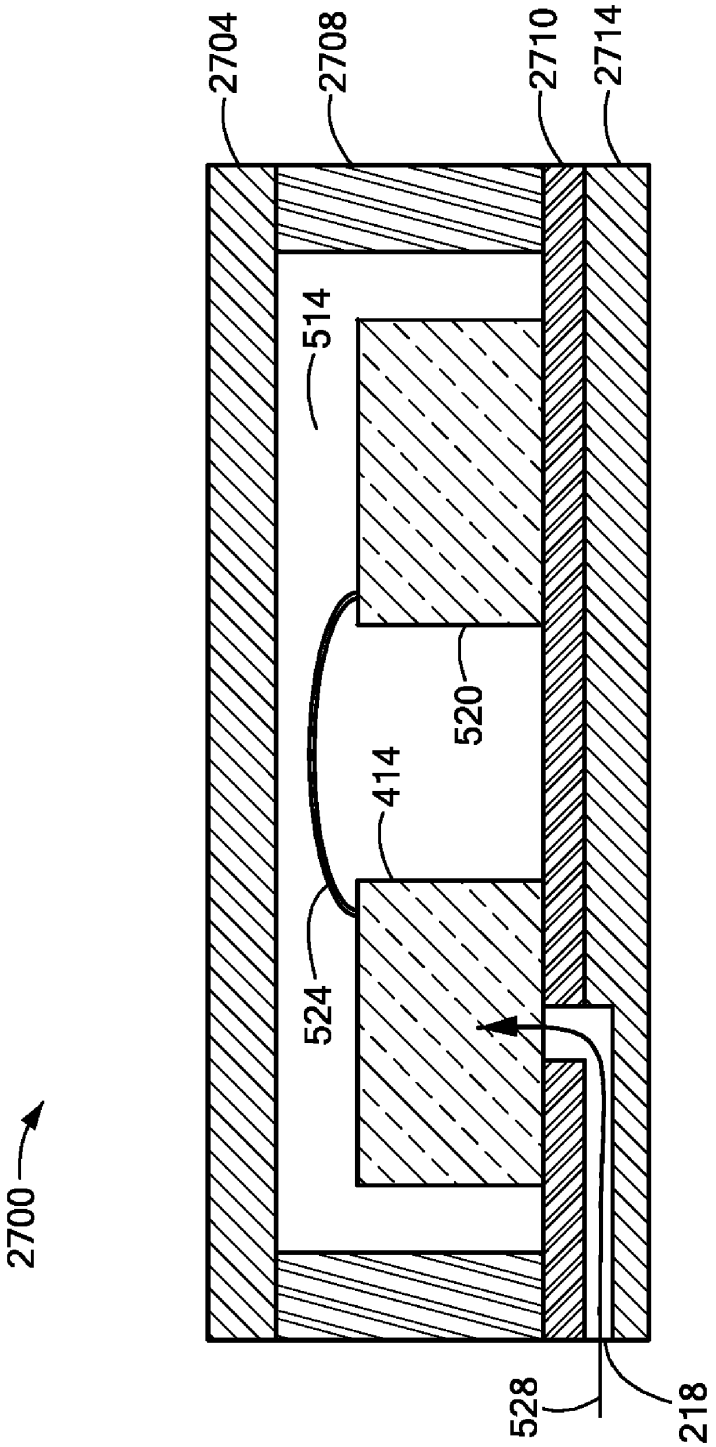


FIG. 28

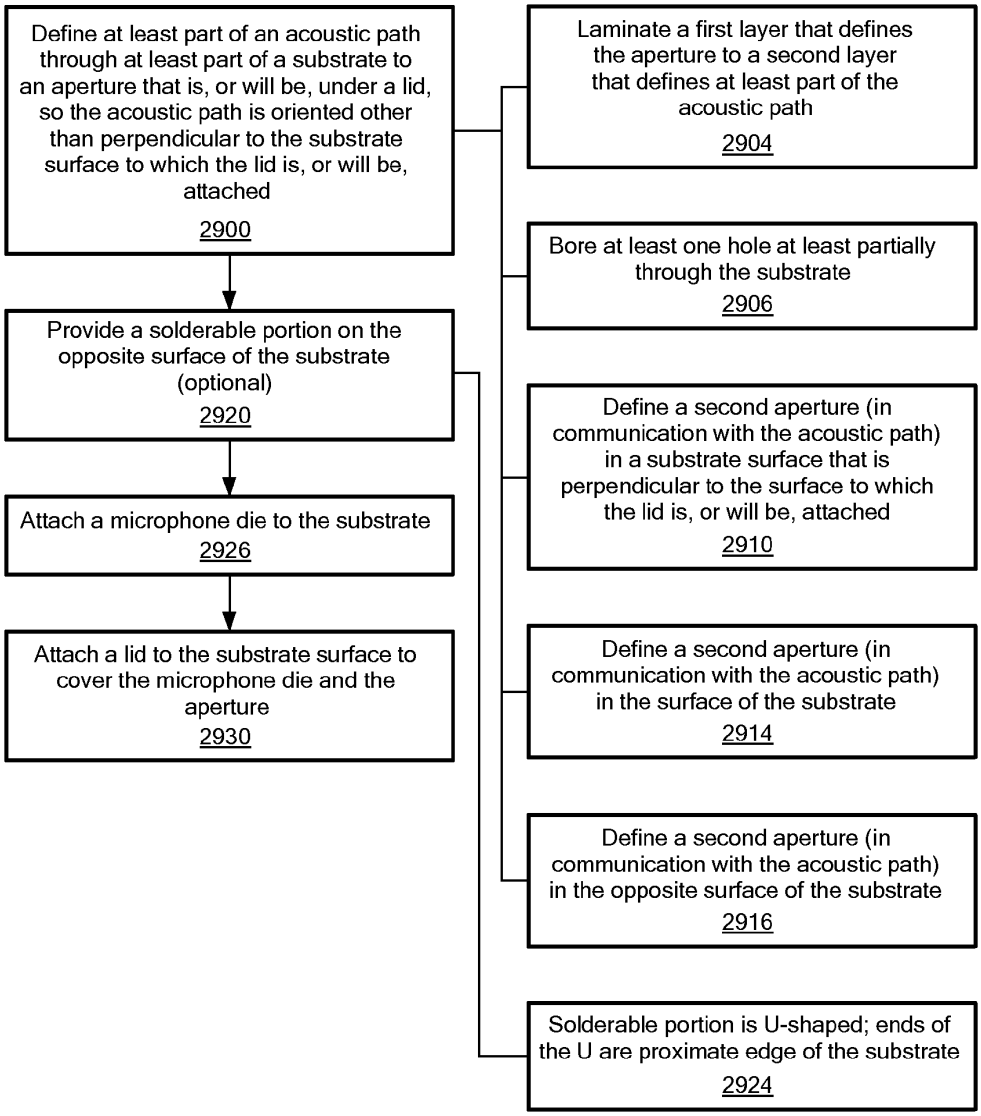

Define at least part of an acoustic path through at least part of a substrate to an aperture that is, or will be, under a lid, so the acoustic path is oriented other than perpendicular to the substrate surface to which the lid is, or will be, attached
2900
Provide a solderable portion on the opposite surface of the substrate (optional)
2920
Attach a microphone die to the substrate
2926
Attach a lid to the substrate surface to cover the microphone die and the aperture
2930
Laminate a first layer that defines the aperture to a second layer that defines at least part of the acoustic path
2904
Bore at least one hole at least partially through the substrate
2906
Define a second aperture (in communication with the acoustic path) in a substrate surface that is perpendicular to the surface to which the lid is, or will be, attached
2910
Define a second aperture (in communication with the acoustic path) in the surface of the substrate
2914
Define a second aperture (in communication with the acoustic path) in the opposite surface of the substrate
2916
Solderable portion is U-shaped; ends of the U are proximate edge of the substrate
2924

FIG. 29

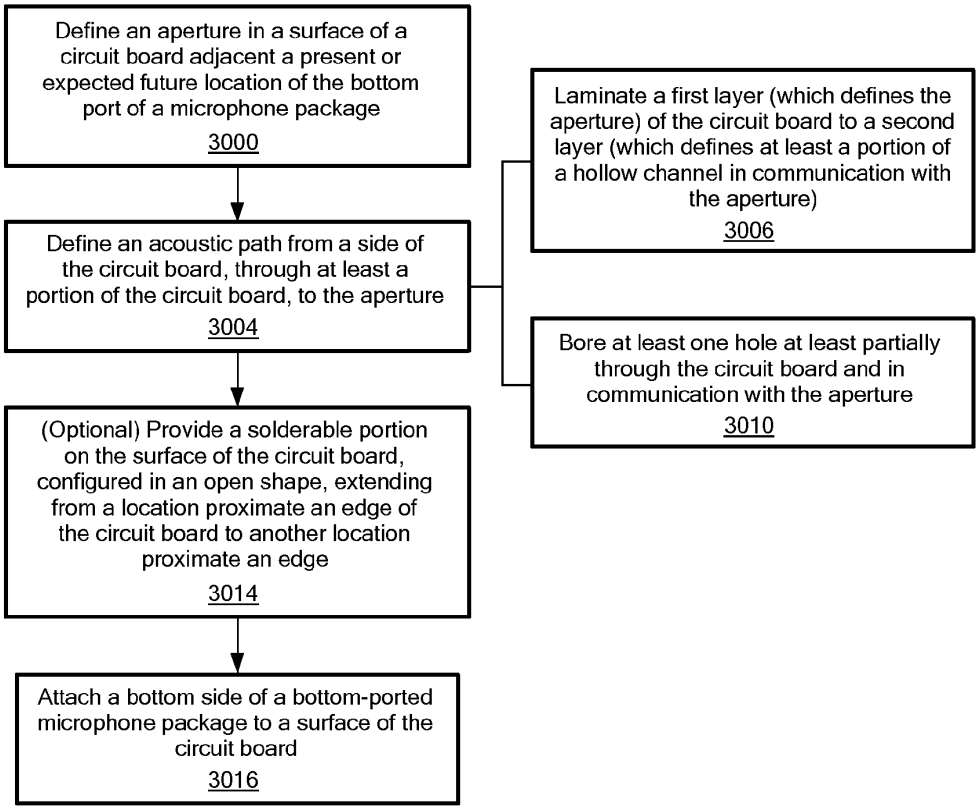

Define an aperture in a surface of a circuit board adjacent a present or expected future location of the bottom port of a microphone package
3000
Define an acoustic path from a side of the circuit board, through at least a portion of the circuit board, to the aperture
3004
(Optional) Provide a solderable portion on the surface of the circuit board, configured in an open shape, extending from a location proximate an edge of the circuit board to another location proximate an edge
3014
Attach a bottom side of a bottom-ported microphone package to a surface of the circuit board
3016
Laminate a first layer (which defines the aperture) of the circuit board to a second layer (which defines at least a portion of a hollow channel in communication with the aperture)
3006
Bore at least one hole at least partially through the circuit board and in communication with the aperture
3010

FIG. 30

SIDE-PORTED MEMS MICROPHONE ASSEMBLY

TECHNICAL FIELD

The present invention relates to microelectromechanical systems (MEMS) microphone packages and, more particularly, to side-ported MEMS microphone packages and related methods.

BACKGROUND ART

A typical microelectromechanical systems (MEMS) microphone package includes a substrate, such as an FR-4 based printed circuit board (PCB), a MEMS microphone die attached to the substrate and a cup-shaped lid or cover that is attached to the substrate to create a chamber, within which the microphone die is protected from environmental insults. The lid or the substrate defines an aperture, through which sound enters the chamber. For example, the substrate may define a through-hole under the microphone die. The microphone die detects the sound and generates corresponding electrical signals. In some implementations, other dies are co-located with the microphone die within the chamber, such as to process the electrical signals generated by the microphone die. The package typically includes electrical contact pads on the bottom surface of the substrate, by which the package can be mechanically and electrically connected to a circuit board, such as by solder or electrically conductive adhesive.

MEMS microphones are commonly used in mobile telephones, laptop computers, voice recorders and other electronic devices. These devices are typically made by placing printed circuit boards inside plastic housings. Often, the housings include user interface buttons or thin membranes, with which users may actuate electrical switches mounted on the printed circuit boards. The housings usually define ports (holes) adjacent the MEMS microphone packages, so the users' speech may enter the housing and be detected by the microphones.

Ergonomic considerations typically lead to microphone ports being located on thin sides, rather than on broad flat surfaces, of the electronic devices. For example, on a mobile telephone, the microphone port is typically located on the thin lower side of the telephone, near the bottom row of dial buttons. Because the main circuit board of the mobile telephone is co-planar with the front of the telephone, i.e., with the plane of the dial buttons, the MEMS microphone package is mounted on a differently oriented auxiliary circuit board (a "daughter board"), so as to orient the aperture of the MEMS microphone package in line with the port, i.e., along an axis that is parallel to the main circuit board.

The height of the resulting structure (i.e., the MEMS microphone package and daughter board) poses problems, because the electronic device must be thick enough to accommodate the height of the structure within the device housing. Demand for ever thinner mobile telephones and other electronic devices translates into a demand for a lower profile solution to mounting MEMS microphone packages on circuit boards.

U.S. Pat. No. 6,781,231 to Minervini, which is hereby incorporated by reference, discloses a MEMS microphone package in which an aperture is defined in a side of the microphone package lid. When the resulting side-ported MEMS microphone package is mounted on a main circuit board of a mobile telephone, the axis of the microphone package aperture is oriented in line with the telephone housing microphone port, and the MEMS microphone package presents a low profile. However, defining apertures in the sides of package lids is difficult and expensive and poses other problems.

SUMMARY OF THE INVENTION

An embodiment of the present invention provides a side-ported microphone package. The package may include a substrate and a lid attached to a first surface of the substrate to define a chamber therebetween. A microphone die may be disposed within the chamber. The substrate may at least partially define an acoustic path from outside the chamber to the microphone die, such that at least a portion of the acoustic path defined by the substrate is oriented other than perpendicular to the first surface of the substrate. For example, the at least a portion of the defined acoustic path may be oriented parallel to the first surface of the substrate.

The substrate may define an aperture in the first surface within a boundary defined by a locus of points where the lid is attached to the substrate. The substrate may also define an opening on a surface of the substrate and outside the boundary. The substrate may further define a hollow passage in communication with the aperture and with the opening, such that the passage, the aperture and the opening form at least a portion of the defined acoustic path.

The opening may be defined by a surface of the substrate that is substantially perpendicular to the first surface or by the first surface.

The substrate may include a second surface on substantially an opposite side of the substrate from the first surface. None or at least a portion of the passage may be open at the second surface.

The second surface may include a solderable portion configured into an open shape, which extends from a location proximate an edge of the substrate to another location proximate the same or a different edge of the substrate. The solderable portion may substantially bound the open portion of the passage between the solderable portion and the edge of the substrate. The solderable portion may be U shaped, with ends of the U shape proximate the edge of the substrate.

The substrate may include a laminate. A first layer of the laminate may define the aperture, and a second layer of the laminate may define at least a portion of the passage.

The aperture may be adjacent to, and in line with, a receiver portion of the microphone die. The microphone die may be disposed between the aperture and the chamber. The aperture may be in communication with the chamber.

The substrate may include a printed circuit board, ceramic and/or plastic. The substrate may include at least one wall, and the lid may be attached to the at least one wall.

Another embodiment of the present invention provides a side-ported microphone package that includes a substrate having a first surface and a second surface. The second surface is on substantially an opposite side of the substrate from the first surface. The microphone package also includes a lid attached to the first surface of the substrate to define a chamber therebetween. A microphone die may be disposed within the chamber. The second surface defines an opening. The substrate at least partially defines an acoustic path between the opening and the microphone die. The second surface comprises a solderable portion configured into an open shape that extends from a location proximate an edge of the substrate to another location proximate the same edge or a different edge of the substrate. The solderable portion substantially bounds the opening between the solderable portion and the edge of the substrate. However, the solderable portion

does not surround the opening. The solderable portion may be U shaped, with ends of the U shape proximate the edge of the substrate.

Yet another embodiment of the present invention provides a side-ported microphone package that includes a substrate and a lid attached to the substrate to define a chamber therebetween. A microphone die may be disposed within the chamber. The substrate defines a sound input port in communication with the microphone die. The sound input port is oriented substantially perpendicular to the microphone die.

Another embodiment of the present invention provides a side-ported microphone package that includes a substrate having a top surface, a bottom surface and an end surface. The end surface defines an opening. A lid is attached to the substrate to define a chamber therebetween. A microphone die may be disposed within the chamber. The substrate defines at least a portion of an acoustic path from the opening to the microphone die.

An embodiment of the present invention provides a kit of parts for assembling a side-ported microphone package. The kit includes a first layer, a second layer, at least one wall and a lid. The first layer defines a hollow channel or notch in a surface of the first layer. The channel or notch extends from an edge of the first layer and terminates at a location away from any edge of the first layer. The second layer defines a bore. When the first and second layers are assembled in registration with each other, the bore is in communication with the channel or notch of the first layer. When the at least one wall is attached to the second layer in registration therewith, the bore in the second layer is in communication with a volume defined by the at least one wall.

The first layer and the second layer may each include a printed circuit board, ceramic, plastic and/or another material. Optionally or alternatively, the lid may include a printed circuit board.

Yet another embodiment of the present invention provides a method for producing a side-ported microphone package. A microphone die is disposed, relative to a substrate. A lid is attached to a first surface of the substrate to define a chamber within which the microphone die is, or will be, disposed. At least a portion of an acoustic path is defined from outside the chamber, through at least a portion of the substrate, to an aperture defined by the first surface. The aperture is within a boundary defined by a locus of points where the lid is, or will be, attached to the substrate. At least a portion of the defined acoustic path is oriented other than perpendicular to the first surface of the substrate.

A plurality of layers of the substrate may be laminated to define the at least a portion of the acoustic path. A first layer of the substrate may define the first aperture, and a second layer of the substrate may define at least a portion of the acoustic path.

The at least a portion of the acoustic path may be defined, at least in part, by boring at least one hole at least partially through the substrate. Optionally or alternatively, the at least a portion of the acoustic path may be defined, at least in part, by defining an opening in a second surface of the substrate that is substantially perpendicular to the first surface, so the opening is in communication with the acoustic path. Optionally or alternatively, the at least a portion of the acoustic path may be defined, at least in part, by defining an opening in the first surface, the opening being in communication with the acoustic path. Optionally or alternatively, the at least a portion of the acoustic path may be defined, at least in part, by defining an opening in a second surface of the substrate, the second surface being on substantially an opposite side of the substrate from the first surface, the opening being in communication with the acoustic path

Optionally or alternatively, a solderable portion may be provided on the second surface. The solderable portion may be configured into an open shape. The solderable portion may extend from a location proximate an edge of the substrate to another location proximate the same or a different edge of the substrate. The solderable portion may substantially bound the opening between the solderable portion and the edge of the substrate.

The solderable portion may be U shaped, with ends of the U shape proximate the edge of the substrate.

another embodiment of the present invention provides a method for producing a side-ported microphone assembly. A bottom side of a bottom-ported microphone package is attached to a surface of a circuit board. An opening is defined in the surface of the circuit board adjacent a present or expected future location of the bottom port of the microphone package. A hollow acoustic path is defined from a side of the circuit board, through at least a portion of the circuit board, to the opening.

The acoustic path may be defined by laminating a plurality of layers of the circuit board. A first layer of the circuit board defines the opening, and a second layer of the circuit board defines at least a portion of a hollow passage in communication with the opening and extending to the side of the circuit board.

The acoustic path may be defined by boring at least one hole at least partially through the circuit board. The hole may be in communication with the opening.

Optionally, a solderable portion may be provided on the surface of the circuit board. The solderable portion may be configured into an open shape. The solderable portion may extend from a location proximate an edge of the circuit board to another location proximate the same or a different edge of the circuit board. The solderable portion may substantially bound the opening between the solderable portion and the edge of the circuit board.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be more fully understood by referring to the following Detailed Description of Specific Embodiments in conjunction with the Drawings, of which:

FIG. **1** is a perspective view of an exemplary mobile telephone, in which embodiments of the present invention may be used;

FIG. **2** is a cross-sectional view of a portion of the mobile telephone of FIG. **1**;

FIG. **3** is a more detailed perspective view of an interior portion of the mobile telephone of FIG. **1**;

FIG. **4** is a perspective view of a side-ported MEMS microphone package, according to embodiments of the present invention;

FIG. **5** is a cross-sectional view of a portion of the MEMS microphone package of FIG. **4**, according to one embodiment of the present invention;

FIG. **6** is a plan view of a layer of a substrate of the side-ported MEMS microphone package of FIG. **5**;

FIG. **7** is a plan view of another layer of the substrate of the side-ported MEMS microphone package of FIG. **5**;

FIG. **8** is a cross-sectional view of a portion of the MEMS microphone package of FIG. **4**, according to another embodiment of the present invention;

FIG. **9** is a cross-sectional view of a portion of the MEMS microphone package of FIG. **4**, according to yet another embodiment of the present invention;

FIG. **10** is a perspective view of a layer of a substrate of the side-ported MEMS microphone package of FIG. **9**;

FIG. **11** is a cross-sectional view of a portion of the MEMS microphone package of FIG. **4**, according to a different embodiment of the present invention;

FIG. **12** is a plan view of a substrate of the side-ported MEMS microphone package of FIG. **11**;

FIG. **13** is a perspective view of a side-ported MEMS microphone package, according to another embodiment of the present invention;

FIG. **14** is a bottom perspective view of the side-ported MEMS microphone package of FIG. **13**;

FIG. **15** is a perspective view of the side-ported MEMS microphone package of FIGS. **13** and **14** and a circuit board, according to an embodiment of the present invention;

FIG. **16** is a cross-sectional view of a portion of the MEMS microphone package of FIGS. **13-15**;

FIG. **17** is a perspective view of a portion of the circuit board of FIG. **15**, according to an embodiment of the present invention;

FIG. **18** is a perspective view of a MEMS microphone package and a circuit board, according to another embodiment of the present invention;

FIG. **19** is a bottom perspective view of the MEMS microphone package of FIG. **18**;

FIG. **20** is a cross-sectional view of a portion of the MEMS microphone package and circuit board of FIGS. **18-19**;

FIG. **21** is a perspective view of a portion of the circuit board of FIGS. **18** and **20**, according to an embodiment of the present invention;

FIG. **22** is a cross-sectional view of a portion of the MEMS microphone package and circuit board of FIGS. **18-19**, according to another embodiment of the present invention;

FIG. **23** is a perspective view of a portion of the circuit board of FIGS. **18** and **22**;

FIG. **24** is a cross-sectional view of a portion of a side-ported MEMS microphone package, according to yet another embodiment of the present invention;

FIG. **25** is a perspective view of a side-ported MEMS microphone package, according to an embodiment of the present invention;

FIG. **26** is a cross-sectional view of a portion of the MEMS microphone package of FIG. **25**;

FIG. **27** is a perspective and an exploded view of a side-ported MEMS microphone package, according to another embodiment of the present invention;

FIG. **28** is a cross-sectional view of the MEMS microphone package of FIG. **27**;

FIG. **29** is a flowchart that describes producing a side-ported MEMS microphone package, according to embodiments of the present invention; and

FIG. **30** is a flowchart that describes producing a side-ported MEMS microphone assembly, according to another embodiment of the present invention.

DETAILED DESCRIPTION OF SPECIFIC EMBODIMENTS

In accordance with embodiments of the present invention, methods and apparatus are disclosed for providing side-ported microphone packages and side-ported microphone assemblies. The substrate of a microelectromechanical systems (MEMS) microphone package may at least partially define an acoustic path from a side of the substrate to a microphone die disposed within a chamber defined by the substrate and a lid attached to the substrate. The acoustic path may be a hollow passage through at least a portion of the substrate. Optionally or alternatively, a circuit board, to which the MEMS microphone package is mounted, may define an acoustic path, such as via a hollow passage, from an edge of the circuit board to a location under the MEMS microphone package, adjacent a bottom port on the microphone package.

Traditionally, a port in a MEMS microphone package is defined in the lid, which is typically at least partially made of electrically conductive metal, so as to protect the microphone die and other circuits inside the package from electrical noise. Alternatively, a traditional MEMS microphone package defines a bottom port, i.e., a hole through the substrate of the package, directly under the microphone die. Both designs suffer from various problems. For example, use of a bottom-ported MEMS microphone package creates the daughter board and structure height problems described above. On the other hand, conventional side-ported packages, with ports in the sides of their lids, can lead to degraded audio quality, due to the audio response characteristics of the chamber, through which sound must travel before reaching the microphone die. These and other shortcomings are overcome by the packages and methods disclosed herein.

Surprisingly, it has been found that a MEMS microphone die can be effectively acoustically coupled to a microphone port on an electronic device, at least in part, via an acoustic path defined within a substrate of a MEMS microphone package. One would not have expected sound to propagate well through such a path, in part due to the sharp angle or tortuous geometry of such a path.

As noted, mobile telephones and other electronic devices use MEMS microphones. These electronic devices provide contexts in which the present inventions may be practiced. One exemplary context, a mobile telephone, will now be described. FIG. **1** is a perspective view of a mobile telephone **100**, FIG. **2** is a cross-sectional view of a portion of the mobile telephone **100**, and FIG. **3** is a more detailed perspective view of a portion of the inside of the mobile telephone **100**. The mobile telephone **100** includes a housing **104** that is typically of a "clam shell" design, i.e., the housing **104** is made of a front half **200** and a back half **204** that snap together or are adhered or welded together. The mobile telephone **100** typically includes a set of dial buttons **108**, which is implemented as a thin membrane **208**, by which a user may actuate electrical switches (typified by switch **210**) mounted on a main circuit board **214** disposed within the housing **104**, i.e., between the front half **200** and the back half **204** of the housing **104**.

The housing **104** defines a microphone port **110**, i.e., a hole through which sound enters the housing **104**. A MEMS microphone package **210** is attached to the circuit board **214** near an edge of the circuit board **214**, preferably adjacent the microphone port **110**. The microphone package **210** defines an input aperture **218**, through which sound enters the microphone package **210**. The MEMS microphone package **210** should be positioned such that the aperture **218** is close to, and oriented toward, the microphone port **110**. Optionally, a resilient material, an adhesive or Another suitable material forms a gasket **220** between the inside of the housing **104** and the microphone package **210** and/or the circuit board **214**, so as to create a relatively sealed acoustic path from the microphone port **110** to the input aperture **218**.

As noted, markets demand ever thinner electronic devices. Thus, there is a need for thin (as measured by thickness **114**) mobile telephones. A side-ported MEMS microphone pack-

age, such as the package **210**, facilitates meeting this need, because the microphone package **210** may be mounted on, and parallel to, the main circuit board **214**, without the need for a daughter board or other arrangement for orienting a lid port or a bottom port of the microphone package towards the microphone port **110** of the mobile telephone **100**. Furthermore, a MEMS microphone package **210** having an input aperture **218** on the side of the package substrate, rather than on the side of the lid (as in the prior art), facilitates locating the microphone package input port **218** as close as desired to the edge of the circuit board **214**, thus closer to the microphone port **110** of the mobile telephone housing **104** than is possible with prior art microphone packages. Optionally, the MEMS microphone package **210** may be mounted so as to partially overhang the edge of the circuit board **214**, thereby locating the input port **218** even closer to the microphone port **110** of the housing **104**.

Several exemplary embodiments of the present invention will now be described. FIG. **4** is a perspective view of an exemplary side-ported MEMS microphone package **210**. The package **210** includes a substrate **400** and a lid **404** attached to one surface **408** of the substrate **400** to define a chamber therebetween. The lid **404** may be attached to the substrate **400** by solder, brazing, adhesive or any other suitable material or method. In some embodiments, the lid is cup shaped. A MEMS microphone die **414** (shown in phantom) is disposed in the chamber. Another surface **410** of the substrate **400** defines the input aperture **218**. A hollow passage provides at least part of an acoustic path from the input aperture **218** to the microphone die **414**, as described in more detail below. Although the passage is described as "hollow," a mechanical filter may be fitted in all or part of the passage to prevent moisture or particles of contaminant from entering the chamber. An acoustic path with such a filter is nevertheless considered "hollow."

Because the input aperture **218** is oriented perpendicular to the typical orientation of the MEMS microphone die **414**, the acoustic path should include a bend or angle to direct sound propagating along the acoustic path toward the microphone die **414**. The acoustic path may be formed in various ways, examples of which will now be described.

According to some embodiments, the substrate **400** is laminated, i.e., formed by uniting a plurality of layers of material together. The acoustic path may be defined by a set of passages formed by holes, notches, cuts, channels or other openings in some or all of the layers of the substrate **400**. In other embodiments, the substrate **400** is solid, and the acoustic path may be formed by holes channels, notches, etc., bored, molded or otherwise formed in the substrate **400**. Exemplary embodiments will be described with reference to cross-sectional views of variations of microphone package **210**, taken according to section plane B-B in FIG. **4**.

FIG. **5** is a cross-sectional view of an exemplary MEMS microphone package **210** that includes a laminated substrate **400**. The substrate **400** shown in FIG. **5** includes three layers **500**, **504** and **508**. However, other numbers of layers may be used. As noted, the lid **404** is attached to one surface **408** of the substrate **400** by solder or adhesive **510**, or another suitable material or method, to define a chamber **514**. The MEMS microphone die **414** (shown as a solid block for simplicity) may be attached to the surface **408** of the substrate **400**, as shown in FIG. **5**. Alternatively, the microphone die **414** may be attached to the underside **518** of the lid **404**, or the microphone die **414** may be otherwise disposed within the chamber **514**. Optionally, other circuits, such as an application-specific integrated circuit (ASIC) **520** may be included in the chamber **514** and may be connected to the microphone die **414** by wire bonds **524** or other suitable electrical connections, such as to process signals generated by the microphone die **414**. The substrate **400** defines an acoustic path **528** through at least a portion of the substrate **400**, from the input aperture **218** to the microphone die **414**.

FIG. **6** is a plan view of a portion of the top layer **500** of the substrate **400**. The top layer **500** defines a hole **600** therethrough. Thus, the top layer **500** defines an aperture in the top surface **408** of the finished substrate **400**. The hole **600** is positioned on the layer **500** such that, when the microphone die **414** is attached to the substrate **400**, the hole **600** is in communication with a receiver portion (not shown) of the microphone die **414**, i.e., the hole is locate and oriented such that an acoustic signal exiting the hole enters the receiver portion of the microphone die **414**, although the acoustic signal may pass through a portion of the chamber **514** before reaching the receiving portion of the microphone die **414**. The hole **600** may be centered under the receiver portion of the microphone die **414**, as indicated by center line **530**. Although FIG. **5** shows no gap between the microphone die **414** and the substrate **400**, a gap may be present due, for example, to the thickness of solder, adhesive or other material used to attach the microphone die **414** to the substrate **400**, or due to other structures (not shown) that may be disposed between the microphone die **414** and the substrate **400**.

FIG. **7** is a plan view of a portion of the middle layer **504** of the substrate **400**. The middle layer **504** defines a notch **700** that extends to a side **704** of the layer **504**. The notch **700** is positioned on the layer **504** such that, when the top layer **500** and the middle layer **504** are laminated together, the hole **600** (FIG. **6**) in the top layer **500** registers with, or at least partially overlaps, a portion **708** of the notch **700** to form part of the acoustic path **528**. Another portion **710** of the notch **700** forms all or part of the input aperture **218**.

When the bottom layer **508** is laminated to the middle layer **504**, the bottom layer **508** seals the bottom portion of the acoustic path **528**. Other numbers of layers may be used. For example, several layers similar or identical to the top layer **500** may be laminated together to obtain a desired thickness, strength or for another reason. For example, as shown in FIG. **5**, Insert A, multiple layers **500***a*, **500***b* and **500***c*, each defining a hole, but at staggered locations, may be laminated together to form a curved or slanted portion of the acoustic path **528**. Similarly, multiple layers similar or identical to the middle layer **504** may be laminated together, and multiple bottom layers, similar or identical to the bottom layer **508**, may be laminated together. Although one right-angled acoustic path **528** is shown in FIG. **5**, any number of, or shape, acoustic path(s) may be formed by an appropriate combination of layers with appropriate holes, holes formed at angles, notches, etc. Furthermore, although a right angle transition **534** is shown between the hole **600** in the top layer and the notch **700** in the middle layer, the transition may be angled (as shown in FIG. **5**, Insert B), curved or have another shape.

Returning to FIG. **5**, it can be seen that the sound input port **218** is oriented horizontally, i.e., substantially perpendicular to the vertical orientation (as indicated by the center line **530**) of the receiver portion of the microphone die **414**. Furthermore, the lid **404** and the microphone die **414** are attached to the top surface **408** of the substrate **400**, while an end surface **532** (which may also be referred to as a "side surface") defines the sound input port **218** (which may also be referred to as an "opening").

During manufacture, the substrates **400** should be singulated by laser dicing, by breaking sheets of substrates along score lines or by some other dry method, rather than cutting the substrate sheets with a water-cooled/lubricated saw, to

avoid water infiltration into the microphone package **210**, such as through the sound input port **218**.

In the embodiment shown in FIGS. **5** and **6**, the portion of the acoustic path **528** defined by the substrate **400** terminates under the microphone die **414**. Alternatively, as shown in FIG. **8**, the portion of the acoustic path **528** defined by the substrate **400** may terminate in the chamber **514**, but spaced away from the microphone die **414**. In this case, the top and middle layers **500** and **504** of the substrate are similar to corresponding layers described above, with respect to FIGS. **5-7**, although the location of the hole **600** and of the portion **708** of the notch **700** do not necessarily coincide with the location of the microphone die **414**. In the embodiment shown in FIG. **8**, a portion **800** of the acoustic path **528** extends through the chamber **514**. This portion **800** of the acoustic path **528** is not defined by the substrate **400**. In addition, the receiver portion of the microphone die **414** may be located at or near the top of the microphone die **414**. Some MEMS microphone dies can receive acoustic signals through both ends.

Other numbers of layers, and other configurations of the various layers, may also be used. For example, FIGS. **9** and **10** illustrate another embodiment of the microphone package **210**, in which the substrate **400** includes at least two layers **900** and **904**. The top layer **900** is similar to the top layer **500** shown in FIGS. **5**, **6** and **8**. However, the bottom layer **904** has a channel **1000**, rather than a notch **700** that extends through the entire thickness of the layer **904**, as in the embodiment described with reference to FIG. **7**. (A "channel," as used herein, has a bottom portion, as opposed to a "notch," which is open through, from top to bottom.) Such a channel **1000** obviates the need for a separate bottom layer **508**, although the bottom layer **904** may need to be thicker than a separate bottom layer **508**. As shown in FIG. **9**, the acoustic path **528** may terminate under the microphone die **414**. Alternatively, the hole and the channel **1000** of the substrate **400** may be located so the acoustic path **528** terminates elsewhere under the lid **404**, as described above, with respect to FIG. **8**.

FIGS. **11** and **12** illustrate yet another embodiment of the microphone package **210**, in which the acoustic path **528** is formed, such as by drilling, into a solid or laminated substrate **400**. For example, a first hole **1100** may be drilled partially or completely (as shown in phantom **1104**) through the substrate **400**. A second hole **1108** may be drilled to intersect with the first hole **1100**. If the first hole **1100** is drilled all the way through the substrate **400**, the aperture defined in the bottom surface **1110** of the substrate **400** may be left open and later blocked by a circuit board, to which the microphone package **210** is attached. Alternatively, the aperture may be blocked by applying an adhesive film or tape (shown in phantom **1114**) to a portion of the bottom surface **1110**. As discussed above, in other embodiments, the first hole **1100** is located so it is not under the microphone die **414**. Optionally or alternatively, the holes **1100** and **1108** may be formed as part of an injection molding, punching or other process.

As noted above, if a portion of the acoustic path **528** is open at the bottom of the microphone package **210**, the opening may be blocked by an adhesive film or by a circuit board, to which the microphone package **210** is attached. In yet another embodiment shown in FIGS. **13**, **14**, **15**, **16** and **17**, a circuit board is used to form a wall of the acoustic path **528**. As best seen in FIG. **14**, the bottom portion of the substrate **400** defines an open channel **1400** in communication with a hole **1404** that is in communication with a microphone die (not visible). The channel **1400**, the hole **1404** and other portions of the acoustic path **528** may be formed by drilling, routing, molding, laminating layers or another process, as described above. The acoustic path **528** may terminate under the microphone die **414** or elsewhere, as described above.

As best seen in FIG. **15**, when the microphone package **210** is attached to a circuit board **1500**, the open bottom portion of the channel **1400** is blocked by the circuit board **1500**. As noted, the microphone package **210** may be attached to the circuit board by solder, an adhesive, or by any other suitable material or method. As best seen in FIG. **14**, the bottom surface **1408** of the substrate **400** may include a solderable portion **1410** that follows the outline shape of the channel **1400**. The solderable portion **1410** should be configured into an open shape, such as a "U" shape. The solderable portion **1410** should extend from a location proximate (but not necessarily all the way to) an edge **1412** of the substrate **400** to another location proximate the same or another edge of the substrate **400**. Thus, the solderable portion **1410** should substantially bound the open portion of the channel **1400** between the solderable portion **1410** and the edge(s) **1412** of the substrate **400**, i.e., collectively, the solderable portion **1410** and the edge(s) **1412** form a boundary around the open portion of the channel **1400**. As can be see in FIG. **14**, the solderable portion **1410** does not completely surround the channel **1400**.

As shown in FIG. **17**, the circuit board **1500** may also include a solderable portion **1700**, shaped similar to the solderable portion **1410** on the bottom of the microphone package **210**. When the microphone package **210** is soldered to the circuit board **1500**, the solder **1600** (FIG. **16**) takes the shape of the solderable portions **1410** and **1700** and, thus, forms the lower portion of a vertical wall of the acoustic path **528**. Conventional contact pads **1414** (FIG. **14**) provide electrical and additional mechanical connections between the microphone package **210** and the circuit board **1500**.

Instead of solder, an adhesive, such as a conductive adhesive, may be used. In this case, the solderable portions **1400** and **1700** may be omitted, or they may be replaced by materials chosen for compatibility with the adhesive.

As shown in FIGS. **18**, **19**, **20** and **21**, a circuit board **1800** may define a channel **1804**, and a bottom-ported microphone package may be mounted on the circuit board **1800**. If the microphone package is attached to the circuit board **1800** by solder, the bottom of the microphone package may include a solderable portion **1410**, as described above, with respect to FIG. **14**.

Alternatively, as shown in FIGS. **22** and **23**, a circuit board **2200** may include a notch **2204** (best seen in FIG. **23**), rather than a channel **1804**. A notch may be easer to form than a channel. The open bottom of the notch **2204** may be sealed with an adhesive film or tape **2208**.

Another embodiment combines aspects of the microphone package described with reference to FIG. **15** with aspects of the circuit board described with reference to FIG. **21** or FIG. **23**. That is, both the substrate **400** of the microphone package and the circuit board **1800** or **2200** may define channels and/or notches.

In yet another alternative embodiment, illustrated in FIG. **24**, the top layer **500** of the substrate **400** defines an aperture **2400**, which functions as the input aperture to the microphone package. Although the aperture **2400** is not on the side **2404** of the substrate **400**, the aperture **2400** is closer to the side **2404** than an aperture defined in the side **2408** of the lid **404** would be. A suitable gasket should be used with the microphone package **210** illustrated in FIG. **24**, as discussed with reference to the gasket **220** shown in FIGS. **1-3**.

In another embodiment, illustrated in FIGS. **25** and **26**, the substrate **400** defines a channel **2500**, which forms at least

part of the acoustic path **528**. The comments regarding a gasket made in reference to FIG. **24** also apply to this embodiment.

FIGS. **27** and **28** illustrate a microphone package **2700**, according to yet another embodiment of the present invention. FIG. **27** shows the microphone package **2700** in assembled and exploded views. The microphone package **2700** includes a top layer or lid **2704** and a wall layer **2708**. The wall layer **2708** may be made of one or more individual layers, as exemplified by dashed line **2709**. The microphone package **2700** also includes a channel layer **2714** that defines a hollow channel **2716**. The channel **2716** extends from an edge **2718** of the channel layer **2714** to a location away from the edge **2718**, such as to a location over which a microphone die (not shown) may be attached. The channel layer **2714** may be made of one or more individual layers, as exemplified by dashed line **2720**.

A bore layer **2710** defines a hole **2720** and is disposed intermediate the wall layer **2708** and the channel layer **2714**, such that when the channel layer **2714** and bore layer **2710** are assembled in registration with each other, the hole **2720** is in communication with the channel **2716**. When the wall layer **2708** and the bore layer **2710** are assembled in registration with each other, the hole **2720** is in communication with a volume defined by the wall layer **2708**.

The microphone package **2700** (or any other embodiment) may be provided as a kit of parts that includes the various layers **2704**, **2708**, **2710** and **2714** described above. The layers **2704-2714** may be made of plastic, ceramic or any other suitable material. The layers may be assembled using any suitable technique. For example, plastic layers may be assembled using an adhesive or welding by heat, ultrasonic energy or solvent, and ceramic layers may be assembled by brazing.

FIG. **29** is a flowchart that describes producing a side-ported MEMS microphone package, according to one embodiment of the present invention. At **2900**, at least part of an acoustic path is defined through at least a part of a substrate to an aperture that is, or will be, under a lid, so the acoustic path is oriented other than perpendicular to the substrate surface to which the lid is, or will be, attached. As noted, the acoustic path may be defined in various ways, including laminating layers that each defines one or more holes, notches, channels, etc. For example, as indicated at **2904**, a first layer that defines the aperture may be laminated to a second layer that at least partially defines the acoustic path, such as by a notch, channel, etc. Optionally or alternatively, as indicated at **2906**, one or more holes may be bored at least partially through the substrate to define at least part of the acoustic path.

Optionally or alternatively, as indicated at **2910**, a second aperture may be defined in a substrate surface that is perpendicular to the surface to which the lid is, or will be, attached. The second aperture is in communication with the acoustic path.

Optionally or alternatively, as indicated at **2914**, a second aperture may be defined in the surface of the substrate. The second aperture is in communication with the acoustic path.

Optionally or alternatively, as indicated at **2916**, a second aperture may be defined in the surface of the substrate opposite the surface to which the lid is, or will be, mounted. The second aperture is in communication with the acoustic path.

Optionally, a solderable portion may be provided on the surface of the substrate opposite the surface to which the lid is, or will be, mounted, as indicated at **2920**. As noted at **2924**, the solderable portion may be in a U shape.

At **2926**, a MEMS microphone die is attached to the surface of the substrate. At **2930**, a lid is attached to the substrate surface to cover the microphone die and the aperture defined in the surface.

As noted, a circuit board may define a portion of the acoustic path. FIG. **30** is a flowchart that describes producing a side-ported MEMS microphone assembly, according to another embodiment of the present invention. At **3000**, an aperture is defined in a surface of a circuit board adjacent a present or expected future location of the bottom port of a MEMS microphone package. At **3004**, an acoustic path is defined from a side of the circuit board, through at least a portion of the circuit board, to the aperture. The acoustic path may be defined in various ways, including laminating layers that each defines one or more holes, notches, channels, etc. For example, at **3006**, a first layer of the circuit board is laminated to a second layer of the circuit board. The first layer defines the aperture. The second layer defines at least a portion of a hollow channel in communication with the aperture. Optionally or alternatively, as indicated at **3010**, one or more holes may be bored at least partially through the circuit board to define at least part of the acoustic path.

Optionally, a solderable portion may be provided on the surface of the circuit board, as indicated at **3014**. The solderable portion is configured in an open shape, such as a U shape, and the solderable portion extends from a location proximate an edge of the circuit board to another location proximate the edge of the circuit board.

At **3016**, a bottom side of a bottom-ported MEMS microphone package is attached to the surface of the circuit board, so the bottom port of the microphone package is in communication with the aperture in the surface of the circuit board.

While the invention is described through the above-described exemplary embodiments, it will be understood by those of ordinary skill in the art that modifications to, and variations of, the illustrated embodiments may be made without departing from the inventive concepts disclosed herein. For example, although some aspects of producing side-ported microphone packages and assemblies have been described with reference to flowcharts, those skilled in the art should readily appreciate that functions, operations, decisions, etc. of all or a portion of each block, or a combination of blocks, of the flowchart may be combined, separated into separate operations or performed in other orders. In addition, although a microphone package has been described as being made from various materials, the disclosed methods and structures may be used with other materials. Furthermore, disclosed aspects, or portions of these aspects, may be combined in ways not listed above. Accordingly, the invention should not be viewed as being limited to the disclosed embodiment(s).

What is claimed is:

1. A side-ported microphone package, comprising:

a substrate having a first surface and a second surface, the second surface being on substantially an opposite side of the substrate from the first surface;

a lid attached to the first surface of the substrate to define a chamber therebetween; and

a microphone die disposed within the chamber;

wherein:

the second surface defines an opening;

the substrate at least partially defines an acoustic path between the opening and the microphone die; and

the second surface comprises a solderable portion configured into an open shape, extending from a location proximate an edge of the substrate to another location proximate an edge of the substrate, so as to substantially bound the opening between the solderable por-

tion and the edge of the substrate, wherein the solderable portion does not surround the opening.

2. A side-ported microphone package according to claim **1**, wherein at least a portion of the acoustic path is oriented parallel to the first surface of the substrate.

3. A side-ported microphone package according to claim **1**, wherein:

a locus of points where the lid is attached to the substrate defines a boundary;

the opening is outside the boundary;

the substrate defines:

an aperture in the first surface within the boundary; and

a hollow passage in communication with the aperture and with the opening, such that the hollow passage, the aperture and the opening form at least a portion of the acoustic path.

4. A side-ported microphone package according to claim **3**, wherein the opening is defined at least in part by a surface of the substrate that is substantially perpendicular to the first surface.

5. A side-ported microphone package according to claim **1**, wherein the solderable portion is U shaped, with ends of the U shape proximate the edge of the substrate.

6. A side-ported microphone package according to claim **3**, wherein:

the substrate comprises a laminate;

a first layer of the laminate defines the aperture; and

a second layer of the laminate defines at least a portion of the hollow passage.

7. A side-ported microphone package according to claim **3**, wherein the aperture is adjacent to, and in line with, a receiver portion of the microphone die.

8. A side-ported microphone package according to claim **3**, wherein the microphone die is disposed between the aperture and the chamber.

9. A side-ported microphone package according to claim **3**, wherein the aperture is in communication with the chamber.

10. A side-ported microphone package according to claim **1**, wherein the substrate comprises a printed circuit board.

11. A side-ported microphone package according to claim **1**, wherein the substrate comprises ceramic.

12. A side-ported microphone package according to claim **1**, wherein the substrate comprises plastic.

13. A side-ported microphone package according to claim **1**, wherein the substrate comprises at least one wall, and the lid is attached to the at least one wall.

14. A kit of parts for assembling a side-ported microphone package, the kit comprising:

a first layer defining at least one of a hollow channel and a notch in a surface of the first layer, the hollow channel or notch, as the case may be, extending from an edge of the first layer and terminating at a location away from any edge of the first layer, the surface of the first layer comprises a solderable portion configured into an open shape, extending from a location proximate the edge of the first layer to another location proximate the edge of the first layer, so as to substantially bound the channel or notch between the solderable portion and the edge of the first layer, wherein the solderable portion does not surround the channel or notch;

a second layer defining a bore, such that when the first and second layers are assembled in registration with each other, the bore is in communication with the channel or notch;

at least one wall configured such that when the at least one wall is attached to the second layer in registration therewith, the bore is in communication with a volume defined by the at least one wall; and

a lid.

15. A kit of parts as recited in claim **14**, wherein the first layer comprises a printed circuit board, and the second layer comprises a printed circuit board.

16. A kit of parts as recited in claim **15**, wherein the lid comprises a printed circuit board.

17. A kit of parts as recited in claim **14**, wherein the lid comprises a printed circuit board.

18. A kit of parts as recited in claim **14**, wherein the first layer, the second layer, the at least one wall and the lid comprise ceramic.

19. A kit of parts as recited in claim **14**, wherein the first layer, the second layer, the at least one wall and the lid comprise plastic.

20. A method for producing a side-ported microphone package, the method comprising:

disposing a microphone die, relative to a substrate;

attaching a lid to a first surface of the substrate to define a chamber within which at least one of:

the microphone die is disposed and

the microphone die will be disposed;

defining an opening in a second surface of the substrate, outside the chamber, the second surface being on a substantially opposite side of the substrate from the first surface;

providing a solderable portion on the second surface, the solderable portion being configured into an open shape, extending from a location proximate an edge of the substrate to another location proximate an edge of the substrate, so as to substantially bound the opening between the solderable portion and the edge of the substrate, wherein the solderable portion does not surround the opening;

defining at least a portion of an acoustic path from the opening, through at least a portion of the substrate, to an aperture defined by the first surface within a boundary defined by a locus of points where at least one of the lid is attached and the lid will be attached to the substrate.

21. A method according to claim **20**, wherein defining the at least a portion of the acoustic path comprises laminating a plurality of layers of the substrate, wherein a first layer of the substrate defines the first aperture; and a second layer of the substrate defines at least a portion of the acoustic path.

22. A method according to claim **20**, wherein defining the at least a portion of the acoustic path comprises boring at least one hole at least partially through the substrate.

23. A method according to claim **20**, wherein the solderable portion is U shaped, with ends of the U shape proximate the edge of the substrate.

24. A method for producing a side-ported microphone assembly, the method comprising:

attaching a bottom side of a bottom-ported microphone package to a surface of a circuit board;

defining an opening in the surface of the circuit board adjacent at least one of a present and an expected future location of the bottom port of the microphone package;

defining a hollow acoustic path from a side of the circuit board, through at least a portion of the circuit board, to the opening; and

providing a solderable portion on the surface of the circuit board, the solderable portion being configured into an open shape, extending from a location proximate an edge of the circuit board to another location proximate

an edge of the circuit board, so as to substantially bound the opening between the solderable portion and the edge of the circuit board, wherein the solderable portion does not surround the opening.

25. A method according to claim **24**, wherein defining the acoustic path comprises laminating a plurality of layers of the circuit board, wherein a first layer of the circuit board defines the opening, and a second layer of the circuit board defines at least a portion of a hollow passage in communication with the opening and extending to the side of the circuit board.

26. A method according to claim **24**, wherein defining the acoustic path comprises boring at least one hole at least partially through the circuit board, the hole being in communication with the opening.

* * * * *

Exhibit PA-U

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(12) **United States Patent**
Nemirovski

(10) **Patent No.: US 6,671,379 B2**
(45) **Date of Patent: *Dec. 30, 2003**

(54) **EAR MICROPHONE APPARATUS AND METHOD**

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(73) Assignee: **Think-A-Move, Ltd.**, Cleveland, OH (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

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(22) Filed: **Jul. 31, 2001**

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(60) Provisional application No. 60/280,282, filed on Mar. 30, 2001.

(51) **Int. Cl.**[7] **H04B 15/00**
(52) **U.S. Cl.** **381/93**; 381/318; 381/312
(58) **Field of Search** 381/93, 318, 312; 704/270

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Assistant Examiner—Elizabeth McChesney
(74) *Attorney, Agent, or Firm*—Renner, Otto, Boisselle & Sklar, LLP

(57) **ABSTRACT**

An earset including a housing positionable with respect to an ear of a person, a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure and a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal. The earset also includes a circuit coupled to receive the microphone signal and the speaker signal and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker to produce a corrected microphone signal.

17 Claims, 3 Drawing Sheets

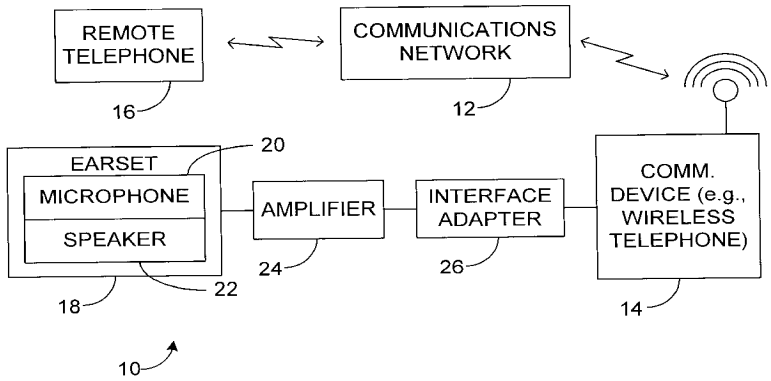


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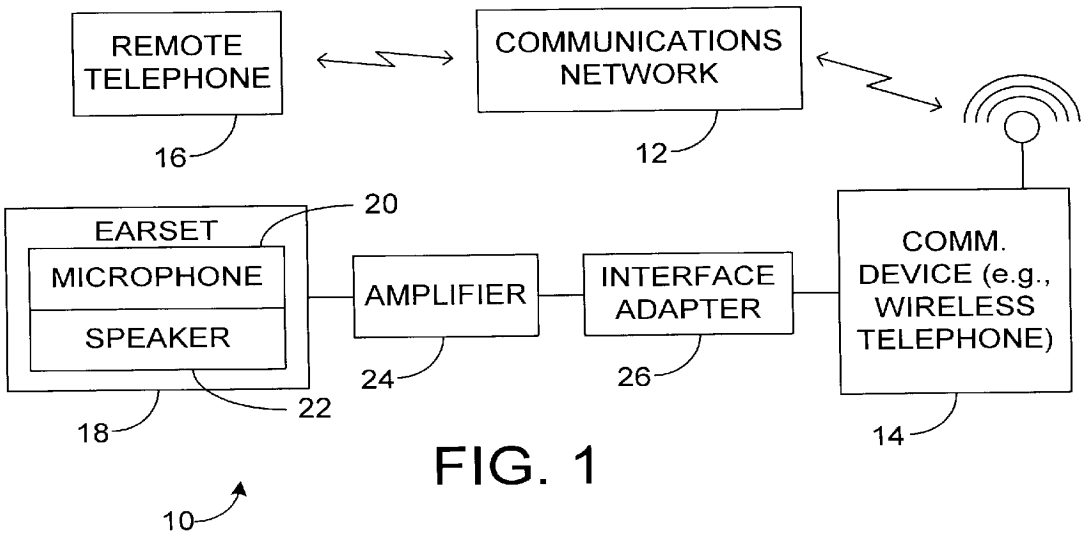

REMOTE TELEPHONE
16
COMMUNICATIONS NETWORK
12
EARSET
20
MICROPHONE
SPEAKER
18
22
AMPLIFIER
24
INTERFACE ADAPTER
26
COMM. DEVICE (e.g., WIRELESS TELEPHONE)
14
10

FIG. 1

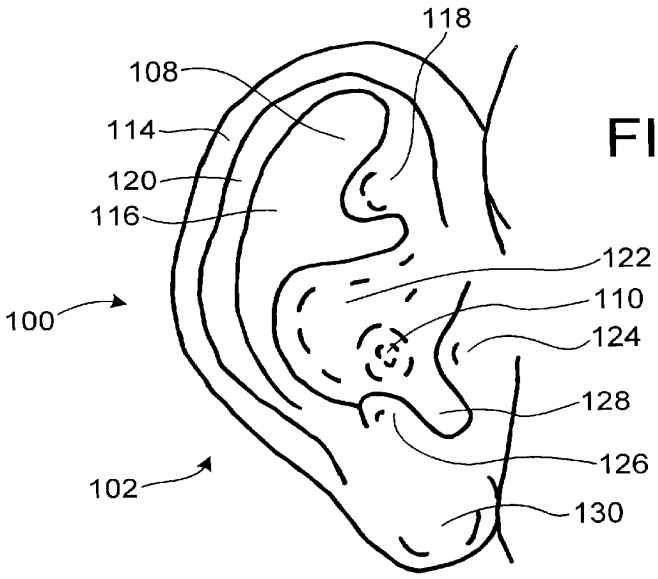

118
108
114
120
116
122
110
124
100
128
126
102
130

FIG. 2

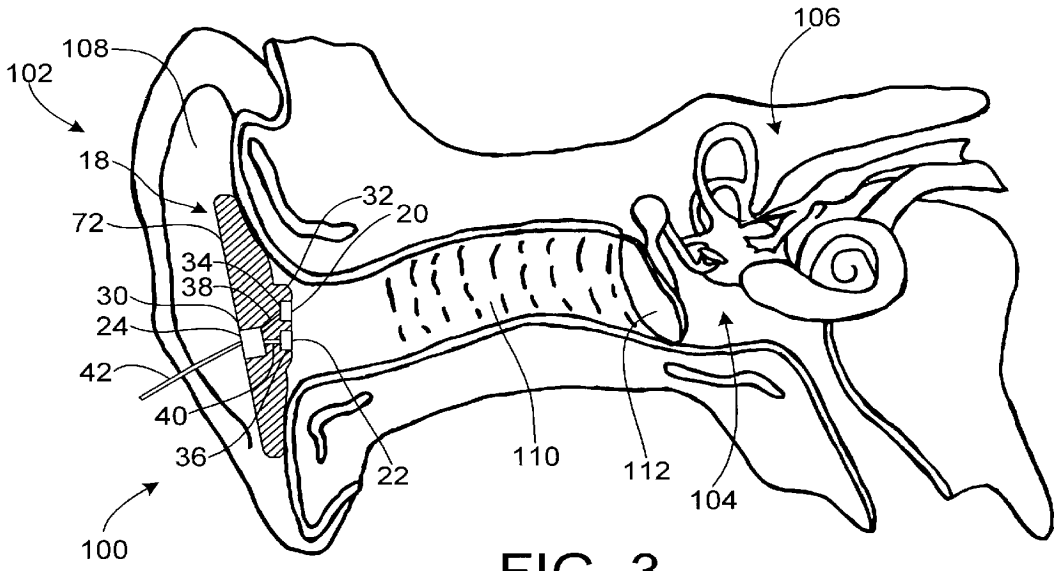

106
108
102
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20
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38
30
24
42
40
36
22
110
112
104
100

FIG. 3

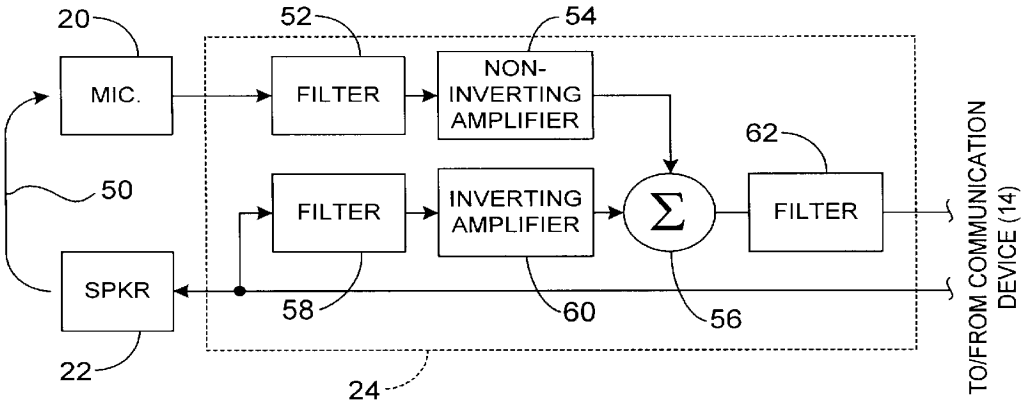

20
MIC.
52
FILTER
54
NON-INVERTING AMPLIFIER
62
50
FILTER
INVERTING AMPLIFIER
Σ
FILTER
SPKR
58
60
56
22
24
TO/FROM COMMUNICATION DEVICE (14)

FIG. 4

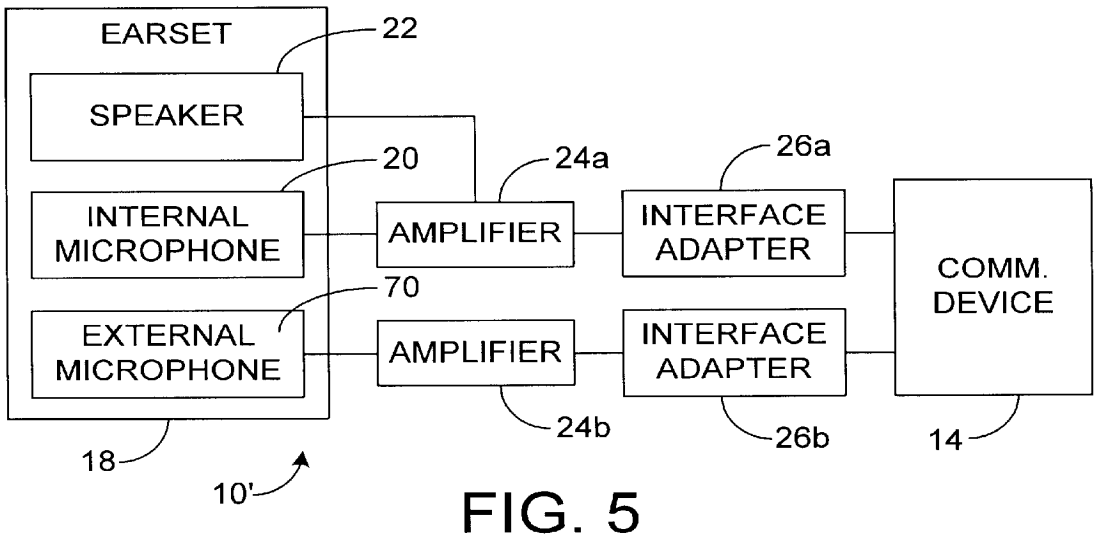

EARSET
22
SPEAKER
20
INTERNAL MICROPHONE
24a
AMPLIFIER
26a
INTERFACE ADAPTER
COMM. DEVICE
70
EXTERNAL MICROPHONE
AMPLIFIER
INTERFACE ADAPTER
24b
26b
14
18
10'

FIG. 5

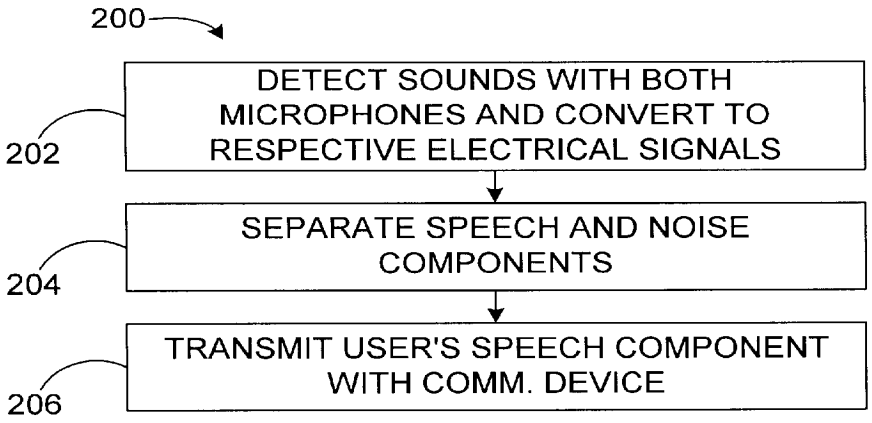

200
DETECT SOUNDS WITH BOTH MICROPHONES AND CONVERT TO RESPECTIVE ELECTRICAL SIGNALS
202
SEPARATE SPEECH AND NOISE COMPONENTS
204
TRANSMIT USER'S SPEECH COMPONENT WITH COMM. DEVICE
206

FIG. 6

EAR MICROPHONE APPARATUS AND METHOD

RELATED APPLICATION(S)

This application is a continuation-in-part of U.S. application Ser. No. 09/897,696, filed Jul. 2, 2001, the entire disclosure of which is incorporated herein by reference, and this application claims the benefit of U.S. Provisional Application No. 60/280,282, filed Mar. 30, 2001, the entire disclosure of which is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention generally relates to a sensor placed with respect to an ear. More particularly, the sensor is used to detect speech and other oral vocalizations of a person with a microphone as well as deliver sound to the ear of the person with a speaker. Feedback from the speaker to the microphone is reduced.

BACKGROUND OF THE INVENTION

Wireless mobile telephones, also referred to as a cellular telephones, have become exceedingly popular communication devices. However, the vast majority of cellular telephones contain a transceiver (e.g., a radio frequency, or RF, transceiver) for establishing a communication link with a remote location, such as a cell phone tower. In order to carry out a conversation with another person using the wireless telephone, the user must hold the telephone adjacent the ear and mouth of the user. This presents the disadvantage of occupying the use of at least one of the user's hands. In many situations, hands free use of the wireless telephone is desirable. The same is true for receivers, or handsets, found in hard-wired telephone systems. For example, the user of a hard-wired telephone system may wish to type on a computer while speaking on the phone. In addition, medical professionals and others have expressed concerns relating to the health of wireless telephone users who engage in prolonged use of an RF transceiver adjacent their head.

There are many commercially available headsets available to users of wireless and/or hard-wired telephone systems. These headsets are intended to assist the user in carrying out a conversation without the use of the user's hands and to locate the telephone (and any RF transceiver) away from the user's head. These headsets typically include an ear piece containing a speaker. The ear piece can be removably placed with respect to the user's ear and broadcasts sounds to the user's ear. The headsets also typically include a microphone disposed on a support member that positions the microphone with respect to the users mouth. The microphone is used to detect speech and other vocalizations emanating from the mouth of the user. The detected sounds are converted into an electrical signal and transmitted by the telephone to a backbone telecommunications network and on to the telephone of another person. In this manner the user can carry out a fully duplexed conversation with the other person.

However, the headsets can be cumbersome to use. More particularly, care must be taken to ensure that the microphone is properly positioned and that the microphone maintains that position. The need to adjust the headset during a conversation can be distracting to the user. In addition, improper positioning of the microphone may lead to poor and/or unreliable detection of the user's speech. This problem is compounded by the common occurrence of the microphone detecting environmental noise, such as the sound of a passing vehicle, conversations taking place near the user and the like. The detected environmental noise is ultimately transmitted by the telephone.

Accordingly, there exists a need in the art for an easy to use telephone headset that accurately detects the user's speech, even when speaking at a low volume, and reduces the detection of environmental noise.

SUMMARY OF THE INVENTION

According to one aspect of the invention, an earset including a housing positionable with respect to an ear of a person; a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure; a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal; and a circuit coupled to receive the microphone signal and the speaker signal and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker to produce a corrected microphone signal.

According to another aspect of the invention, a method of detecting speech of a person including the steps of placing a microphone at least partially within an ear of the person; detecting a change in air pressure within the ear of the person using the microphone, the change in air pressure corresponding to the speech of the person; producing an electrical microphone signal corresponding to the internally detected change in air pressure; placing a speaker with respect to the ear of the person; generating a sound with the speaker corresponding to an electrical speaker signal; and generating a corrected microphone signal by reducing a feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound generated by the speaker to produce a corrected microphone signal.

According to yet another aspect of the invention, a method of enhancing a signal representing a detected first sound pattern, including directly coupling the signal representing the detected first sound pattern with a signal for producing a second sound pattern.

BRIEF DESCRIPTION OF THE DRAWINGS

These and further features of the present invention will be apparent with reference to the following description and drawings, wherein:

FIG. **1** is a block diagram illustrating a communications system according to the present invention;

FIG. **2** is a schematic view of an ear;

FIG. **3** is a cross-sectional view of an ear microphone assembly, or earset, disposed with respect to the ear of a person, and the ear also shown in cross-section;

FIG. **4** is a block diagram of the ear microphone assembly;

FIG. **5** is a block diagram of an ear microphone assembly according to another aspect of the invention; and

FIG. **6** is a flow chart of a method of processing signals generated by the ear microphone assembly of FIG. **5**.

DETAILED DESCRIPTION OF THE INVENTION

The following is a detailed description of the present invention made in conjunction with the attached figures,

wherein like reference numerals will refer to like elements throughout. To illustrate the present invention in a clear and concise manner, the drawings may not necessarily be to scale and certain features may be shown in somewhat schematic form.

The present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly. The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone.

Additionally, the earset assembly of the present invention allows a user to speak more quietly (e.g., such as at a whisper or near whisper) than with conventional headsets. This allows for more private conversations and less disruption to others. There is also a body of evidence indicating that the softer one speaks, the less concentration is needed to maintain the conversation, thereby allowing the individual at least to partially engage in other activities while speaking.

The earset of the present invention does not rely on directionality of the microphone or a detection of sound that has emanated from the user's mouth. Therefore, the need to repeatedly adjust the position of the earset that would otherwise distract the user and require the use of the user's hands. Also, the size and arrangement of the earset is small, resulting in a more cosmetically appealing device. Such a device can be used unobtrusively. For example, the device would not be noticed as much by others when used in public, or by a person being observed by others, such as, for example, a television news anchor or a secret service agent.

It is noted that the term air pressure changes is used in its broadest sense and includes, for example, sound waves (whether audible to the user or not), pressure fluctuations, vibrations, resonations and the like. In addition, the term air pressure changes as used herein includes vibrations conducted by the bones and tissue of the user that are carried to ear. These conducted vibrations can vibrate the housing leading to sound detection by the microphone. The conducted vibrations may also cause anatomical parts of the ear to resonate, and/or amplify sounds, thereby leading to sound detection by the microphone. The air pressure changes may be caused by one or more factors, including vibrations of the ear drum, vibrations of bone within the ear, vibrations of other anatomical structures within the ear and vibrations conducted by the bones and/or tissue of the user to the ear and which invoke an air pressure change in the ear.

As a result, the sensor can be used to detect a person's speech. It is also noted that the term speech is used in its broadest sense and includes spoken words and utterances, as well as other vocalizations produced by the user, including, for example, grunts, whistles, singing, coughs, "clicking" sounds made by movement of the lips or tongue, and the like. To facilitate the description herein, the event of sensing or detecting by the microphone will be referred to as detecting and that which is detected will be referred to as a change within the ear, or simply an air pressure change. The present invention monitors changes within the human ear which occur instantaneously, or nearly instantaneously, in response to the speech of the person to provide a substantially real-time speech detection system.

Turning now to the figures, FIG. **1** is a block diagram which illustrates a communications system **10** for establishing duplexed (two-way) audio communication between two or more individuals. The communications system **10** includes a communications network **12**, or backbone network. The communications network **12** establishes communication links with at least one communication device **14**, thereby enabling a user of the communication device **14** to carry on a conversation with a remotely located person using a remotely located telephone **16** as is known in the art. In the illustrated embodiment, the communication device **14** and the remote telephone **16** are wireless telephones, such as cellular telephones. However, one skilled in the art will appreciate that the communication device **14** and/or the remote telephone **16** can be other types of devices, including hardwired (land line) telephones, radios, personal digital assistants (PDAs), portable or stationary computers, etc. The communications network **12** can also be alternative type of network, such as the Internet, a WAN, or a LAN.

The communications system **10** includes an ear microphone assembly, or earset **18**, having a microphone **20** and a speaker **22** supported by a housing **30** (FIG. **3**). The physical arrangement of the earset **18** will be described in more detail below. The microphone **20** is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone **20** converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier **24**. Next, the amplified electrical signal is received by the communication device **14** via an appropriate interface adapter **26**. The interface adapter **26** can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device **14**.

The speaker **22** is used to transmit (i.e., broadcast) sounds to the user. These sounds include sounds generated in response to signals received by the communication device **14** over the communications network **12**. In this way, the earset **18** and communication device **14** can be used as a bi-directional communication apparatus.

In one embodiment, the earset **18** is coupled to the communication device using three conductors. A first of the conductors is used as a common ground. A second of the conductors carries electrical signals to the speaker **22** to be converted into sound waves. The third of the conductors carries electrical signals representing detected sounds from the microphone **20** to the communication device **14** for transmission to the communications network **12**. The third conductor also carries power for the microphone **20** and amplifier **24**. In this embodiment, the interface adapter **26** cen be implemented with a 2.5 mm jack as is known in the art.

Referring to FIGS. **2** and **3**, an external view and a cross-sectional view of an ear **100** are respectively illustrated. FIG. **3** also shows the earset **18** in cross-section. According to Henry Gray's famous text "Anatomy", the human ear is divided into three parts, including the external ear **102**, the middle ear (or tympanum) **104** and the internal ear (or labyrinth) **106**. The middle ear **104** and the internal

ear **106** will not be described in great detail herein. The external ear **102** includes an expanded portion, or a pinna **108** (also referred to as an auricle), and an ear canal **110** (also referred to as a meatus or auditory canal). The pinna **108** serves to collect vibrations of the air surrounding the person's head. The ear canal **110** conducts those vibrations to the tympanum, or ear drum **112**.

The pinna **108** has a generally ovoid form with a larger end directed upward and having an outer surface that is irregularly concave and directed slightly forward. The pinna **108** has a number of eminences and depressions. Typically, the ear **100** has a prominent and curved rim, or helix **114**. Generally parallel to the helix **114** is another curved prominence, or antihelix **116**. The antihelix **116** bifurcates to form a triangular depression, or a fossa of the antihelix **118** (also referred to as a fossa triangularis). A narrow, curved depression located between the helix **114** and antihelix **116** is referred to as fossa of the helix, or scapha **120**. The antihelix **116** also curves around a deep, capacious cavity, or the concha **122** (the concha **122** being divided by the commencement of the helix **114**, or crus helicis, into an upper part, termed the cymba conchae, and a lower part, termed the cavum conchae). The concha **122** leads inward to an opening of the ear canal **110**. In front of the concha **122** and projecting backward (usually over the opening of the ear canal **110**) is a pointed eminence, or tragus **124**. Opposite the tragus **124** is a tubercle, or antitragus **126**. A notch-like depression, or incisura intertragica **128**, is disposed between the tragus **124** and antitragus **126**. A lobule **130** is present under the tragus **124** and antitragus **126**.

The ear canal **110** is an oval cylindrical passage extending from a bottom of the concha **122** to the ear drum **112**. The ear canal **110** is about an inch and a half in length when measured from the tragus **124** to the ear drum **112**. When measured from the bottom of the concha **122** to the ear drum **112**, the ear canal is about an inch long. The ear canal **110** forms a gradual "S-shaped" curve and is directed, at first, inward, forward and slightly upward (i.e., pars externa). The ear canal **110** then passes inward and backward (i.e., pars media) and then passes inward, forward and slightly downward (i.e., pars interna).

It is not certain what physical, chemical or neural mechanism causes or generates the changes in air pressure in or near the ear or sounds to some from the ear in response to various actions of the user. However, due to the connection of the oral cavity to the ear via the eustachian tube, speech and movements of the mouth may cause an air pressure, a change in air pressure or an air flow to or from the ear leading to a detectable air pressure which can be detected by the microphone **20**. Regardless of the exact physical, chemical or neural mechanism, empirical testing has confirmed that the user's speech generates pressure changes in or near the ear of the person. Consequently, the air pressure changes can be monitored near the ear and used to detect the speech of a user.

The present invention uses various forms of the terms "changes in air pressure", "changes within the ear" and "sounds emanating or coming from the ear" in their broadest sense to characterize the parameter being measured. Changes in air pressure may alternatively be characterized as sound waves. These sound waves (or vibrations) may propagate through mediums other than air, such as bone and tissue. As is well known by those skilled in the art, as a sound wave spreads out from its source its intensity falls off (the energy per unit area decreases with the inverse square of the distance), but the total energy is constant. Consequently, it is desirable to have the microphone **20** sufficiently close to the source of the sound wave so that the intensity level of the detected changes will be as large as possible.

FIG. **3** illustrates the earset **18** inserted at least partially into the ear **100** of a person (i.e., at least within the cavity defined by the pinna **108**, if not deeper within the ear **100** such as within the concha **122**, at the opening of the ear canal **110** or slightly into the ear canal **110**).

The earset **18** includes a housing **30**, the microphone **20**, the speaker **22** and the amplifier **24** (optionally the amplifier **24** can be located remotely from the earset **18** or as part of the communication device **14**). The housing **30** can take on a number of different configurations. The housing **30** can resemble the housing design of a hearing aid, and particularly a digital hearing aid, for similar insertion, or partial insertion, into the ear **100** and/or ear canal **1 10**. Alternatively, the housing **30** can resemble a miniature ear phone as found in conventional wireless telephone headsets or as used with personal audio/music players. The earset **30** can be retained by insertion into the ear **100**, by a member disposed over or hanging from the ear and/or by a headset assembly.

The illustrated housing **30** is made from any suitable material, such as plastic, rubber or a gel-like material. Optionally, the housing **30**, or portions thereof, can be made of pliable material, sound absorbing (or sound proofing) material and/or include sound insulating material such as foam.

The housing **30** is wider than an opening of the ear canal **110** and engages the pinna **108**. In one embodiment, the housing **30** fits within the concha **122** and is retained, at least in part, by the tragus **124** and/or the antitragus **126**. Such arrangement at least partially insulates the microphone **20** from externally generated noise and air pressure changes.

The microphone **20** is disposed on an interior portion **32** of the housing **30**. Accordingly, the interior portion **32** is sized and shaped to position the microphone **20** with respect to the ear **100** and/or ear canal **110** as desired. In one embodiment, the interior portion **32** positions the microphone **20** and/or the speaker **22** in the concha **122**. In another embodiment, the interior portion **32** positions the microphone **20** and/or the speaker **22** at the opening of the ear canal **110** where the ear canal **110** meets the concha **122**. It should be appreciated that the interior portion **32**, microphone **20** and/or the speaker **22** need not enter the ear canal **110**. In yet another embodiment, the interior portion **32** extends into the ear canal **110** and positions the microphone **20** and/or the speaker **22** within the ear canal **110**.

The housing **30** can be custom designed for the individual to form a close and comfortable fit with the ear of the individual. Alternatively, the housing can have a standard, or "stock", design for all individuals which is fabricated in a number of sizes. As one skilled in the art will appreciate, many alternative configurations for the housing **30** are possible and each are considered to fall within the scope of the present invention.

The interior portion **32** of the housing **30** has a recess **34** in which the microphone **20** is placed. Preferably, the air adjacent the ear drum **112** and/or the air in the ear canal **110** will be in fluid communication with the air adjacent the microphone **20**. In this way, sounds or air pressure changes generated within the ear **100** will be transmitted directly to the microphone **20** through an air medium. Other sounds conducted by the bones and tissue of the user may be transmitted to the microphone **20** mechanically via the housing **30**. In one embodiment, the microphone **20** is held

in place with an adhesive. The microphone **20** in one embodiment is less than two mm in size. In an alternative embodiment, the microphone **20** is disposed within a hollow cavity defined by the housing **30** and an opening in a wall of the housing **30** defining the cavity is provided to establish fluid communication of the air inside the cavity and the air inside the ear canal **110**.

By inserting the microphone **20** in the ear **100** and/or ear canal **110**, the microphone **20** is a least partially shielded from environmental noise. More specifically, the housing **30** and the head of the user at least partially blocks externally generated sound waves before reaching the microphone **20**.

In the illustrated embodiment, the microphone **20** is an omnidirectional microphone.

It should be appreciated that the microphone **20** can be moved closer to or further away from various anatomical structures within the ear **100** as desired for factors such as comfort and to optimize detection of the user's speech. It is noted that the microphone **20** is capable of detecting speech of the user speaking at a variety of volumes.

The microphone **20** can take on a number of different forms, such as a silicon or semiconductor microphone (also referred to as a pressure sensor) such as the devices disclosed in U.S. Pat. Nos. 4,533,795, 4,922,471, and 6,156,585, each of which are incorporated herein by reference in their entireties. It is noted that the foregoing devices are exemplary and other electroacoustic transducers from a number of different vendors can be used.

It has been shown that the use of one earset **18** with the microphone **20** in one of the ears **100** of an individual can be used to detect speech such that the resulting transmission by the communication device **14** allows the remotely located person using the remote telephone **16** to hear the speech of the user. However, empirical testing has shown that by using two such earsets, one in each of the ears of an individual, the performance of system **10** is improved due to the detection of more information. It is noted that in certain circumstances, the use of an earset **18** in both ears is discouraged (such as when using a cellular telephone while driving a car). It is noted that the circuitry, processing and logic routines described herein can be modified to process signals from a second earset **18**. Such modifications will be apparent to one of ordinary skill in the art.

The interior portion **32** of the housing **30** has a second recess **36** in which the speaker **22** is placed. Preferably, the speaker **22** broadcasts sounds corresponding to the speech of the remotely located person using the remote telephone **16** and received by the communication device **14**. The sounds are a reproduction of the remotely located person's speech and are made audible to the user by the speaker **22**.

The microphone **20** is electrically coupled to the amplifier **24** with a conductor **38**. The speaker **22** is also electrically coupled to the amplifier **24** with a conductor **40**. The amplifier **24** is electrically coupled to the interface adapter **26** (FIG. **1**) with a conductor **42**. More specifically, the conductor **42** carries the amplified signals from the microphone **20** to the interface adapter **26** and carries signals to be converted into sound from the interface adapter **26** to the speaker. As one skilled in the art will appreciate, the amplifier **24** can be omitted or by-passed and the microphone **20** and/or speaker **22** can be coupled directly to the interface adapter **26** by way of a conductor or conductors.

A block diagram of the ear microphone assembly, and particularly the amplifier **24**, is illustrated in FIG. **4**. Since the microphone **20** is placed in close proximity to the speaker **22**, the microphone **20** will tend to detect sounds generated by the speaker **22**. These detections are in addition to detections of the user's speech. The sounds that have been generated by the speaker **22** and detected by the microphone **20** will be referred to herein as feedback and is represented in FIG. **4** by arrow **50**. In one embodiment, the sensitivity of the earset **18** is tuned to minimize detection of sounds from the speaker **22** and/or to minimize the delivery of a feedback component to the communication device **14**. For example, a potentiometer can be used to adjust the gain of the amplifier **24** as desired. Physical separation of the speaker **22** and the microphone **20** to reduce the detection of feedback is also possible.

In the illustrated embodiment, the amplifier **24** is configured to cancel at least some, if not all, of the feedback component of the electrical signal generated by the microphone **20**. In this embodiment, the speaker **22** and the microphone **20** are placed in close physical proximity so that the feedback detected by the microphone **20** closely resembles the sound produced by the speaker **22**. FIG. **4** illustrates an exemplary amplifier circuit **24** for canceling at least a portion of the feedback component. As one skilled in the art will appreciate, a variety of alternative analog or digital circuits can be used, each falling within the scope of the present invention.

The illustrated amplifier circuit **24** receives a electrical signal output from the microphone **22**, referred to herein a microphone signal. The microphone signal is filtered by a filter **52**. The filter **52** can be, for example, a filter to remove low frequency components of the microphone signal (e.g., frequencies below about 400 Hz). The filtered microphone signal is then amplified by a amplifier **54**, such as a low power, non-inverting operational amplifier. The gain of the amplifier **54** is selected to condition the microphone signal to a voltage level meeting the input requirements of the communication device **14**.

As indicated, the speaker **22** is coupled to the communication device **14** to receive an electrical signal for conversion into sound to be broadcast to the user of the earset **18**. This signal will be referred to herein as a speaker signal. The speaker signal is also filtered by a filter **58**. The filter **58** can be, for example, a filter to remove low frequency components of the speaker signal (e.g., frequencies below about 400 Hz). The filtered speaker signal is then amplified by an amplifier **60**. The amplifier **60** can be, for example, a low power, inverting operational amplifier. The gain of the amplifier **60** is selected to condition the speaker signal to have an amplitude equal to, or approximately equal to, the amplified feedback component of the microphone signal.

The amplified microphone signal and the amplified speaker signal are then combined so as to cancel the feedback component of the microphone signal and generate a corrected microphone signal. It is noted that the polarity or phase of the amplified microphone signal and/or the amplified speaker signal may be adjusted to correctly combine the signals. The combination of the signals can be carried out by an adder **56**. As one skilled in the art will appreciate, in an analog system, the signals can be combined by simply wiring the signals together. In one example, the conductors carrying the signals are directly connected together. Alternatively, analog or digital circuitry or processing can be used to carry out the combining function of the adder **56**.

It is noted that the speaker signal can be delayed by a delay element if needed to compensate for any propagation delay of the feedback **50**. An output of the adder **56** is coupled to a filter **62** for further conditioning of the corrected microphone signal, such as removing high frequency com-

ponents of the corrected microphone signal (e.g. frequencies above about 4 KHz for the communication system described herein).

Many alternatives to the illustrated amplifier circuit **24** are contemplated and each are considered to fall within the scope of the present invention.

As another example, the high and low frequency filters can be replaced by a band pass filter. In another example modification, the speaker signal can be amplified and/or filtered before being applied to the speaker.

In yet another example modification, the analog microphone and speaker signals can be converted to digital signals and processed using digital signal processing techniques to derive a microphone signal with no, or very little, feedback component.

Processing of the microphone signal by the communication device **14** for transmission is well known in the art and will not be described in greater detail.

With additional reference to FIG. **5**, in another embodiment of the present invention, the earset **18** can include an external microphone **70** in addition to the microphone **20** directed toward the ear canal of the user (also referred to herein as an internal microphone **20**). The external microphone **70** detects air pressure changes, or sounds, originating outside the user's ear. As described below in more detail, these sounds are converted into electrical signals and compared with electrical signals corresponding to the sounds detected by the internal microphone **20**. The respective signals can be processed to discern air pressure changes within the ear from externally produced noise, speech and the like.

The external microphone **70** is disposed on a exterior portion **72** of the housing **30** and is pointed away from the wearer of the earset **18** assembly. The external microphone **70** can be secured to the housing **30** with adhesive and can be disposed in a recess defined by the exterior portion **72**. Similar to the internal microphone **20**, the external microphone **70** is small (e.g., less than two mm in size).

The position of the external microphone **70** allows the external microphone **70** to detect air pressure changes present near the wearer's ear but having a source external to the user of the earset **18**. In one embodiment, the external microphone **70** is an omnidirectional microphone.

FIG. **5** is a block diagram that illustrates a system **10'** for detecting speech of the user. The system **10'** includes the earset **18** having the internal microphone **20**, the speaker **22** and amplifier **24***a* as described in greater detail above. In addition, the earset **18** of the system **10'** has the external microphone **70**. As one skilled in the art will appreciate, the use of the external microphone **70** is optional. The external microphone **70** is disposed on the housing **30** (FIG. **3**) and generally points away from the person wearing the earset **18**. As a result, the external microphone **70** detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset **18**, when the speech travels from the mouth of the user and through the air to the external microphone **70**.

The external microphone **70** generates an electrical signal corresponding to the sounds detected by the external microphone **70**. The external microphone **70** is electrically coupled to the communication device **14** using a separate and distinct input, or channel, than is used for the internal microphone **20**. The signal generated by the external microphone can be amplified by an amplifier **24***b* and coupled to the communication device **14** with an interface adapter **26***b*. The communication device **14** correlates the two signals produced by the microphones **20**, **70** as described below. Alternatively, the internal microphone **20** and the external microphone **70** are respectively connected to the left and right channel inputs of a processing circuit that correlates the signals as discussed below.

The unique signals generated by the internal and external microphone **20**, **70** can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system **10'**, the system **10'** can be adapted to cancel externally generated noise from the detections of the internal microphone **20**.

When the user speaks both the internal microphone **20** and the external microphone **70** detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle, machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone **70** and the internal microphone **20** detect the noise. Accordingly, the communication device **14** (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network **12** by the communication device **14**.

In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is linear (the system features are additive and homogeneous), the system can be represented by equations 1 and 2 as follows (the asterisk, or "*", represents a convolution operator):

$$F_{MI}*S+F_{MI}*N=Y_I \quad \text{Eq. 1}$$

$$F_{ME}*S+F_{ME}*N=Y_E \quad \text{Eq. 2}$$

By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N.

A method **200** for detecting speech using the system **10'** is illustrated in FIG. **6**. The method **200** includes the detection of changes in air pressure with both the internal microphone **20** and the external microphone **70**. More specifically, in step **202**, changes in air pressure within the ear **100** are detected by the internal microphone **20**. These detections can be the result of the user speaking, but can also include a noise component generated external to the person wearing the earset **18** and/or a sound component generated by speech of the user and traveling from the user's mouth, through the air and to the user's ear **100** area. Also in step **202**, changes in air pressure near, but outside, the ear **100** of the user are detected by the external microphone **70**. These detections are mainly the result of noise generated external to the user and/or a sound component generated by speech of the user and traveling from the user's mouth, through the air and to the user's ear **100** area. The detections are converted by the microphones **20** and **70** into respective electrical signals, herein referred to as an internal signal and an external signal. The internal signal can be processed to

reduce any detected feedback from the speaker **22** as discussed in more detail above.

Next, in step **204**, the speech and noise components are separated as described above. The resulting speech signal from step **204** is used in step **206** to provide the signal transmitted by the communication device **14**.

Although the invention has been shown and described with respect to a certain preferred embodiment or embodiments, it is obvious that equivalent alterations and modifications will occur to others skilled in the art upon the reading and understanding of this specification and the annexed drawings. In particular regard to the various functions performed by the above described components (assemblies, devices, circuits, etc.), the terms (including a reference to a "means") used to describe such components are intended to correspond, unless otherwise indicated, to any component which performs the specified function of the described component (i.e., that is functionally equivalent), even though not structurally equivalent to the disclosed structure which performs the function in the herein illustrated exemplary embodiments of the invention. In addition, while a particular feature of the invention may have been disclosed with respect to only one of several embodiments, such feature may be combined with one or more other features of the other embodiments as may be desired and advantageous for any given or particular application.

What is claimed is:

1. An earset, comprising:

a housing positionable with respect to an ear of a person;

a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure;

a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal input to the speaker; and

a circuit coupled to receive the microphone signal produced by the microphone and the electrical speaker signal is input to the speaker, and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker wherein the circuit includes:

an amplifier for amplifying the microphone signal produced by the microphone;

an amplifier for amplifying the electrical speaker signal received by the circuit and outputting an electrical feedback indication signal; and

a combining means for combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

2. The earset according to claim **1**, wherein the combining means is a signal adder.

3. The earset according to claim **1**, wherein the microphone is disposed for placement in a concha of the ear.

4. The earset according to claim **1**, wherein the speaker is disposed for placement in a concha of the ear.

5. The earset according to claim **1**, wherein the microphone is disposed for placement in an ear canal of the ear.

6. The earset according to claim **1**, wherein the speaker is disposed for placement in an ear canal of the ear.

7. In combination, a communications system including the earset according to claim **1**, and further comprising a communication device coupled to receive the corrected microphone signal and transmit an output corresponding to the corrected microphone signal to a remote location, and the communication device adapted to receive a signal from which the speaker signal is produced by the communication device.

8. The earset according to claim **1**, further comprising a second microphone disposed with respect to the housing and located adjacent the ear and operable to detect a change in air pressure external to the ear of the person and to produce an electrical signal corresponding to the externally detected change in air pressure.

9. The earset according to claim **8**, wherein a processing circuit executes logic to compare a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the electrical signal corresponding to the externally detected change in air pressure to distinguish detections of air pressure changes produced by the person's speech from air pressure changes produced by a source external to the person.

10. A method of detecting speech of a person, comprising the steps of:

placing a microphone at least partially within an ear of the person;

detecting a change in air pressure within the ear of the person using the microphone, the change in air pressure corresponding to the speech of the person;

producing an electrical microphone signal corresponding to the internally detected change in air pressure;

placing a speaker with respect to the ear of the person;

generating a sound with the speaker corresponding to an electrical speaker signal input to the speaker; and

generating a corrected microphone signal by reducing a feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound generated by the speaker and wherein reducing the feedback includes:

amplifying the microphone signal produced by the microphone;

amplifying the electrical speaker signal to generate an electrical feedback indication signal; and

combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

11. The method according to claim **10**, wherein the microphone is placed in a concha of the ear.

12. The method according to claim **10**, wherein the speaker is placed in a concha of the ear.

13. The method according to claim **10**, wherein the microphone is placed in an ear canal of the ear.

14. The method according to claim **10**, wherein the speaker is placed in an ear canal of the ear.

15. The method according to claim **10**, further comprising the steps of:

coupling the corrected microphone signal to a communication device;

transmitting an output corresponding to the corrected microphone signal by the communication device;

receiving a signal from which the speaker signal is produced by the communication device; and

outputting the speaker signal by the communication device.

16. The method according to claim **10**, further comprising the steps of detecting an air pressure change adjacent and external to the ear of the person with a second microphone and producing an electrical signal corresponding to the externally detected change in air pressure.

17. The method according to claim 16, further comprising the step of comparing a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the electrical signal corresponding to the externally detected change in air pressure to distinguish detections of air pressure changes produced by the speech from air pressure changes produced by a source external to the person.

* * * * *

Exhibit PA-U

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 6,671,379 B2
DATED : December 30, 2003
INVENTOR(S) : Nemirovski

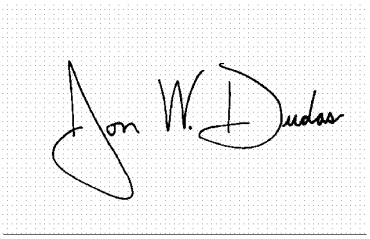
It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 1,
Line 51, replace “users” with -- user’s --

Column 11,
Line 41, replace “is” with -- as --

Signed and Sealed this

Twenty-eighth Day of September, 2004



JON W. DUDAS
Director of the United States Patent and Trademark Office

CLAIMS

What is claimed is,

1. An earpiece, comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound;

at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal; and

a processor operatively coupled to the ASM and the at least one ECR, where the processor is configured to monitor a change in an ambient sound level for detecting an acute sound from the change, and where upon detecting the acute sound, reproducing the acute sound within the ear canal via the ECR.

2. The earpiece of claim 1, wherein the processor passes sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal.

3. The earpiece of claim 1, wherein the processor maintains an approximately constant ratio between an audio content level (ACL) presented to the earpiece and an internal ambient sound level (iASL) measured within the ear canal.

4. The earpiece of claim 1, wherein the processor measures an external ambient sound level (xASL) of the ambient sound with the ASM and subtracts an attenuation level of the earpiece from the xASL to estimate the internal ambient sound level (iASL) within the ear canal.

5. The earpiece of claim 1, further comprising:

an Ear Canal Microphone (ECM) to measure an ear canal sound level (ECL) within the ear canal,

wherein the processor estimates an internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL.

6. The earpiece of claim 5, wherein the processor measures a voltage level of the audio content sent to the ECR, and applies a transfer function of the ECR to convert the voltage level to the ACL.

7. The earpiece of claim 1, wherein the processor is located external to the earpiece on a portable computing device.

8. An earpiece, comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound;

at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal;

an audio interface configured to receive audio content; and

a processor operatively coupled to the ASM, the at least one ECR, and the audio interface, where the processor is configured to monitor a change in the ambient sound level to detect an acute sound from the change, where the processor adjusts an audio content level (ACL) of an audio content delivered to the ear canal, and where upon detecting an acute sound, the acute sound is reproduced within the ear canal via the ECR.

9. The earpiece of claim 8, wherein the audio interface receives the audio content from at least one among a portable music player, a cell phone, and a portable communication device.

10. The earpiece of claim 8, wherein the processor maintains an approximately constant ratio between the audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal.

11. The earpiece of claim 10, wherein the processor mutes the audio content and passes the acute sound to the ECR for reproducing the acute sound within the ear canal.

12. The earpiece of claim 10, wherein the processor amplifies the acute sound with respect to the audio content level (ACL).

13. A method for acute sound detection and reproduction the method comprising:

measuring an ambient sound level (xASL) external to an ear at least partially occluded by an earpiece;

monitoring a change in the xASL for detecting an acute sound; and

upon detecting the acute sound, reproducing the acute sound within the ear canal.

14. The method of claim 13, wherein the step of reproducing produces sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal.

15. The method of claim 13, further comprising:

receiving audio content from an audio interface that is directed to the ear canal;

maintaining an approximately constant ratio between a level of the audio content (ACL) and a level of an internal ambient sound level (iASL) measured within the ear canal.

16. The method of claim 15, wherein the ACL is determined by

measuring a voltage level of the audio content sent to the ECR; and

applying a transfer function of the ECR to convert the voltage level to the ACL.

17. The method of claim 15, further comprising:

measuring an Ear Canal Level (ECL) within the ear canal; and

subtracting the ACL from the ECL to estimate the iASL.

18. The method of claim 15, wherein the iASL is estimated by subtracting an attenuation level of the earpiece from the xASL.

19. The method of claim 15, further comprising:

muting the audio content and passing the acute sound to an Ear Canal Receiver (ECR) for reproducing the acute sound within the ear canal.

20. The method of claim 15, further comprising:
amplifying the acute sound with respect to the audio content level (ACL).

21. A method for acute sound detection and reproduction, the method comprising:
measuring an external ambient sound level (xASL);
monitoring a change in the xASL for detecting an acute sound;
estimating a proximity of the acute sound; and
upon detecting the acute sound and it's proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece.

22. The method of claim 21, wherein the step of estimating a proximity comprises:
performing a cross correlation analysis between at least two microphones;
identifying a peak in the cross correlation and an associated time lag; and
determining the direction from the associated time lag.

23. The method of claim 21, further comprising identifying whether the acute sound is a vocal signal produced by a user operating the earpiece or a sound source external from the user.

24. A method for acute sound detection and reproduction the method comprising:
measuring an external ambient sound level (xASL) due to ambient sound outside of an ear canal at least partially occluded by the earpiece;
measuring an internal ambient sound level (iASL) due to residual ambient sound within the ear canal at least partially occluded by the earpiece;
monitoring a high frequency change between the xASL and the iASL with respect to a low frequency change between the xASL and the iASL for detecting an acute sound; and
reproducing the xASL within the ear canal responsive to detecting the high frequency change.

25. The method of claim 24, further comprising determining a proximity of a sound source producing the acute sound.

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	2750315
Application Number:	12017878
International Application Number:	
Confirmation Number:	7135
Title of Invention:	Method and Device for Acute Sound Detection and Reproduction
First Named Inventor/Applicant Name:	Steven GOLDSTEIN
Customer Number:	22191
Filer:	Richard E. Kurtz
Filer Authorized By:	
Attorney Docket Number:	PERS-0014-US
Receipt Date:	22-JAN-2008
Filing Date:	
Time Stamp:	17:00:30
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	Specification	PERS-0014-US-SPEC.pdf	293825 67e8a38d03af05fbba292ec0b020148c6cc695ee	no	21

Warnings:	
Information:	

Exhibit PAT-C

2	Claims	PERS-0014-US-CLAIMS.pdf	49811 a75755527af2e9dcf8a213e5a677bcf089fa676d	no	5
Warnings:					
Information:					
3	Abstract	PERS-0014-US-ABSTRACT.pdf	36173 9e176360c411bf4dbc73cef55a5ef3d3351db0e3	no	1
Warnings:					
Information:					
4	Drawings-only black and white line drawings	PERS-0014-US-FIGS.pdf	467273 0890d90c50740c9f9250343ec0088fd93a68e69c	no	9
Warnings:					
Information:					
5	Application Data Sheet	PERS-0014-US_ADS_Form.pdf	1103352 37d487ea677fd84da1610ab359ae90785e0aeea7	no	5
Warnings:					
Information:					
		Total Files Size (in bytes):	1950434		

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.

Exhibit PAT-C

UNITED STATES PATENT AND TRADEMARK OFFICE

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

APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
12/017,878	01/22/2008	Steven Wayne Goldstein	PERS-0014-US

CONFIRMATION NO. 7135

22191
GREENBERG TRAURIG, LLP
1750 TYSONS BOULEVARD, 12TH FLOOR
MCLEAN, VA 22102

FORMALITIES LETTER

OC000000028147854

Date Mailed: 02/08/2008

NOTICE TO FILE MISSING PARTS OF NONPROVISIONAL APPLICATION

FILED UNDER 37 CFR 1.53(b)

Filing Date Granted

Items Required To Avoid Abandonment:

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

- The statutory basic filing fee is missing.
  *Applicant must submit $**75** to complete the basic filing fee for a small entity.*
- The oath or declaration is missing.
  A properly signed oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.
  Note: If a petition under 37 CFR 1.47 is being filed, an oath or declaration in compliance with 37 CFR 1.63 signed by all available joint inventors, or if no inventor is available by a party with sufficient proprietary interest, is required.

The application is informal since it does not comply with the regulations for the reason(s) indicated below.

The required item(s) identified below must be timely submitted to avoid abandonment:

- Replacement drawings in compliance with 37 CFR 1.84 and 37 CFR 1.121(d) are required. The drawings submitted are not acceptable because:
  - The drawings submitted to the Office are not electronically reproducible because portions of figures 1, 4-10 are missing and/or blurry.

Applicant is cautioned that correction of the above items may cause the specification and drawings page count to exceed 100 pages. If the specification and drawings exceed 100 pages, applicant will need to submit the required application size fee.

The applicant needs to satisfy supplemental fees problems indicated below.

The required item(s) identified below must be timely submitted to avoid abandonment:

- Additional claim fees of $**335** as a small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due.
- To avoid abandonment, a surcharge (for late submission of filing fee, search fee, examination fee or oath or declaration) as set forth in 37 CFR 1.16(f) of $**65** for a small entity in compliance with 37 CFR 1.27, must be submitted with the missing items identified in this notice.

SUMMARY OF FEES DUE:

Total additional fee(s) required for this application is $**835** for a small entity
- $**75** Statutory basic filing fee.
- $**65** Surcharge.
- The application search fee has not been paid. Applicant must submit $**255** to complete the search fee.
- The application examination fee has not been paid. Applicant must submit $**105** to complete the examination fee for a small entity in compliance with 37 CFR 1.27.
- Total additional claim fee(s) for this application is $**335**
  - $**210** for **2** independent claims over 3.
  - $**125** for **5** total claims over 20.

Replies should be mailed to:

Mail Stop Missing Parts
Commissioner for Patents
P.O. Box 1450
Alexandria VA 22313-1450

Registered users of EFS-Web may alternatively submit their reply to this notice via EFS-Web.
https://sportal.uspto.gov/authenticate/AuthenticateUserLocalEPF.html

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

If you are not using EFS-Web to submit your reply, you must include a copy of this notice.

/tnguyen/

Office of Initial Patent Examination (571) 272-4000 or 1-800-PTO-9199

Exhibit PAT-C

UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
12/017,878	01/22/2008	2615	0.00	PERS-0014-US	25	5

CONFIRMATION NO. 7135

22191
GREENBERG TRAURIG, LLP
1750 TYSONS BOULEVARD, 12TH FLOOR
MCLEAN, VA 22102

FILING RECEIPT

OC000000028147853

Date Mailed: 02/08/2008

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

Applicant(s)

Steven Wayne Goldstein, Delray Beach, FL;
Marc Andre Boillot, Plantation, FL;
John Usher, Montreal, CANADA;

Assignment For Published Patent Application

PERSONICS HOLDINGS INC., Boca Raton, FL

Power of Attorney: None

Domestic Priority data as claimed by applicant

This appln claims benefit of 60/885,917 01/22/2007

Foreign Applications

If Required, Foreign Filing License Granted: 02/06/2008

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 12/017,878**

Projected Publication Date: To Be Determined - pending completion of Missing Parts

Non-Publication Request: No

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

Exhibit PAT-C

Title

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

Preliminary Class

381

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

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

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

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

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

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

LICENSE FOR FOREIGN FILING UNDER

Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

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The applicant has been granted a license under 35 U.S.C. 184, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" followed by a date appears on this form. Such licenses are issued in all applications where the conditions for issuance of a license have been met, regardless of whether or not a license may be required as

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This license is to be retained by the licensee and may be used at any time on or after the effective date thereof unless it is revoked. This license is automatically transferred to any related applications(s) filed under 37 CFR 1.53(d). This license is not retroactive.

The grant of a license does not in any way lessen the responsibility of a licensee for the security of the subject matter as imposed by any Government contract or the provisions of existing laws relating to espionage and the national security or the export of technical data. Licensees should apprise themselves of current regulations especially with respect to certain countries, of other agencies, particularly the Office of Defense Trade Controls, Department of State (with respect to Arms, Munitions and Implements of War (22 CFR 121-128)); the Bureau of Industry and Security, Department of Commerce (15 CFR parts 730-774); the Office of Foreign AssetsControl, Department of Treasury (31 CFR Parts 500+) and the Department of Energy.

NOT GRANTED

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Exhibit PAT-C

PATENT
Docket No.: **092624.011700 (PERS-0014-US)**

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of: **Steven W. Goldstein et al.**	Confirmation No. **7135**
Serial No.: **12/017,878**	Art Unit: **2615**
Filing Date: **January 22, 2008**	Examiner: **Unknown**
Title: **METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION**	

RESPONSE TO NOTICE TO FILE MISSING PARTS

MAIL STOP MISSING PARTS
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

In response to the Notice to File Missing Parts mailed February 8, 2008, Applicant encloses:

1. Signed Declaration and Power of Attorney;
2. Replacement Drawings; and
3. Please charge the following fees to Deposit Account No. 50-4284:

Basic Filing Fee for Small Entity:	$ 75.00
Surcharge:	65.00
Additional Claims Fees:	
• 2 independent claim over 3	210.00
• 5 Claims in excess of 20	125.00
Search Fee:	255.00
Examination Fee:	105.00
Total:	**$835.00**

The Commissioner is further authorized to charge any deficiencies or credit any overpayment to Deposit Account No. 50-4284.

Respectfully submitted,

/Richard E. Kurtz/Reg. #33936

Filed: April 8, 2008

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Customer No. 22191

REK:cmu

Exhibit PAT-C

UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of: **Steven W. Goldstein et al.**	Docket No.: **092624.011700 (PERS-0014-US)**
Serial No.: **12/017,878**	Art Unit: **Unknown**
Filing Date: **January 22, 2008**	Examiner: **Unassigned**
Title: **METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION**	

DECLARATION FOR PATENT APPLICATION AND APPOINTMENT OF ATTORNEY

DECLARATION OF SOLE OR JOINT INVENTOR

As a below-named inventor, I hereby declare that my residence, post office address and citizenship are as stated below next to my name; I believe that I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention (Design, if applicable) entitled above, the specification of which is either attached hereto, or was filed on the date listed above as with the Application Serial Number listed above (whichever is applicable).

REVIEW AND UNDERSTOOD CLAUSE

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment(s). I acknowledge the duty to disclose information which is material to the patentability of this application in accordance with *Title 37, Code of Federal Regulations, §1.56(a).*

DOMESTIC PRIORITY CLAIM

I hereby claim the benefit under *Title 35, United States Code, § 120* of any United States application(s) or PCT International Application(s) designating The United States of America listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in that/those prior application(s) in the manner provided by the first paragraph of *Title 35, United States Code § 112*, I acknowledge the duty to disclose information which is material to patentability as defined in *Title 37, Code of Federal Regulations, § 1.56(a)* which occurred between the filing date of the prior application(s) and the national or PCT international filing date of this application:

APPLICATION NUMBER	FILING DATE	STATUS (Issued, Pending, etc.)
60/885,917	**January 22, 2007**	**Expired**

DECLARATION FOR PATENT APPLICATION AND APPOINTMENT OF ATTORNEY

FOREIGN PRIORITY CLAIM

I hereby claim foreign priority benefits under *Title 35, United States Code § 119* of any foreign application(s) for patent or inventor's certificate or of any PCT International Application(s) designating at least one country other than the United States of America listed below and have also identified below any foreign application(s) for patent or inventor's certificate or of any PCT International application(s) designating at least one country other than the United States of America filed by me on the same subject matter having a filing date before that of the application(s) on which the priority is claimed.

APPLICATION NUMBER	FILING DATE	COUNTRY

WILLFUL FALSE STATEMENT CLAUSE

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

CORRESPONDENCE ADDRESS

Please send all correspondence regarding the above-captioned application to:

CUSTOMER NUMBER: 22191

Direct all phone inquiries to: 407-418-2356

POWER OF ATTORNEY

I (We) hereby appoint as my (our) attorneys, with full powers of substitution and revocation, to prosecute this application and transact all business in the Patent and Trademark Office connected therewith, the Patent Practitioners associated with Customer Number 22191.

FIRST INVENTOR INFORMATION	
Full Name	*Citizenship*
Steven Wayne Goldstein	**U.S.**
Mailing Address	*City:*
16390 Maddalena Place	**Delray Beach**
State	*Zip Code*
Florida	**33446**
Country	
United States of America	
Date:	*Signature:*
3 April 2008	

SECOND INVENTOR INFORMATION	
Full Name	*Citizenship*
Marc Andre Boillot	**U.S.**
Mailing Address	*City:*
9110 NW 11th Court	**Plantation**
State	*Zip Code*
Florida	**33322**
Country	
United States of America	
Date:	*Signature:*
3 April 2008	

THIRD INVENTOR INFORMATION	
Full Name	*Citizenship*
John Usher	**Canada**
Mailing Address	*City*
55 de Bresoles, Apt. 400	**Montreal**
State	*Zip Code*
	H2Y1V7
Country	
Canada	
Date:	*Signature:*
4/4/08	J. Usher

Exhibit PAT-C

REPLACEMENT SHEET
1/9

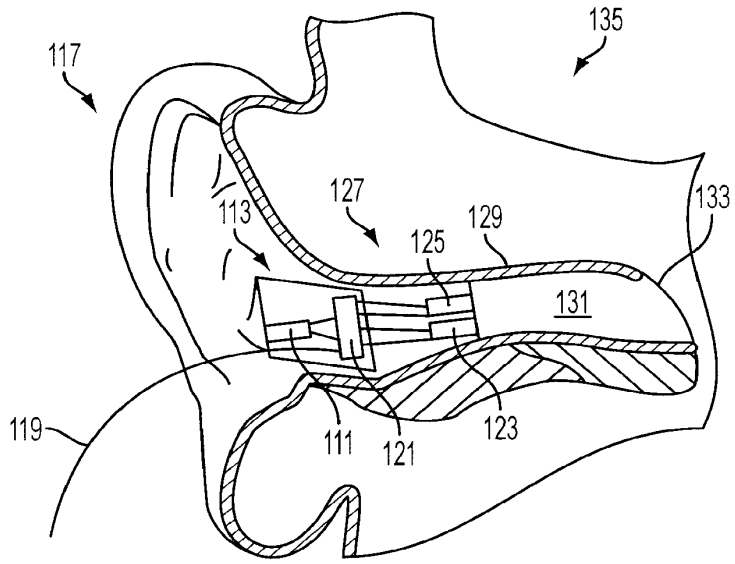


FIG. 1

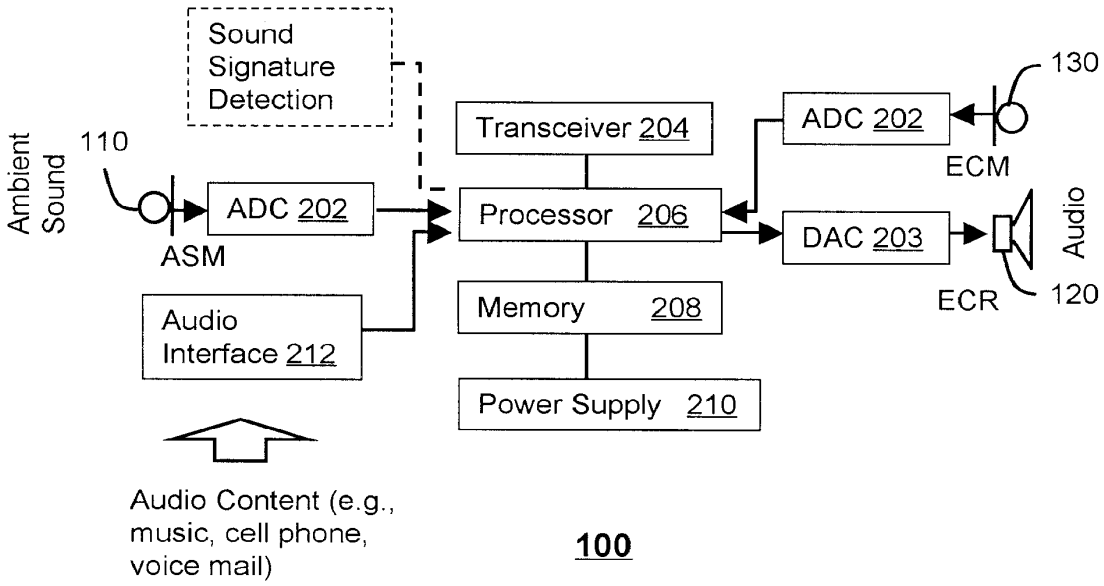


FIG. 2

Exhibit PAT-C

REPLACEMENT SHEET
3/9

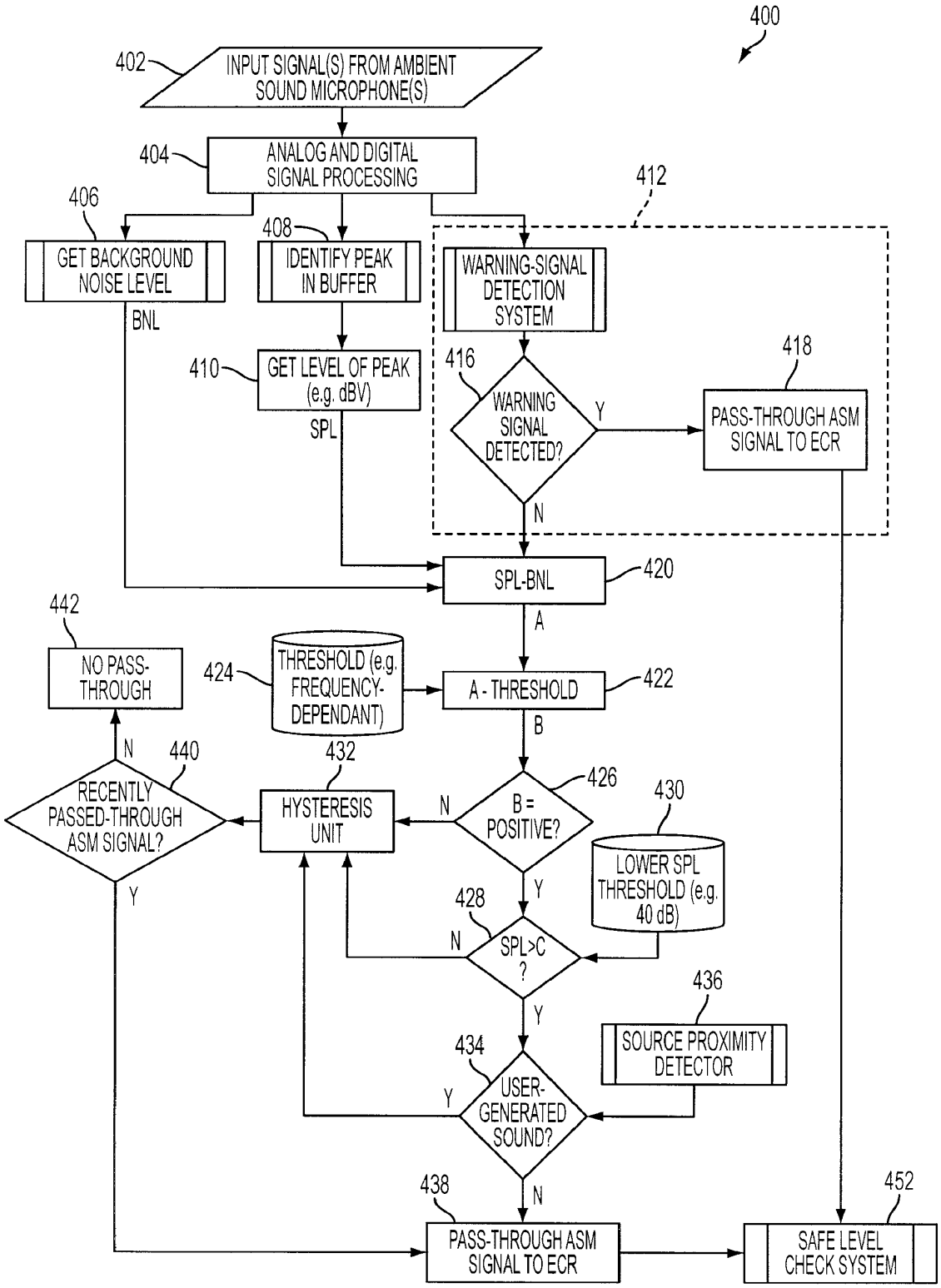


FIG. 4

Exhibit PAT-C

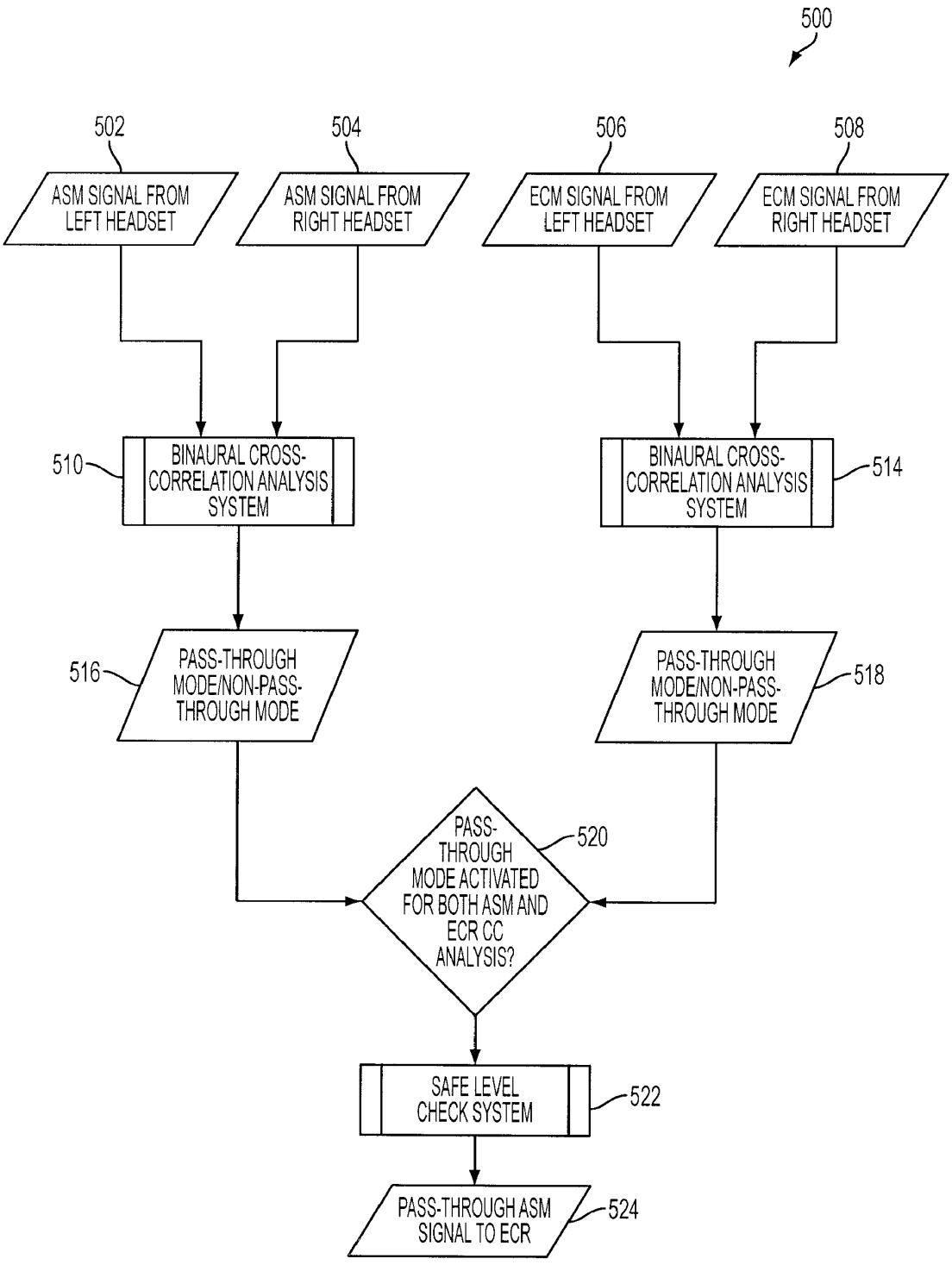


FIG. 5

Exhibit PAT-C

REPLACEMENT SHEET
5/9

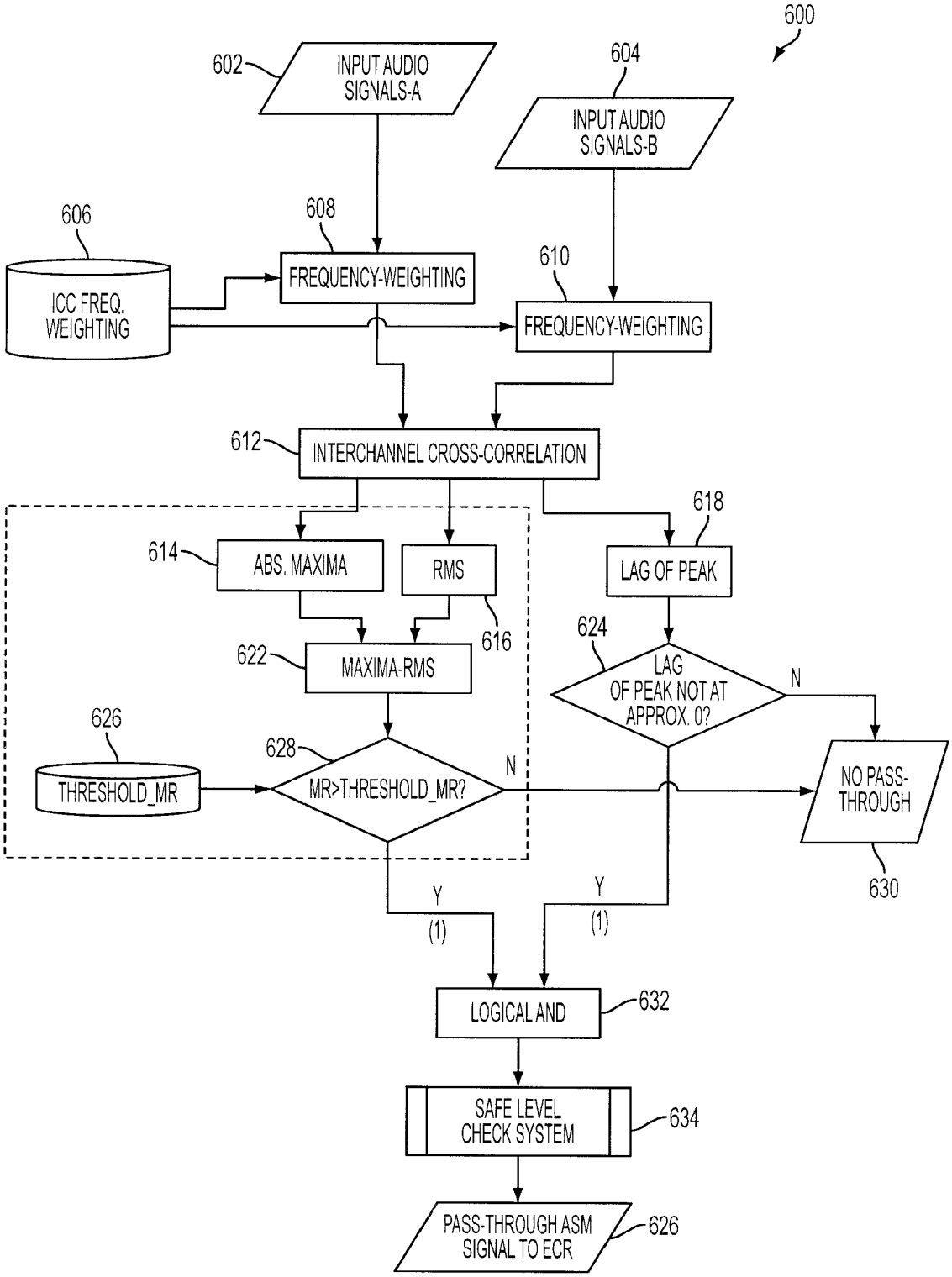


FIG. 6

Exhibit PAT-C

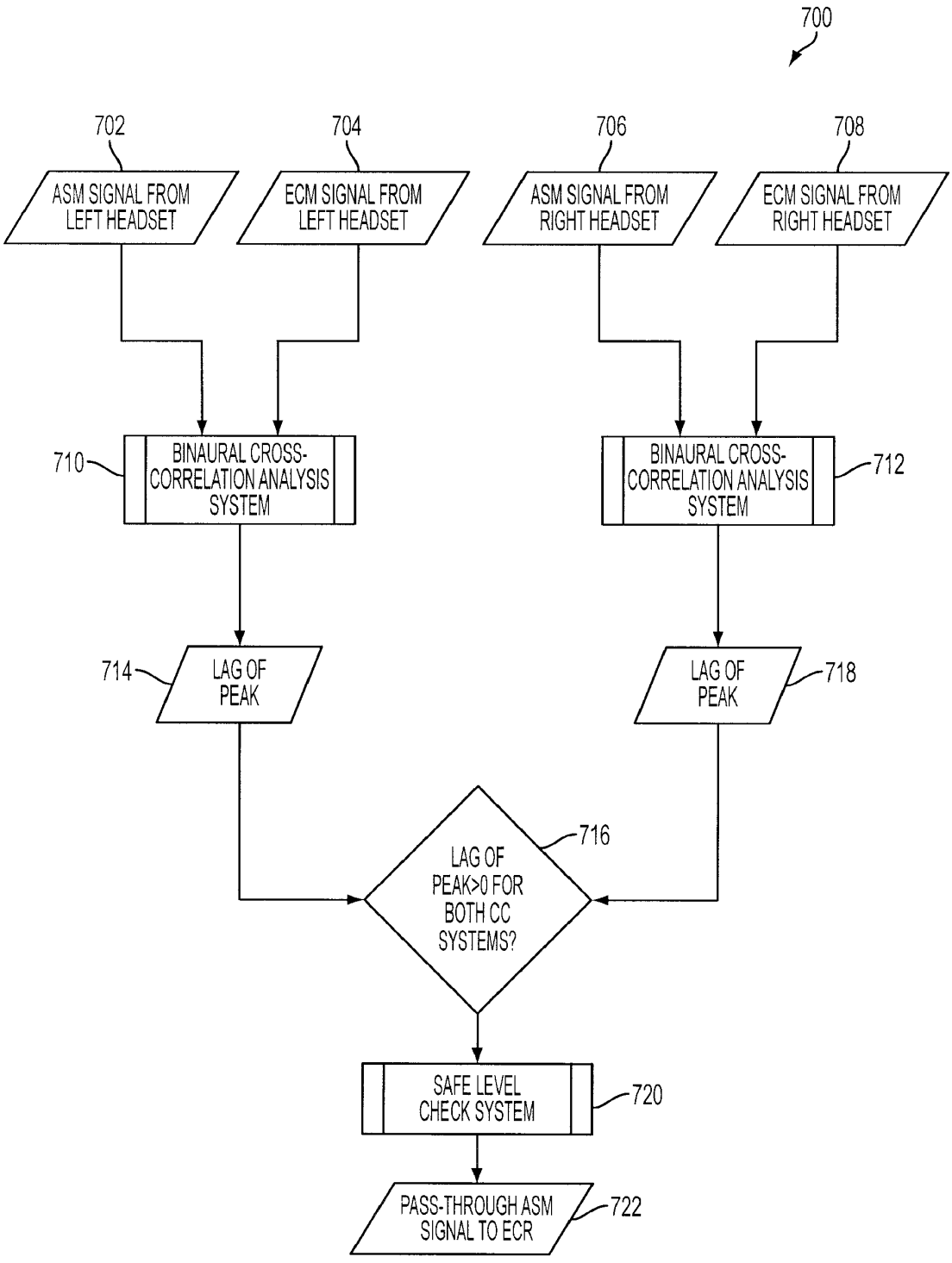


FIG. 7

Exhibit PAT-C

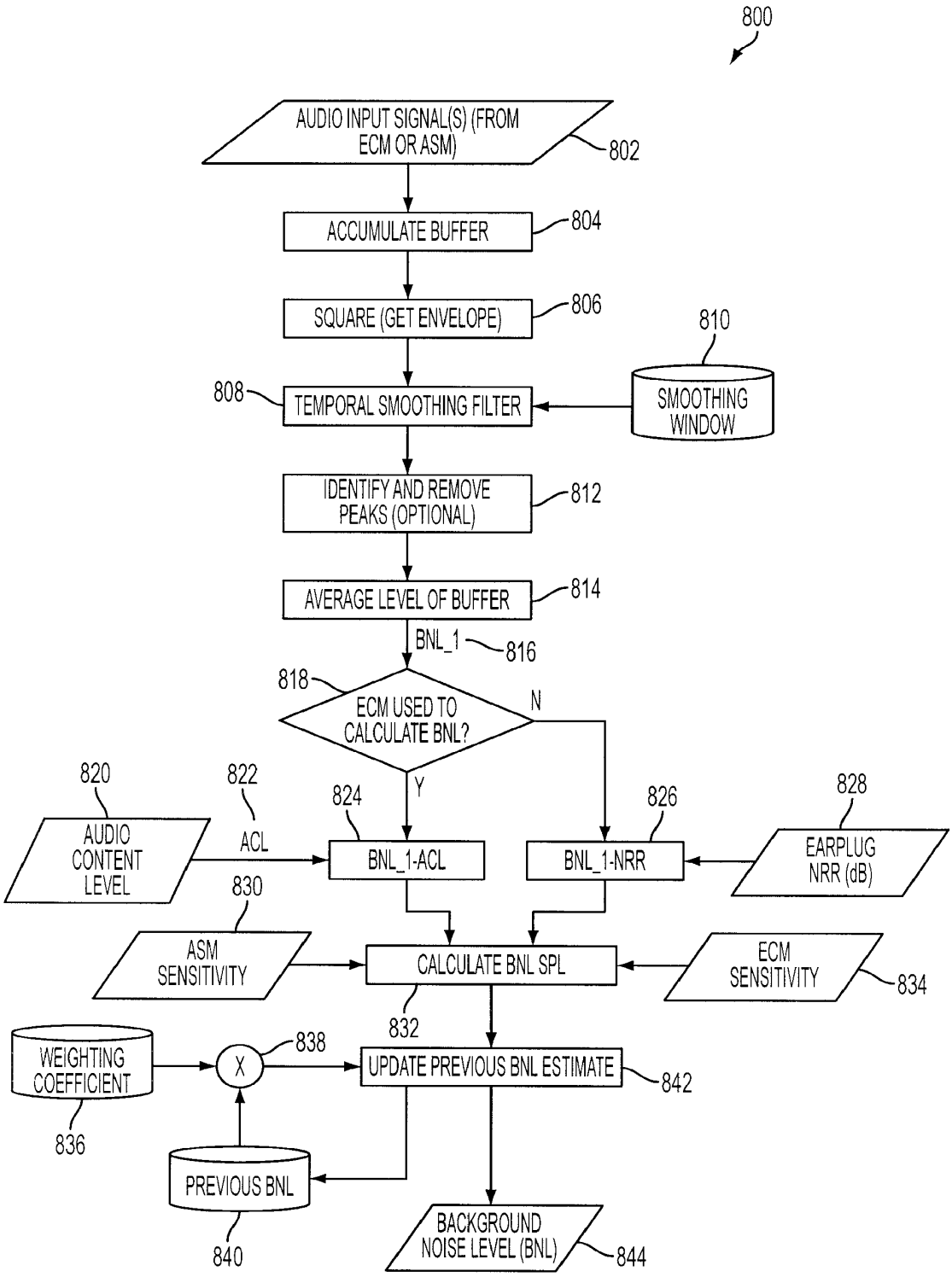


FIG. 8

Exhibit PAT-C

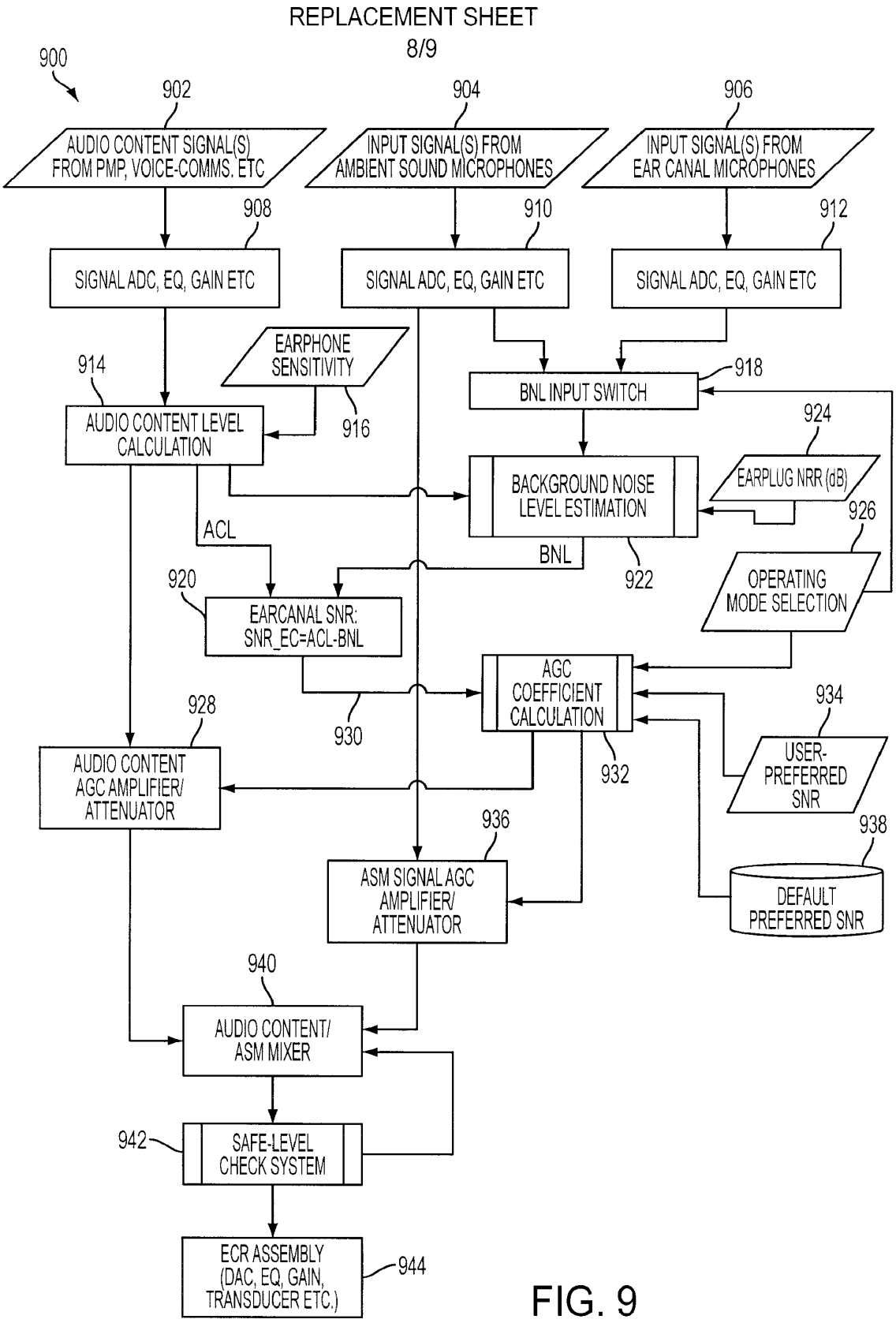


FIG. 9

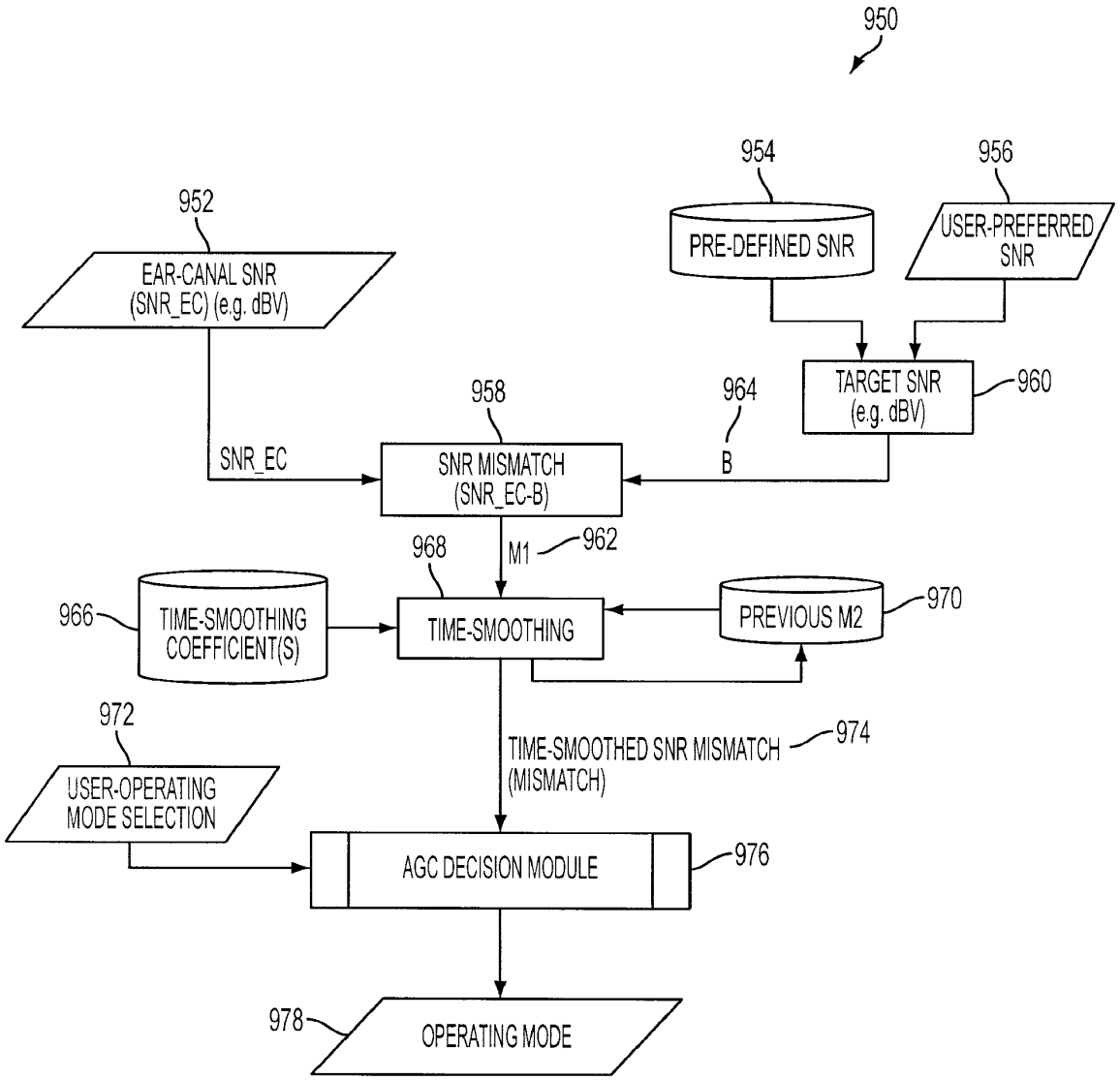

950
954
956
952
PRE-DEFINED SNR
USER-PREFERRED SNR
EAR-CANAL SNR (SNR_EC) (e.g. dBV)
TARGET SNR (e.g. dBV)
960
958
964
SNR_EC
SNR MISMATCH (SNR_EC-B)
B
968
M1
962
966
TIME-SMOOTHING COEFFICIENT(S)
TIME-SMOOTHING
PREVIOUS M2
970
972
TIME-SMOOTHED SNR MISMATCH (MISMATCH)
974
USER-OPERATING MODE SELECTION
AGC DECISION MODULE
976
978
OPERATING MODE

FIG. 10

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	3118405
Application Number:	12017878
International Application Number:	
Confirmation Number:	7135
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	22191
Filer:	Richard E. Kurtz/Catherine Updegraff
Filer Authorized By:	Richard E. Kurtz
Attorney Docket Number:	092624.011700 (PERS-014)
Receipt Date:	08-APR-2008
Filing Date:	22-JAN-2008
Time Stamp:	12:34:49
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Deposit Account
Payment was successfully received in RAM	$835
RAM confirmation Number	6980
Deposit Account	504284
Authorized User	
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows: Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees) Charge any Additional Fees required under 37 C.F.R. Section 1.21 (Miscellaneous fees and charges)	

Exhibit PAT-C

File Listing:

Document Number	Document Description	File Name	File Size(Bytes) /Message Digest	Multi Part /.zip	Pages (if appl.)
1	Applicant Response to Pre-Exam Formalities Notice	Missing_Parts_Response.pdf	97918 469a7beb4e987577385838bd33dd53b624fd4b26	no	1
Warnings:					
Information:					
2		Executed_Declaration.pdf	176419 65e7912bc8184f0867cd0d4ce424cb231e171d3a	yes	4

Multipart Description/PDF files in .zip description

Document Description	Start	End
Oath or Declaration filed	1	2
Power of Attorney	3	4

Warnings:					
Information:					
3	Drawings-only black and white line drawings	Replacement_Drawings.pdf	363086 6ecb83936356113ac74d148eaa35b59c570a03c0	no	8
Warnings:					
Information:					
4	Fee Worksheet (PTO-06)	fee-info.pdf	8808 cd07c18edae2e1b3213cdd83520f21a1063a9ebc	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	646231		

Exhibit PAT-C

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.

Exhibit PAT-C

UNITED STATES PATENT AND TRADEMARK OFFICE

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

APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
12/017,878	01/22/2008	2615	835	092624.011700 (PERS-0014-	25	5

CONFIRMATION NO. 7135

22191
GREENBERG TRAURIG, LLP
1750 TYSONS BOULEVARD, 12TH FLOOR
MCLEAN, VA 22102

UPDATED FILING RECEIPT



Date Mailed: 04/21/2008

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

Applicant(s)

Steven Wayne Goldstein, Delray Beach, FL;
Marc Andre Boillot, Plantation, FL;
John Usher, Montreal, CANADA;

Assignment For Published Patent Application

PERSONICS HOLDINGS INC., Boca Raton, FL

Power of Attorney: The patent practitioners associated with Customer Number 22191

Domestic Priority data as claimed by applicant

This appln claims benefit of 60/885,917 01/22/2007

Foreign Applications

If Required, Foreign Filing License Granted: 02/06/2008

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 12/017,878**

Projected Publication Date: 07/31/2008

Non-Publication Request: No

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

Exhibit PAT-C

Title

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

Preliminary Class

381

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

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

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

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

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

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

LICENSE FOR FOREIGN FILING UNDER

Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

GRANTED

The applicant has been granted a license under 35 U.S.C. 184, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" followed by a date appears on this form. Such licenses are issued in all applications where the conditions for issuance of a license have been met, regardless of whether or not a license may be required as

set forth in 37 CFR 5.15. The scope and limitations of this license are set forth in 37 CFR 5.15(a) unless an earlier license has been issued under 37 CFR 5.15(b). The license is subject to revocation upon written notification. The date indicated is the effective date of the license, unless an earlier license of similar scope has been granted under 37 CFR 5.13 or 5.14.

This license is to be retained by the licensee and may be used at any time on or after the effective date thereof unless it is revoked. This license is automatically transferred to any related applications(s) filed under 37 CFR 1.53(d). This license is not retroactive.

The grant of a license does not in any way lessen the responsibility of a licensee for the security of the subject matter as imposed by any Government contract or the provisions of existing laws relating to espionage and the national security or the export of technical data. Licensees should apprise themselves of current regulations especially with respect to certain countries, of other agencies, particularly the Office of Defense Trade Controls, Department of State (with respect to Arms, Munitions and Implements of War (22 CFR 121-128)); the Bureau of Industry and Security, Department of Commerce (15 CFR parts 730-774); the Office of Foreign AssetsControl, Department of Treasury (31 CFR Parts 500+) and the Department of Energy.

NOT GRANTED

No license under 35 U.S.C. 184 has been granted at this time, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" DOES NOT appear on this form. Applicant may still petition for a license under 37 CFR 5.12, if a license is desired before the expiration of 6 months from the filing date of the application. If 6 months has lapsed from the filing date of this application and the licensee has not received any indication of a secrecy order under 35 U.S.C. 181, the licensee may foreign file the application pursuant to 37 CFR 5.15(b).

Exhibit PAT-C

Form PTO-1449	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTY. DKT. NO. **092624.011700**	SERIAL NO. **12/017,878**
INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Use several sheets if necessary)		APPLICANTS **Steven W. Goldstein et al.**	EXAMINER: **Unassigned**
		FILING DATE **January 22, 2008**	GROUP **2615**

U.S. PATENT DOCUMENTS

Examiner's Initial		Document No.	Publication Date	Name	Class	Subclass	Filing Date
	AA	2006/0045282	03/02/2006	Reber			
	AB	2001/0046304	11/29/2001	Rast			
	AC						
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						
	AL						
	AM						
	AN						
	AO						
	AP						
	AQ						
	AR						

FOREIGN PATENT DOCUMENTS

Examiner's Initial		Document Number	Date	Country	Class	Subclass	Translation	
							Yes	**No**
	AS							
	AT							
	AU							
	AV							

OTHER DOCUMENTS (Including Author, Title, Date, Pertinent Pages, Etc.)

	AW	
	AX	
	AY	
	AZ	

Examiner	**Date Considered**

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	3825753
Application Number:	12017878
International Application Number:	
Confirmation Number:	7135
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	22191
Filer:	Richard E. Kurtz/Catherine Updegraff
Filer Authorized By:	Richard E. Kurtz
Attorney Docket Number:	092624.011700 (PERS-14)
Receipt Date:	22-AUG-2008
Filing Date:	22-JAN-2008
Time Stamp:	15:21:29
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	Information Disclosure Statement Letter	IDS_2_.pdf	87583 54e4dfea98c658989448809a6348f514e7f369bf	no	1

Warnings:	
Information:	

Exhibit PAT-C

2	Information Disclosure Statement (IDS) Filed (SB/08)	PTO_1449.pdf	110994 f218e66d82a2a20002fd82e342b33fad870289b3	no	1

Warnings:	
Information:	
This is not an USPTO supplied IDS fillable form	
Total Files Size (in bytes):	198577

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.

Exhibit PAT-C

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.
12/017,878	01/22/2008	Steven Wayne Goldstein	PRS-118US (PERS-0014US)	7135

71158 7590 10/05/2011

RatnerPrestia
P.O. Box 980
Valley Forge, PA 19482

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	

MAIL DATE	DELIVERY MODE
10/05/2011	PAPER

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.

PTOL-90A (Rev. 04/07)

Exhibit PAT-C

Office Action Summary	Application No.	Applicant(s)	
	12/017,878	GOLDSTEIN ET AL.	
	Examiner	Art Unit	
	CUONG NGUYEN	2818	

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

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 1 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *22 January 2008*.

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

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

4)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

5)☒ Claim(s) *1-25* is/are pending in the application.

5a) Of the above claim(s) _____ is/are withdrawn from consideration.

6)☐ Claim(s) _____ is/are allowed.

7)☐ Claim(s) _____ is/are rejected.

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

9)☒ Claim(s) *1-25* are subject to restriction and/or election requirement.

Application Papers

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

11)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

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

Priority under 35 U.S.C. § 119

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

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

Exhibit PAT-C

Election/Restrictions

1. This application contains claims directed to the following patentably distinct species:

I. First embodiment: a method for acute sound detection and reproduction including maintaining an approximately constant ratio

II. Second embodiment: a method for acute sound detection and reproduction including estimating proximity

III. Third embodiment: a method for acute sound detection and reproduction including monitoring the change in frequency

2. The species are independent or distinct because species I describes the method including a maintenance of an approximately constant ratio between a level of the audio content and a level of an internal ambient sound level measured within the ear; while species II describes the method including an estimation of a proximity of the acute sound wherein performing a cross correlation analysis between two microphones and while species III describes the method including monitoring a high frequency change with respect to a low frequency change between external ambient sound level and internal ambient sound level.

Applicant is required under 35 U.S.C. 121 to elect a single disclosed species, or a single grouping of patentably indistinct species, for prosecution on the merits to which the claims shall be restricted if no generic claim is finally held to be allowable. Currently, no claims are generic.

There is a search and/or examination burden for the patentably distinct species as set forth above because at least the following reason(s) apply:

The species or grouping of patentably indistinct species requires a different field of search (e.g. searching in different classes/subclasses or requiring different search strategies or search queries).

Applicant is advised that the reply to this requirement to be complete must include (i) an election of a species or a grouping of patentably indistinct species to be examined even though the requirement may be traversed (37 CFR 1.143) **and (ii) identification of the claims encompassing the elected species or grouping of patentably indistinct species**, including any claims subsequently added. An argument that a claim is allowable or that all claims are generic is considered nonresponsive unless accompanied by an election.

The election may be made with or without traverse. To preserve a right to petition, the election must be made with traverse. If the reply does not distinctly and specifically point out supposed errors in the election of species requirement, the election shall be treated as an election without traverse. Traversal must be presented at the time of election in order to be considered timely. Failure to timely traverse the requirement will result in the loss of right to petition under 37 CFR 1.144. If claims are added after the election, applicant must indicate which of these claims are readable on the elected species or grouping of patentably indistinct species.

Should applicant traverse on the ground that the species, or groupings of patentably indistinct species from which election is required, are not patentably distinct,

applicant should submit evidence or identify such evidence now of record showing them to be obvious variants or clearly admit on the record that this is the case. In either instance, if the examiner finds one of the species unpatentable over the prior art, the evidence or admission may be used in a rejection under 35 U.S.C. 103(a) of the other species.

Upon the allowance of a generic claim, applicant will be entitled to consideration of claims to additional species which depend from or otherwise require all the limitations of an allowable generic claim as provided by 37 CFR 1.141.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to CUONG NGUYEN whose telephone number is (571)270-1509. The examiner can normally be reached on Monday to Friday 8:30am-5:00pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Steven Loke can be reached on 571-272-1657. 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.

CN

/STEVEN LOKE/
Supervisory Patent Examiner, Art Unit 2818

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Appln. No:	12/017,878
Applicant:	Steven Wayne Goldstein et al.
Filed:	January 22, 2008
Title:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
TC/A.U.:	2818
Examiner:	Cuong B. Nguyen
Confirmation No.:	7135
Docket No.:	PRS-118US

RESPONSE TO RESTRICTION REQUIREMENT

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

S I R :

This is in response to the Restriction Requirement stated in the Office Letter dated **October 5, 2011**.

The Examiner asserts that this application contains claims directed to the following patentably distinct species and requires that one of the following be elected for prosecution:

I. First embodiment: a method for acute sound detection and reproduction including maintaining an approximately constant ratio.

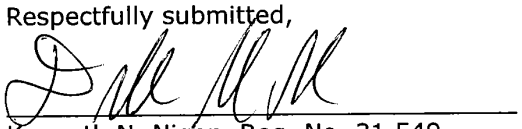
II. Second embodiment: a method for acute sound detection and reproduction including estimating proximity.

III. Third embodiment: a method for acute sound detection and reproduction including monitoring the change in frequency.

Applicants assert that Species I corresponds to claims 3, 10-12, 15-20; Species II corresponds to claims 21-23; and Species III corresponds to claims 24 and 25. Applicants also note that claims 1, 2, 4-9, 13 and 14 are generic with respect to all of the species. Applicants elect to prosecute **Species I (claims 1-20)**. This election is made without traverse.

On page 2 of the Office Action, the Examiner indicates that no claims are generic. Applicants respectfully disagree because claims 1, 2, 4-9, 13 and 14 are not limited to any of the species and, thus, are generic to all of the species.

Respectfully submitted,



Kenneth N. Nigon, Reg. No. 31,549
Deborah M. Grove, Reg. No. 61,091
Attorney/Patent Agent for Applicants

KNN/DMG/kpc

Dated: November 3, 2011

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Valley Forge, PA 19482
(610) 407-0700

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	11326822
Application Number:	12017878
International Application Number:	
Confirmation Number:	7135
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	71158
Filer:	Deborah M. Grove/Kathleen Carney
Filer Authorized By:	Deborah M. Grove
Attorney Docket Number:	PRS-118US (PERS-0014US)
Receipt Date:	03-NOV-2011
Filing Date:	22-JAN-2008
Time Stamp:	13:11:03
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	Response to Election / Restriction Filed	ResponseRestrictionRequirement.pdf	46319 54852a008523d89ca939ffbb6179321d465c8116	no	1

Warnings:

Information:

Exhibit PAT-C

Total Files Size (in bytes):	46319

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.

Exhibit PAT-C

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
12/017,878	01/22/2008	Steven Wayne Goldstein	PRS-118US (PERS-0014US)	7135

71158 7590 01/18/2012

RatnerPrestia
P.O. Box 980
Valley Forge, PA 19482

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	

MAIL DATE	DELIVERY MODE
01/18/2012	PAPER

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.

PTOL-90A (Rev. 04/07)

Exhibit PAT-C

Office Action Summary	Application No.	Applicant(s)	
	12/017,878	GOLDSTEIN ET AL.	
	Examiner	Art Unit	
	CUONG NGUYEN	2818	

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

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *11/03/2011*.

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

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

4)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

5)☒ Claim(s) *1-20* is/are pending in the application.

5a) Of the above claim(s) *21-25* is/are withdrawn from consideration.

6)☐ Claim(s) _____ is/are allowed.

7)☒ Claim(s) *1-20* is/are rejected.

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

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

Application Papers

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

11)☒ The drawing(s) filed on *08 April 2008* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

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

Priority under 35 U.S.C. § 119

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

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *08/22/2008*.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Election/Restrictions

1. Applicant's election without traverse of Species I (Claims 1-20) in the reply filed on October 5, 2011 is acknowledged.

Regarding to Applicant's arguments about no claims are generic. The Examiner agrees with the Applicants that claims 1, 2, 4-9, 13 and 14 are not limited to any of the species and therefore are generic to all of species.

Claim Objections

2. Claim 1 is objected to because of the following informalities: the limitation "a processor operatively coupled to ASM and **the at least** one ECR" in line 5 is grammatically wrong. Appropriate correction is required.

3. Claim 8 is objected to because of the following informalities: the limitation "a processor operatively coupled to ASM, **the at least** one ECR" in line 5 is grammatically wrong. Appropriate correction is required.

Double Patenting

4. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. A nonstatutory

obviousness-type double patenting rejection is appropriate where the conflicting claims are not identical, but at least one examined application claim is not patentably distinct from the reference claim(s) because the examined application claim is either anticipated by, or would have been obvious over, the reference claim(s). See, e.g., *In re Berg*, 140 F.3d 1428, 46 USPQ2d 1226 (Fed. Cir. 1998); *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970); and *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) or 1.321(d) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent either is shown to be commonly owned with this application, or claims an invention made as a result of activities undertaken within the scope of a joint research agreement.

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

5. Claims 1, 8, and 13 are provisionally rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claims 1, 6, 7, and 8 of copending Application No. 11/966457. Although the conflicting claims are not identical, they are not patentably distinct from each other because even though the language of claims in both application are different, the Co-pending Application includes all the

Exhibit PAT-C

limitations correspond to the limitations of the instant application as shown in the table below.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

Instant Application:	Co-pending Application (11/966457):
Claim 1. An earpiece, comprising: an **Ambient Sound Microphone (ASM**) configured to capture ambient sound; at least one **Ear Canal Receiver (ECR)** configured to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the at least one ECR, where the **processor** is configured to **monitor a change in an ambient sound level for detecting an acute sound** from the change, and where upon detecting the acute sound, **reproducing the acute sound within the ear canal** via the ECR	**Claim 1.** An acoustic device, comprising: an **Ambient Sound Microphone (ASM)** configured to capture ambient sound; at least one **Ear Canal Receiver (ECR)** configured to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the ECR, where the **processor monitors the ambient sound for a target sound** and **adjusts the audio delivered to the ear canal** based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound
Claim 8.	**Claim 1.**

Exhibit PAT-C

An earpiece, comprising: an **Ambient Sound Microphone (ASM)** configured to capture ambient sound; at least one E**ar Canal Receiver (ECR)** configured to deliver audio to an ear canal; an audio interface configured to receive audio content; and a **processor** operatively coupled to the ASM, the at least one ECR, and the audio interface, where the processor is configured to **monitor a change in the ambient sound** level to **detect an acute sound** from the change, where the processor **adjusts an audio content level (ACL) of an audio content delivered to the ear canal,** and where upon detecting an acute sound, the **acute sound is reproduced** within the ear canal via the ECR.	An acoustic device, comprising: an **Ambient Sound Microphone (ASM)** configured to capture ambient sound; at least one **Ear Canal Receiver (ECR)** configured to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the ECR, where the **processor** monitors **the ambient sound for a target sound** and **adjusts the audio** delivered to the ear canal based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound **Claim 6.** The acoustic device of claim 1, where the acoustic device is earpiece, wherein the processor detects sound signatures in the ambient sound and adjusts the audio delivered to the ear canal based on detected sound signatures.The acoustic

	device of claim 1, where the acoustic device is an earpiece, further comprising an audio interface operatively coupled to the processor configured to receive audio content from a media player or cell phone, wherein the processor selectively **adjusts a volume of the audio content delivered to the ear canal** when the target sound is detected.
Claim 13. A method for acute sound detection and reproduction the method comprising: **measuring an ambient sound level** (xASL) external to an ear at least partially occluded by an earpiece; **monitoring a change** in the xASL for detecting an acute sound; and upon detecting the acute sound, **reproducing the acute sound within the ear canal.**	**Claim 7.** A method for personalized listening, the method comprising: **capturing ambient sound** with an Ambient Sound Microphone (ASM); **monitoring the ambient sound for a target sound** by generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound; adjusting a delivery of audio by an Ear Canal Receiver (ECR) in an earpiece to an ear canal based on the target sound.

	Claim 8. The method of claim 7, further comprising: **passing the target sound to the ECR** for delivery to the ear canal.

Claim Rejections - 35 USC § 102

6. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

7. Claims 1-2, 4-5, 8-9 and 13-14 are rejected under 35 U.S.C. 102(e) as being anticipated by Goldstein et al (Pub. No.: US 2008/0240458 A1), hereinafter as Goldstein.

The applied reference has a common assignee with the instant application. Based upon the earlier effective U.S. filing date of the reference, it constitutes prior art under 35 U.S.C. 102(e). This rejection under 35 U.S.C. 102(e) might be overcome either by a showing under 37 CFR 1.132 that any invention disclosed but not claimed in the reference was derived from the inventor of this application and is thus not the invention “by another,” or by an appropriate showing under 37 CFR 1.131.

Regarding claim 1, Goldstein discloses an earpiece in Figs. 1- 3, comprising: an Ambient Sound Microphone (ASM) (110) configured to capture ambient sound (see Fig. 2 and [0025]) ; at least one Ear Canal Receiver (ECR) (120) configured to deliver audio to an ear canal (see Fig. 2 and [0025]); and a processor (206) operatively coupled to the ASM and the at least one ECR, where the processor is configured to monitor a change in an ambient sound level for detecting an acute sound from the change, and where upon detecting the acute sound, reproducing the acute sound within the ear canal via the ECR (see Fig. 2 and [0028]).

Regarding claim 2, Goldstein discloses the earpiece of claim 1, wherein the processor passes sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal (adjusts the properties of sound delivered to the ear canal so the sound within the occluded ear canal is the same as the ambient sound outside the ear (see [0040]).

Regarding claim 4, Goldstein discloses the earpiece of claim 1, wherein the processor measures an external ambient sound level (xASL) of the ambient sound with the ASM and subtracts an attenuation level of the earpiece from the xASL to estimate the internal ambient sound level (iASL) within the ear canal (ear seal profile can be generated by evaluating amplitude and phase difference between the ASM 110 and the ECM 202 for known signals produced by ECR 120) (see [0037]).

Regarding claim 5, Goldstein discloses the earpiece of claim 1, The earpiece of claim 1, further comprising: an Ear Canal Microphone (ECM) to measure an ear canal

sound level (ECL) within the ear canal, wherein the processor estimates an internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL.

Regarding claim 8, Goldstein discloses an earpiece in Figs. 1-3, comprising: an Ambient Sound Microphone (ASM) (110) configured to capture ambient sound (see Fig. 2 and [0025]); at least one Ear Canal Receiver (ECR) (120) configured to deliver audio to an ear canal (see Fig. 2 and [0025]); an audio interface (212) configured to receive audio content; and a processor operatively coupled to the ASM (see Fig. 2 and [0029]), the at least one ECR, and the audio interface, where the processor is configured to monitor a change in the ambient sound level to detect an acute sound from the change, where the processor adjusts an audio content level (ACL) of an audio content delivered to the ear canal, and where upon detecting an acute sound, the acute sound is reproduced within the ear canal via the ECR (see [0029]).

Regarding claim 9, Goldstein discloses the earpiece of claim 8, wherein the audio interface receives the audio content from at least one among a portable music player, a cell phone, and a portable communication device (see Fig. 2).

Regarding claim 13, Goldstein discloses a method for acute sound detection and reproduction in Fig. 1-3, the method comprising: measuring an ambient sound level (xASL) external (by ASM 110) to an ear at least partially occluded by an earpiece (see Fig. 2 and [0025]); monitoring a change in the xASL for detecting an acute sound (sound signatures for previously learned target sounds) (see [0028]); and upon

detecting the acute sound, reproducing the acute sound within the ear canal (report the target to the user via audio delivered from ECR 1200) (see [0028]).

Regarding claim 14, Goldstein discloses the method of claim 13, wherein the step of reproducing produces sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal (see [0040]).

Claim Rejections - 35 USC § 103

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

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

3. Claims 1-6 and 8-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Svean et al (U.S Patent No. 6,754,359 B1), hereinafter as Svean in view of Rothschild (US Pub No. 2006/0083388 A1).

Regarding claim 1, Svean discloses an earpiece in Figs. 1- 3, comprising: an Ambient Sound Microphone (ASM) (M1) configured to capture ambient sound (see Fig. 1, column 6 and lines 35-32); at least one Ear Canal Receiver (ECR) (SG) configured to deliver audio to an ear canal (see Fig. 1, column 4 and lines 45-50); and a processor (DSP E3) operatively coupled to at least one of the ASM and the ECR (see Fig. 2).

Svean might not disclose the earpiece comprising: where the processor is configured to monitor a change in an ambient sound level for detecting an acute sound

from the change, and where upon detecting the acute sound, reproducing the acute sound within the ear canal via the ECR.

Rothschild discloses earpiece system in Fig. 1, comprising: an Ambient Sound Microphone (MID 122) to capture ambient sound (see [0023]), an audio output device (AOD 106) for audibly producing sounds (see [0019]), an audio signal processing control unit (ASPCU 104) for monitoring a change in an ambient sound level for detecting an acute sound from the change, and where upon detecting the acute sound, reproducing the acute sound to the audio output device (see [0024], [0027] and [0028]).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify the audio signal processing control unit of earpiece system of Rothschild into the earpiece of Svean. This is so because it would allow the user to hear external noises outside the user's listening environment when needed by selective switched sound processing (see Rothschild and [0007]).

Regarding claim 2, the combination of Svean and Rothschild discloses the earpiece of claim 1, wherein the processor passes sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal (see Rothschild and [0024]).

Regarding claim 3, the combination of Svean and Rothschild discloses the earpiece of claim 1, and Rothschild discloses the earpiece can lower the volume of the audio content and increase the volume of ambient sound signals (see [0032]). Therefore, due to changes of the level of ambient sound the processor can maintains an

approximately constant ratio between an audio content level (ACL) presented to the earpiece and an internal ambient sound level (iASL) measured within the ear canal.

Regarding claim 4, the combination of Svean and Rothschild discloses the earpiece of claim 1, Svean discloses wherein the processor measures an external ambient sound level (xASL) of the ambient sound with the ASM (M1 pick up external sounds) (see column 5 and lines 40-43) and the internal ambient sound level (iASL) within the ear canal (M2 picks up the sound in the meatus 3) (see column 6 and lines 39-41). Therefore, the internal ambient sound level (iASL) can also be estimating by subtracting a loss of propagated signal (attenuation level) from the external ambient sound.

Regarding claim 5, the combination of Svean and Rothschild discloses the earpiece of claim 1, Svean disclose the earpiece further comprising: an Ear Canal Microphone (ECM) (microphone M2 and T1) to measure an ear canal sound level (ECL) within the ear canal (see Fig. 1, column 5 and lines 42-44), wherein the processor estimates an internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL (see rejection of claim 4).

Regarding claim 6, the combination of Svean and Rothschild discloses the earpiece of claim 5, wherein the processor measures a voltage level of the audio content sent to the ECR (lowering the volume of the audio content must involve measurement of voltage level) (see Rothschild, Fig. 1 and [0032]), and applies a transfer function of the ECR to convert the voltage level to the ACL (see Svean, column 8 and lines 30-31).

Regarding claim 8, Svean discloses an earpiece in Figs. 1- 3, comprising: an Ambient Sound Microphone (ASM) (M1) configured to capture ambient sound (see Fig. 1, column 6 and lines 35-32); at least one Ear Canal Receiver (ECR) (SG) configured to deliver audio to an ear canal (see Fig. 1, column 4 and lines 45-50); a digital interface (E12) for connecting to other electrical units by a cable or wireless (see Fig. 2, column 6 and lines 59-63); and a processor (DSP E3) operatively coupled to at least one of the ASM, the ECR and the digital interface (see Fig. 2).

Svean might not disclose where the processor is configured to monitor a change in the ambient sound level to detect an acute sound from the change, where the processor adjusts an audio content level (ACL) of an audio content delivered to the ear canal, and where upon detecting an acute sound, the acute sound is reproduced within the ear canal via the ECR.

Rothschild discloses earpiece system in Fig. 1, comprising: an Ambient Sound Microphone (MID 122) to capture ambient sound (see [0023]), an audio interface (116 and 118) configured to receive audio content (see Fig. 1), an audio output device (AOD 106) for audibly producing sounds (see [0019]), an audio signal processing control unit (ASPCU 104) for monitoring a change in an ambient sound level for detecting an acute sound from the change (see [0024] and [0027]), where the processor adjusts an audio content level (ACL) of an audio content delivered to the ear canal (see [0032]) and where upon detecting the acute sound, reproducing the acute sound to the audio output device (see [0032]).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify the audio signal processing control unit of earpiece system of Rothschild into the earpiece of Svean. This is so because it would allow the user to hear external noises outside the user's listening environment when needed by selective switched sound processing (see Rothschild and [0007]).

Regarding claim 9, the combination of Svean and Rothschild discloses the earpiece of claim 8, wherein the audio interface receives the audio content from at least one among a portable music player, a cell phone, and a portable communication device (see Rothschild and [0020]).

Regarding claim 10, the combination of Svean and Rothschild discloses the earpiece of claim 8, and Rothschild discloses the earpiece can lower the volume of the audio content and increase the volume of ambient sound signals (see [0032]). Therefore, due to the level of ambient sound the processor can maintains an approximately constant ratio between an audio content level (ACL) presented to the earpiece and an internal ambient sound level (iASL) measured within the ear canal.

Regarding claim 11, the combination of Svean and Rothschild discloses the earpiece of claim 10, wherein the processor mutes the audio content (pause the audio content) and passes the acute sound to the ECR for reproducing the acute sound within the ear canal (see Rothschild, [0031] and [0033]).

Regarding claim 12, the combination of Svean and Rothschild discloses the earpiece of claim 10, wherein the processor amplifies the acute sound (play the sounds

from MID at higher volume) with respect to the audio content level (ACL) (see Rothschild and [0032]).

Regarding claim 13, Svean discloses a method for ambient sound detection and reproduction the method comprising: measuring an ambient sound level (xASL) (by M1) external to an ear at least partially occluded by an earpiece (see Fig. 1, column 6 and lines 35-32); delivering the audio to an ear canal (by SG) (see Fig. 1, column 4 and lines 45-50).

Svean might not disclose a method for monitoring a change in the xASL for detecting an acute sound; and upon detecting the acute sound, reproducing the acute sound within the ear canal.

Rothschild discloses a method for ambient sound detection and reproduction comprising: capturing ambient sound (by MID 122) (see [0019]); monitoring a change in the xASL for detecting an acute sound (see [0027]); and upon detecting the acute sound, reproducing the acute sound within the ear canal (see [0024] and [0028]).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify the method of Rothschild into the method of Svean. this so because it would allow the user to hear external noises outside the user's listening environment when needed by selective switched sound processing (see Rothschild and [0007]).

Regarding claim 14, the combination of Svean and Rothschild discloses the method of claim 13, wherein the step of reproducing produces sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal (see Rothschild and [0024]).

Regarding claim 15, the combination of Svean and Rothschild discloses the method of claim 13, further comprising: receiving audio content from an audio interface that is directed to the ear canal (see Svean, Fig. 2, column 6, lines 59-67). Rothschild discloses the earpiece can lower the volume of the audio content and increase the volume of ambient sound signals (see [0032]). Therefore, due to the change of level of ambient sound the method can maintain an approximately constant ratio between a level of the audio content (ACL) and a level of an internal ambient sound level (iASL) measured within the ear canal.

Regarding claim 16, the combination of Svean and Rothschild discloses the method of claim 15, wherein the ACL is determined by measuring a voltage level of the audio content sent to the ECR (lowering the volume of the audio content must involve measurement of voltage level) (see Rothschild, Fig. 1 and [0032]), and applies a transfer function of the ECR to convert the voltage level to the ACL (see Svean, column 8 and lines 30-31).

Regarding claim 17, the combination of Svean and Rothschild discloses the method of claim 15, Svean discloses measuring an Ear Canal Level (ECL) (by M2 and T1) within the ear canal (see Fig. 1, column 5 and lines 42-44); measuring an external ambient sound level (xASL) of the ambient sound with the ASM (M1 pick up external sounds) (see column 5 and lines 40-43) and the internal ambient sound level (iASL) within the ear canal (M2 picks up the sound in the meatus 3) (see column 6 and lines 39-41). Therefore, the internal ambient sound level (iASL) can also be estimating by

subtracting a loss of propagated signal (attenuation level) from the external ambient sound. and subtracting the ACL from the ECL to estimate the iASL.

Regarding claim 18, the combination of Svean and Rothschild discloses the method of claim 15, wherein the iASL is estimated by subtracting an attenuation level of the earpiece from the xASL (see the rejection of claim 17).

Regarding claim 19, the combination of Svean and Rothschild discloses the method of claim 15, muting the audio content (pause the audio content) and passing the acute sound to an Ear Canal Receiver (ECR) (SG and T2) (see Svean, Fig.1) for reproducing the acute sound within the ear canal (see Rothschild, [0031] and [0033].

Regarding claim 20, the combination of Svean and Rothschild discloses the method of claim 15, further comprising: amplifying the acute sound (play the sounds from MID at higher volume) with respect to the audio content level (ACL) (see Rothschild and [0032]).

4. Claim 7 is rejected under 35 U.S.C. 103(a) as being unpatentable over Svean et al (U.S Patent No. 6,754,359 B1), hereinafter as Svean in view of Rothschild (US Pub No. 2006/0083388 A1) and further in view of Anderson (US Patent No.: 5,721,783).

Regarding claim 7, the combination of Svean and Rothschild discloses the earpiece of claim 1, but might not disclose wherein the processor is located external to the earpiece on a portable computing device.

Anderson discloses an earpiece (10) in Fig. 1 and 7 comprising: a remote processor unit (16) (see Fig. 1, column 4 and lines 1-3) is on a portable computing device (RPU case 70) (see Fig. 7, column 20 and lines 60-65).

It would have been obvious to one of ordinary skill in the art at the time of invention to modify the processor of Svean to be on a portable computing device. This is so because it would allow the earpiece having a small size and light weight and freedom movement with a cost effective design (see Anderson, column 2 and lines 33-39).

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to CUONG B. NGUYEN whose telephone number is (571)270-1509. The examiner can normally be reached on Monday to Friday 8:30am-5:00pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor Steven Loke can be reached on 571-272-1657. 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.

CN

/STEVEN LOKE/
Supervisory Patent Examiner, Art Unit 2818

Exhibit PAT-C

Notice of References Cited	Application/Control No. 12/017,878	Applicant(s)/Patent Under Reexamination GOLDSTEIN ET AL.	
	Examiner CUONG NGUYEN	Art Unit 2818	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	Classification
*	A	US-6,754,359 B1	06-2004	Svean et al.	381/328
*	B	US-2006/0083388 A1	04-2006	Rothschild, Leigh M.	381/081
*	C	US-2008/0240458 A1	10-2008	Goldstein et al.	381/72
	D	US-			
	E	US-			
	F	US-			
	G	US-			
	H	US-			
	I	US-			
	J	US-			
	K	US-			
	L	US-			
	M	US-			

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	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.

Receipt date: 08/21/2008 Exhibit PAT-C

Form PTO-1449	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTY. DKT. NO. 092624.011700	SERIAL NO. 12/017,878
INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Use several sheets if necessary)		APPLICANTS Steven W. Goldstein et al.	EXAMINER: /Cuong Nguyen/ ~~Unassigned~~
		FILING DATE January 22, 2008	GROUP ~~2615~~

12017878 - GAU: 2818

U.S. PATENT DOCUMENTS

Examiner's Initial		Document No.	Publication Date	Name	Class	Subclass	Filing Date
/C.N./	AA	2006/0045282	03/02/2006	Reber			
/C.N./	AB	2001/0046304	11/29/2001	Rast			
	AC						
	AD						
	AE						
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	AK						
	AL						
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	AO						
	AP						
	AQ						
	AR						

FOREIGN PATENT DOCUMENTS

Examiner's Initial		Document Number	Date	Country	Class	Subclass	Translation Yes	Translation No
	AS							
	AT							
	AU							
	AV							

OTHER DOCUMENTS (Including Author, Title, Date, Pertinent Pages, Etc.)

	AW	
	AX	
	AY	
	AZ	

Examiner /Cuong Nguyen/	**Date Considered** 01/11/2012

ALL REFERENCES CONSIDERED EXCEPT WHERE LINED THROUGH. /C.N./

Exhibit PAT-C

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Appln. No:	12/017,878
Applicant:	Steven W. Goldstein et al.
Filed:	January 22, 2008
Title:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
T.C./A.U.:	2818
Examiner:	Cuong B. Nguyen
Confirmation No.:	7135
Docket No.:	PRS-118US

AMENDMENT

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

Sir:

Responsive to the Office Action dated January 18, 2012, please amend the above-identified application as follows:

☒ **Amendments to the Specification** begin on page 2 of this paper.

☒ **Amendments to the Claims** are reflected in the listing of claims which begins on page 3 of this paper.

☒ **Amendments to the Drawings** begin on page 8 of this paper and include an attached replacement sheet(s).

☒ **Amendments to the Abstract** are on page 9 of this paper. A clean version of the Abstract is on page 15 of this paper.

☒ **Remarks/Arguments** begin on page 10 of this paper.

Exhibit PAT-C

Amendments to the Specification:

Please replace the original specification with the attached Substitute Specification.

Exhibit PAT-C

Amendments to the Claims: This listing of claims will replace all prior versions, and listings, of claims in the application

Listing of Claims:

1. (Currently Amended) An earpiece comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound, the ambient sound containing an acute sound having an abrupt high energy sound profile;

at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal; and

a processor operatively coupled to the ASM and the at least one ECR, ~~where~~ the processor ~~is~~being configured to:

monitor a change in an ambient sound level ~~for detecting an acute sound from the change, and where~~ of the ambient sound,

detect an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold, and

upon detecting the on-set of the acute sound, ~~reproducing~~reproduce the acute sound within the ear canal via the ECR.

2. (Currently Amended) The earpiece of claim 1, wherein the processor passes the ambient sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal.

3. (Original) The earpiece of claim 1, wherein the processor maintains an approximately constant ratio between an audio content level (ACL) presented to the earpiece and an internal ambient sound level (iASL) measured within the ear canal.

4. (Currently Amended) The earpiece of claim 1, wherein the processor measures an external ambient sound level (xASL) of the ambient sound with the ASM and subtracts an attenuation level of the earpiece from the xASL to estimate ~~the~~an internal ambient sound level (iASL) within the ear canal.

5. (Original) The earpiece of claim 1, further comprising:

an Ear Canal Microphone (ECM) to measure an ear canal sound level (ECL) within the ear canal,

wherein the processor estimates an internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL.

6. (Currently Amended) The earpiece of claim 5, wherein the processor measures a voltage level of ~~the~~ audio content sent to the ECR, and applies a transfer function of the ECR to convert the voltage level to the ACL.

7. (Original) The earpiece of claim 1, wherein the processor is located external to the earpiece on a portable computing device.

8. (Currently Amended) An earpiece, comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound, the ambient sound containing an acute sound having an abrupt high energy sound profile;

at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal;

an audio interface configured to receive audio content; and

a processor operatively coupled to the ASM, the at least one ECR, and the audio interface, ~~where~~ the processor ~~is~~being configured to:

monitor a change in ~~the~~an ambient sound level of the ambient sound,

~~to~~ detect an on-set of the acute sound ~~from the change, where the processor adjusts~~when the ambient sound level is greater than a predetermined threshold,

adjust an audio content level (ACL) of ~~an~~the audio content delivered to the ear canal, and ~~where~~

upon detecting ~~an~~the on-set of the acute sound, ~~the acute sound is reproduced~~reproduce the acute sound within the ear canal via the ECR.

9. (Original) The earpiece of claim 8, wherein the audio interface receives the audio content from at least one among a portable music player, a cell phone, and a portable communication device.

10. (Original) The earpiece of claim 8, wherein the processor maintains an approximately constant ratio between the audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal.

11. (Original) The earpiece of claim 10, wherein the processor mutes the audio content and passes the acute sound to the ECR for reproducing the acute sound within the ear canal.

12. (Original) The earpiece of claim 10, wherein the processor amplifies the acute sound with respect to the audio content level (ACL).

13. (Currently Amended) A method for acute sound detection and reproduction the method comprising:

measuring an ambient sound level (xASL) of ambient sound external to an ear at least partially occluded by an earpiece, the ambient sound containing an acute sound having an abrupt high energy sound profile;

monitoring a change in the xASL ~~for detecting an acute sound; and~~ of the ambient sound;

detecting an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold; and

upon detecting the on-set of the acute sound, reproducing the acute sound within ~~the~~an ear canal.

14. (Original) The method of claim 13, wherein the step of reproducing produces sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal.

15. (Currently Amended) The method of claim 13, further comprising:

receiving audio content from an audio interface that is directed to the ear canal;

maintaining an approximately constant ratio between a level of the audio content (ACL) and ~~a level of~~ an internal ambient sound level (iASL) measured within the ear canal.

16. (Currently Amended) The method of claim 15, wherein the ACL is determined by

measuring a voltage level of the audio content directed ~~sent~~ to the ear canal, the audio content being directed via an Ear Canal Receiver ~~ECR~~(ECR); and

applying a transfer function of the ECR to convert the voltage level to the ACL.

17. (Original) The method of claim 15, further comprising:

measuring an Ear Canal Level (ECL) within the ear canal; and

subtracting the ACL from the ECL to estimate the iASL.

18. (Original) The method of claim 15, wherein the iASL is estimated by subtracting an attenuation level of the earpiece from the xASL.

19. (Original) The method of claim 15, further comprising:

muting the audio content and passing the acute sound to an Ear Canal Receiver (ECR) for reproducing the acute sound within the ear canal.

20. (Original) The method of claim 15, further comprising:

amplifying the acute sound with respect to the audio content level (ACL).

21. (Withdrawn) A method for acute sound detection and reproduction, the method comprising:

measuring an external ambient sound level (xASL);

monitoring a change in the xASL for detecting an acute sound;

estimating a proximity of the acute sound; and

upon detecting the acute sound and it's proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece.

22. (Withdrawn) The method of claim 21, wherein the step of estimating a proximity comprises:

performing a cross correlation analysis between at least two microphones;

identifying a peak in the cross correlation and an associated time lag; and

determining the direction from the associated time lag.

23. (Withdrawn) The method of claim 21, further comprising identifying whether the acute sound is a vocal signal produced by a user operating the earpiece or a sound source external from the user.

24. (Withdrawn) A method for acute sound detection and reproduction the method comprising:

measuring an external ambient sound level (xASL) due to ambient sound outside of an ear canal at least partially occluded by the earpiece;

measuring an internal ambient sound level (iASL) due to residual ambient sound within the ear canal at least partially occluded by the earpiece;

monitoring a high frequency change between the xASL and the iASL with respect to a low frequency change between the xASL and the iASL for detecting an acute sound; and

reproducing the xASL within the ear canal responsive to detecting the high frequency change.

25. (Withdrawn) The method of claim 24, further comprising determining a proximity of a sound source producing the acute sound.

Exhibit PAT-C

Amendments to the Drawings:

The attached sheets of drawings include changes to Figures 1, 2, 6, 8. These sheets replace the original sheets.

Attachment

Exhibit PAT-C

Amendment to the Abstract:

The Abstract has been amended. A revised Abstract is attached.

ABSTRACT

~~A method~~Earpieces and methods for acute sound detection and reproduction ~~is~~are provided ~~that~~. A method can include measuring an ambient sound level external to an ear canal at least partially occluded by the earpiece, monitoring a change in the ambient sound level for detecting an acute sound, estimating a proximity of the acute sound, and reproducing the acute sound within the ear canal responsive to detecting the acute sound and the proximity. ~~Other embodiments are disclosed.~~

Attachment