Exhibit PAT-C

Remarks/Arguments:

Claims 1, 2, 4, 6, 8, 13, 15 and 16 have been amended. No new matter is introduced herein. Of pending claims 1-25, claims 21-25 have been withdrawn.

There appears to be an error in the Office Action. Namely, the PTO-892 form included with the Office Action did not indicate Anderson (US 5,721,783), which was a reference considered in the Office Action. The Examiner is respectfully requested to furnish a new PTO-892 form with citations to all considered references.

Claims Amendments

Claim 1 has been amended to clarify that: the ambient sound contains an acute sound having an abrupt high energy sound profile; that an on-set of the acute sound is detected when the ambient sound level is greater than a predetermined threshold; and reproducing the acute sound upon detecting the on-set of the acute sound. Claims 8 and 13 have been amended similarly to claim 1. No new matter is introduced herein. Support for the amendments may be found, for example, at Paragraphs [0036], [0046], [0047], [0055] and [0056]; and Figs. 3 and 4 of the original specification.

Claims 2, 4, 6, 15 and 16 have been amended to clarify the language and/or for antecedent basis. No new matter is introduced herein.

Specification Amendments

A substitute specification has been included to correct for typographical errors and to correspond with the figures. No new matter is introduced herein.

Abstract Amendments

The Abstract has been amended to clarify the language and to remove any legal phraseology. No new matter is introduced herein.

Drawing Amendments

Figs. 1, 2, 6 and 8 have been amended. In Fig. 1, feature 100, described, for example, at paragraphs [0033] and [0034] of the original specification, was not included in Fig. 1. In Fig. 2, features 111, 123 and 125 were mistakenly labeled as "110," "130" and "120" respectively. In Fig. 6, feature 636 was mistakenly labeled as "626." In Fig. 8, audio content 820 was mistakenly labeled as "audio content level." These features have been amended accordingly. No new matter is introduced herein.

Claim Objections

Claims 1 and 8 have been objected to because it is asserted that the phrase "the at least one ECR" is grammatically incorrect. Applicants respectfully disagree. Applicants note that the

phrase "the at least one ECR" is required for antecedent basis. (See MPEP § 2173.05(e).) Thus, the phrase "the at least one ECR" is proper. Accordingly, Applicants respectfully request that the objection to claims 1 and 8 be withdrawn.

Double Patenting Rejections

Claims 1, 8 and 13 have been provisionally rejected on the ground of nonstatutory obviousness-type double patenting as being unapatentable over claims 1 and 6-8 of the co-pending application no. 11/966,457. It is submitted that because this is a provisional double patenting rejection, Applicants are not required to substantively respond to the provisional rejection until the cited application issues and the rejection is thus, no longer provisional.

§102, 103 Rejections

Claims 1, 2, 4, 5, 8, 9, 13 and 14 have been rejected under 35 U.S.C. §102(e) as being anticipated by Goldstein et al. (US 2008/0240458, referred to herein as Goldstein). Claims 1-6 and 8-20 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Svean et al. (US 6,754,359, referred to herein as Svean) in view of Rothschild (US 2006/0083388). It is respectfully submitted, however, that these claims are patentable over the cited art for the reasons set forth below.

Claim 1 includes features neither disclosed nor suggested by the cited art, namely:

> monitor a change in an ambient sound level 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, reproduce the acute sound within the ear canal via the ECR.

Goldstein discloses, in Fig. 3, a method for earpiece monitoring and target detection which detects target sounds (such as an alarm, a horn, a voice or a noise) within the environment based on sound signatures (steps 302 and 304). The sound signatures relate to identifiable features that characterize the sound, such as statistical properties or parametric properties of the target sound. Feature extraction techniques are applied to the ambient sound to generate the sound signatures. The processor can adjust sound delivered to the ear canal in view of a detected target sound (step 306). (Paragraphs [0034 – 0036].)

Goldstein, however, does not disclose or suggest detecting an on-set of an acute sound (having an abrupt high energy sound profile) when the ambient sound level is greater than a predetermined threshold; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1 (emphasis added). Instead, Goldstein monitors the target

sounds for identifiable features, i.e., a sound signature, with pattern recognition approaches being applied to detect the target sounds. Thus, Goldstein does not include all of the features of claim 1.

Svean discloses, in Fig. 1, an earplug including sealing section 2, microphone M1 (for picking up external sound), microphone M2 (for picking the acoustic signal in meatus 3) and sound generator SG. (Col. 4, lines 29-34 and col. 5, lines 29-51.) Svean discloses, in Fig. 2, electric circuitry 11 of the earplug including processing unit E3. (Col. 6, lines 45-51.)

As acknowledged by the Examiner on pages 10-11 of the Office Action, Svean does not disclose or suggest that the processor is configured to monitor a change in an ambient sound level for detecting an acute sound from the change and reproducing the acute sound within the ear canal via the ECR upon detecting the acute sound. Accordingly, Svean cannot disclose or suggest detecting an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1. Svean is silent regarding these features. Thus, Svean does not include all of the features of claim 1.

Rothschild discloses, in Fig. 1, system 100 for selectably switching between a plurality of audio channels. System 100 includes audio content device (ACD) 102, audio signal processing control unit 104, microphone input device (MID) 122 and audio output (AOD) device 106. Control unit 104 listens for predetermined sounds (such as from MID 122) and, if necessary, interrupts the audio content from ACD 102 to play the externally generated sound to the user. Control unit 104 includes switching mechanism 128 to select which channel of audio is played to the user. (Paragraphs [0019], [0023] and [0024].) Microprocessor 124 of control unit 104 monitors ACD 102 and MID 122 inputs for preselected sound patterns (such as human voice patterns, noise patterns or other sound patterns). The patterns can be recognized such as by speech recognition algorithms. (Paragraphs [0027] and [0028].) Control unit 104 can also lower the volume of the audio content from ACD 102, instead of completely switching between channels. (Paragraph [0032].)

Rothschild, however, does not disclose or suggest detecting an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1 (emphasis added). Instead, Rothschild monitors audio inputs for preselected sound patterns, where the sound patterns can be recognized by speech recognition algorithms. Thus, Rothschild does not include all of the features of claim 1, nor does Rothschild make up for the deficiencies of Goldstein or Svean with respect to claim 1.

In summary, none of the cited art disclose or suggest detecting an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1. These features are absent from the cited art. Accordingly, allowance of claim 1 is respectfully requested.

Claims 8 and 13, although not identical to claim 1, includes features similar to claim 1 which are neither disclosed nor suggested by the cited art. Accordingly, these claims are also patentable over the cited art for at least the same reasons as claim 1.

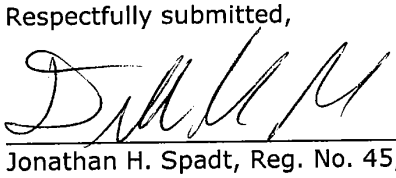
Claims 2-6, 9-12 and 14-20 include all of the features of respective claims 1, 8 and 13 from which they depend. Accordingly, claims 2-6, 9-12 and 14-20 are also patentable over the cited art for at least the same reasons as respective claims 1, 8 and 13.

Claim 7 has been rejected under 35 U.S.C. §103(a) as being unpatentable over Svean in view of Rothschild and further in view of Anderson (US 5,721,783). This claim, however, includes all of the features of claim 1 from which it depends and is patentable over Svean and Rothschild for at least the same reasons as claim 1. Anderson does not make up for the deficiencies of Svean and Rothschild with respect to claim 1. Accordingly, claim 7 which depends from claim 1, is also patentable over Svean and Rothschild combined with Anderson for at least the same reasons as claim 1.

Exhibit PAT-C

In view of the amendments and argument set forth above, the above-identified application is in condition for allowance which action is respectfully requested.

Respectfully submitted,



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JHS/DMG/emp

Attachments: Figures 1, 2, 6 and 8 (3 sheets)
Abstract
Substitute Specification

Dated: May 18, 2012

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The Director is hereby authorized to charge or credit Deposit Account No. **18-0350** for any additional fees, or any underpayment or credit for overpayment in connection herewith.

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Exhibit PAT-C

ABSTRACT

Earpieces and methods for acute sound detection and reproduction are provided. 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.

Exhibit PAT-C

PRS-118US

Substitute Specification

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This Application is a Non-Provisional and claims the priority benefit of Provisional Application No. 60/885,917 filed on January 22, 2007, the entire disclosure of which is incorporated herein by reference.

FIELD

[0002] 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

[0003] 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.

[0004] 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.

[0005] 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.

[0006] 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 ~~level~~levels to compensate for ambient noise.

[0007] A need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties.

SUMMARY

[0008] Embodiments in accordance with the present invention provide a method and device for acute sound detection and reproduction.

[0009] 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.

[0010] 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.

[0011] 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.

[0012] 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.

[0013] 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 ~~passes~~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).

[0014] 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.

[0015] 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.

[0016] 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.

[0017] 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

[0018] FIG. 1 is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

[0019] FIG. 2 is a block diagram of the earpiece in accordance with an exemplary embodiment;

[0020] FIG. 3 is a flowchart of a method for acute sound detection in accordance with an exemplary embodiment;

[0021] FIG. 4 is a more detailed approach to the method of FIG. 3 in accordance with an exemplary embodiment;

[0022] FIG. 5 is a flowchart of a method for acute sound source proximity in accordance with an exemplary embodiment;

[0023] FIG. 6 is a flowchart of a method for binaural analysis in accordance with an exemplary embodiment;

[0024] FIG. 7 is a flowchart of a method for logic control in accordance with an exemplary embodiment;

[0025] FIG. 8 is a flowchart of a method for estimating background noise level in accordance with an exemplary embodiment;

[0026] 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

[0027] FIG. 10 is a flowchart of a method for adjusting audio content gain in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

[0028] 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.

[0029] 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.

[0030] 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.

[0031] 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.

[0032] 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.

[0033] 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.

[0034] 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 ~~users~~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 5cc 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 ~~users~~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.

[0035] 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 ~~ear seal~~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.

[0036] 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 ~~110~~111, ECR ~~120~~125, and ECM ~~130~~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 ~~110~~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 ~~a~~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.

[0037] 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.

[0038] 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.

[0039] 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.

[0040] 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.

[0041] 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~~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 ~~device~~devices.

[0042] 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, ~~an~~and robots to name a few.

[0043] 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 22-26dB noise reduction. However, portions of ambient sounds higher than 26dB 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.

[0044] 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.

[0045] 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.

[0046] 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.

[0047] 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.

[0048] 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:

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

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

[0049] 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.

[0050] 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 ~~can~~ processor 206 measures a voltage level of the audio content sent to the ECR 125, and ~~apply~~applies a transfer function of the ECR 125 to convert the voltage level to the ACL.

[0051] 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.

[0052] 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.

[0053] 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~~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 ~~device~~devices.

[0054] 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.

[0055] 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 ~~hold~~ 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" 40dB) 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.

[0056] Briefly, ~~I~~if a transient, high-level sound (or acute sound) is ~~detecting~~detected in the ambient sound signal (ASM input signal), then it is converted to a level, and ~~it's~~its magnitude compared with the BNL is calculated. The magnitude of this resulting difference (signal "A") is compared with the threshold (see step ~~424~~422). If the value is positive, and the level of the transient is greater than a predefined threshold (see step ~~430~~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.

[0057] 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~~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 ~~device~~devices.

[0058] Briefly, FIG. 5 describes a method 500 for Source Proximity ~~Detector~~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 a~~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 a~~from left ~~a~~and right earpiece 100 ~~assembly~~assemblies placed ~~on a~~on left and right ~~ear~~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.

[0059] 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 ~~522~~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.

[0060] 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~~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 ~~device~~devices.

[0061] 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. ~~a~~, left and right ASM

~~signal~~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 ~~network~~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 ~~433~~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~~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 ~~440~~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.

[0062] 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~~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 ~~device~~devices.

[0063] 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-~~512~~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 ~~720~~722 if so. A safe level check may be performed by processor 206 at step 720.

[0064] 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~~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 ~~device~~devices.

[0065] 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 ~~810~~ (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~~ 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.

[0066] 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~~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 ~~device~~devices.

[0067] 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.

[0068] 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.

[0069] 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 ~~AGC's~~AGCs 928 and 936, the two signals are mixed at step 940.

[0070] 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~~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.

[0071] 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~~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 ~~device~~devices.

[0072] Method 950 describes calculation of AGC coefficients. The method 950 receives as ~~it's~~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 ~~958~~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 ~~mode decision~~decision module 976 (step 932~~)~~ in FIG. 9).

[0073] 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.

Exhibit PAT-C

PRS-118US

Clean Specification

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This Application is a Non-Provisional and claims the priority benefit of Provisional Application No. 60/885,917 filed on January 22, 2007, the entire disclosure of which is incorporated herein by reference.

FIELD

[0002] 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

[0003] 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.

[0004] 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.

[0005] 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.

[0006] 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.

[0007] A need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties.

SUMMARY

[0008] Embodiments in accordance with the present invention provide a method and device for acute sound detection and reproduction.

[0009] 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.

[0010] 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.

[0011] 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.

[0012] 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.

[0013] 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).

[0014] 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.

[0015] 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.

[0016] 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.

[0017] 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

[0018] FIG. 1 is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

[0019] FIG. 2 is a block diagram of the earpiece in accordance with an exemplary embodiment;

[0020] FIG. 3 is a flowchart of a method for acute sound detection in accordance with an exemplary embodiment;

[0021] FIG. 4 is a more detailed approach to the method of FIG. 3 in accordance with an exemplary embodiment;

[0022] FIG. 5 is a flowchart of a method for acute sound source proximity in accordance with an exemplary embodiment;

[0023] FIG. 6 is a flowchart of a method for binaural analysis in accordance with an exemplary embodiment;

[0024] FIG. 7 is a flowchart of a method for logic control in accordance with an exemplary embodiment;

[0025] FIG. 8 is a flowchart of a method for estimating background noise level in accordance with an exemplary embodiment;

[0026] 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

[0027] FIG. 10 is a flowchart of a method for adjusting audio content gain in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

[0028] 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.

[0029] 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.

[0030] 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.

[0031] 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.

[0032] 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.

[0033] 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.

[0034] 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 5cc 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.

[0035] 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.

[0036] 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.

[0037] 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.

[0038] 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.

[0039] 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.

[0040] 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.

[0041] 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.

[0042] 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.

[0043] 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 22-26dB noise reduction. However, portions of ambient sounds higher than 26dB 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.

[0044] 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.

[0045] 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.

[0046] 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.

[0047] 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.

[0048] 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:

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

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

[0049] 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.

[0050] 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.

[0051] 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.

[0052] 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.

[0053] 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.

[0054] 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.

[0055] 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" 40dB) 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.

[0056] 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.

[0057] 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.

[0058] 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.

[0059] 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.

[0060] 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.

[0061] 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.

[0062] 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.

[0063] 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.

[0064] 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.

[0065] 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.

[0066] 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.

[0067] 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.

[0068] 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.

[0069] 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.

[0070] 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.

[0071] 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.

[0072] 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).

[0073] 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.

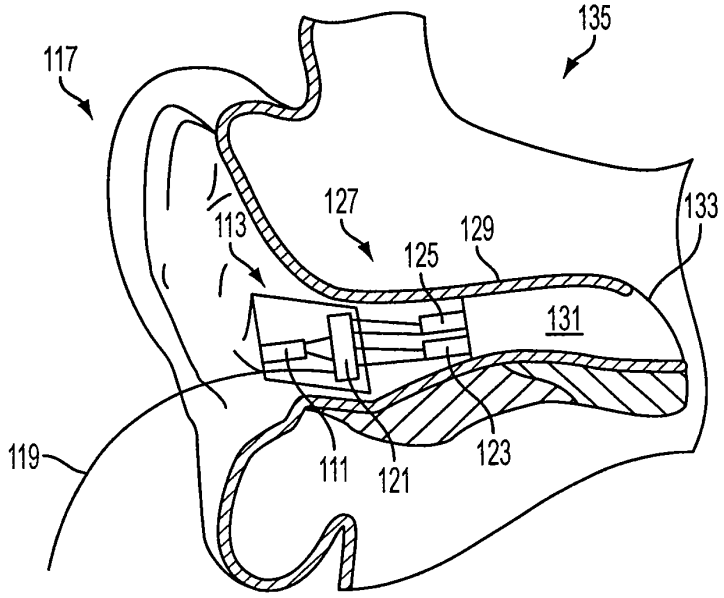


FIG. 1 100

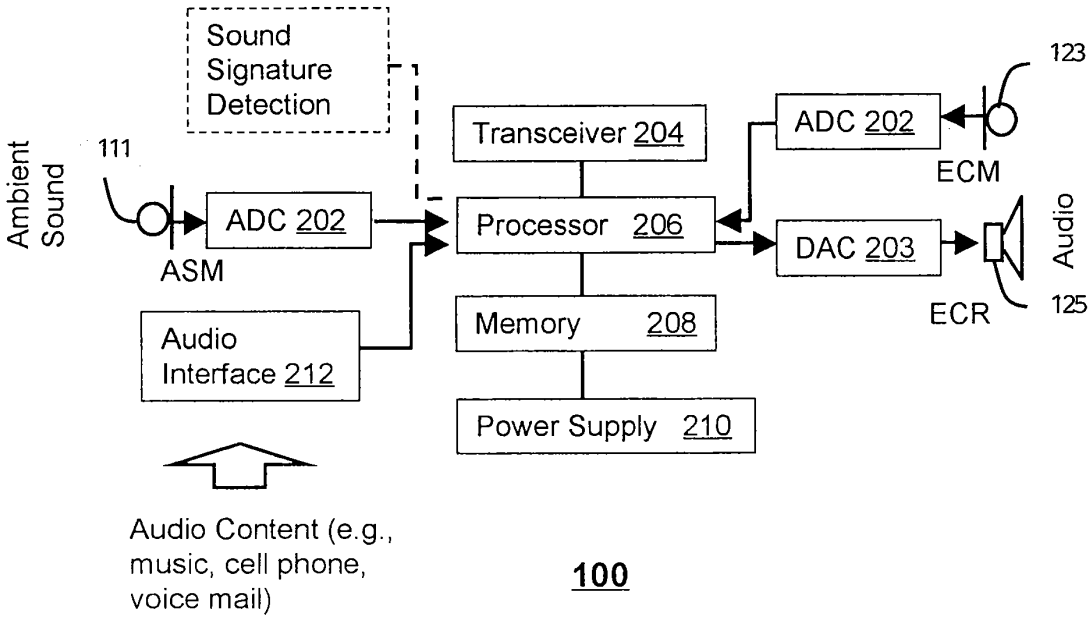


FIG. 2

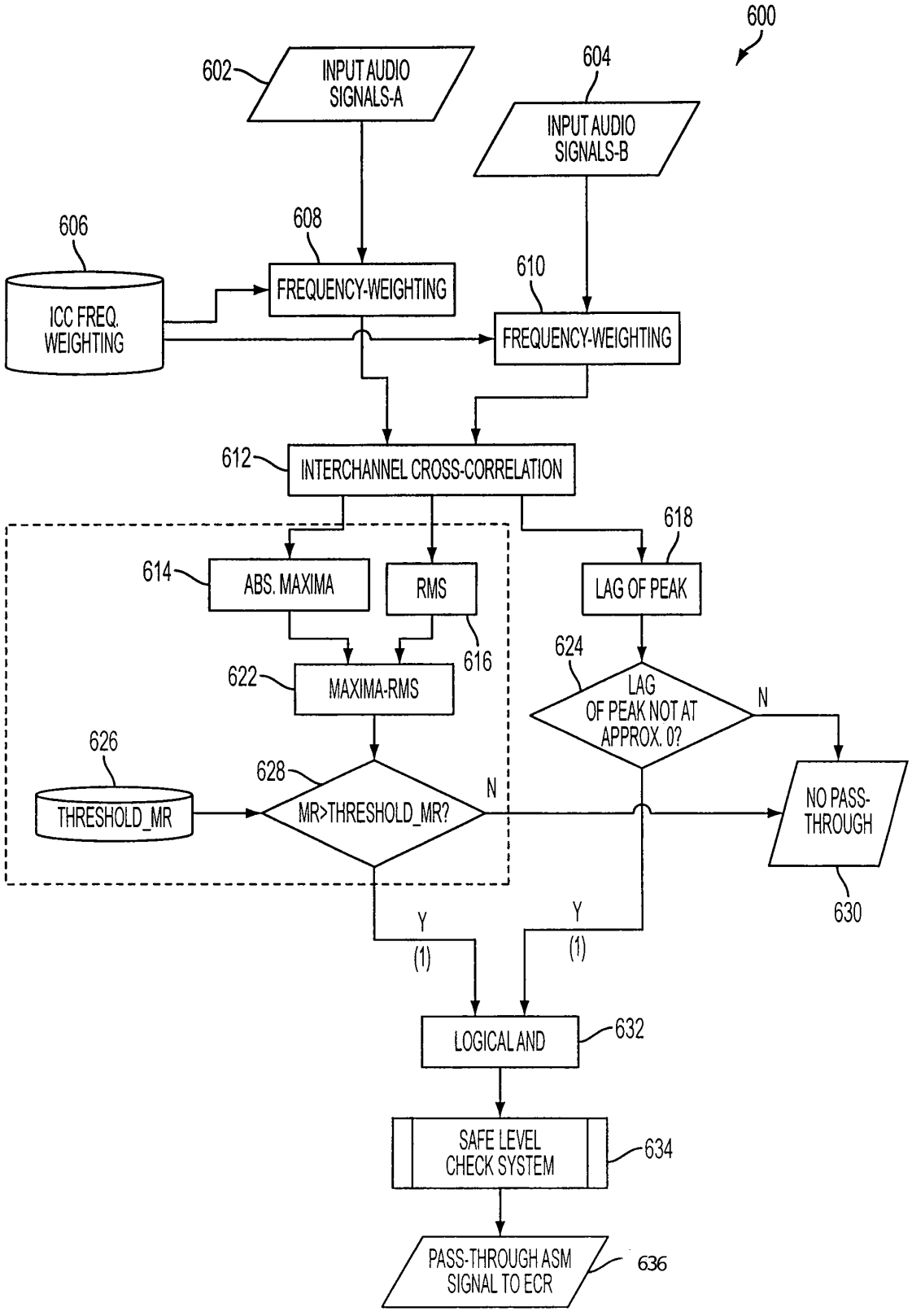

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH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

FIG. 6

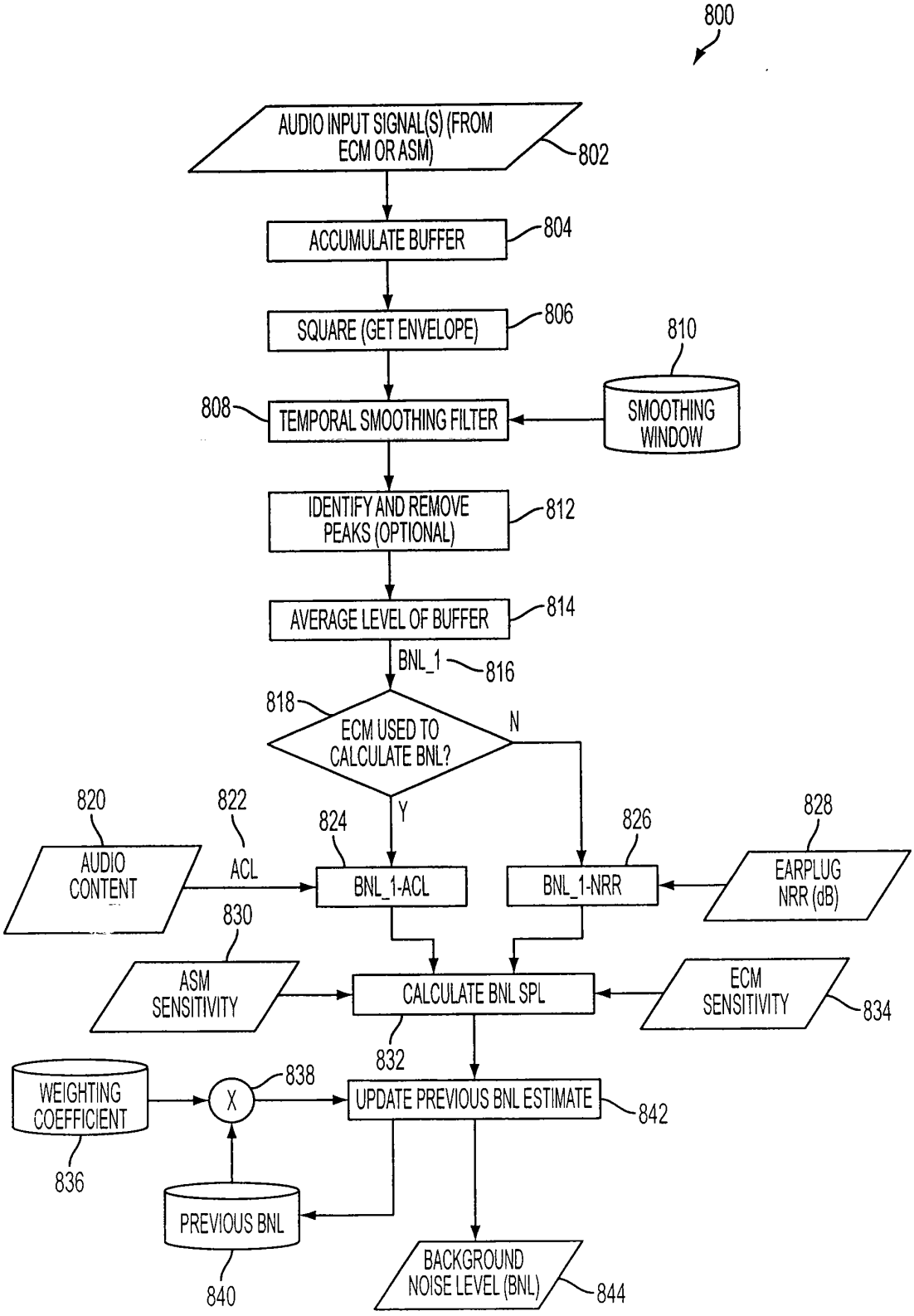

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT
822
ACL
824
BNL_1-ACL
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

Exhibit PAT-C

PTO/SB/22 (09-11)
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PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a)	Docket Number (Optional) PRS-118US
Application Number 12/017,878	Filed January 22, 2008
For METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION	
Art Unit 2818	Examiner Cuong B. Nguyen

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

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

	Fee	Small Entity Fee	
☑ One month (37 CFR 1.17(a)(1))	$150	$75	$ 75
☐ Two months (37 CFR 1.17(a)(2))	$560	$280	$
☐ Three months (37 CFR 1.17(a)(3))	$1270	$635	$
☐ Four months (37 CFR 1.17(a)(4))	$1980	$990	$
☐ Five months (37 CFR 1.17(a)(5))	$2690	$1345	$

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

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

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

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

☑ The Director is hereby authorized to charge any fees which may be required, or credit any overpayment, to Deposit Account Number 18-0350.

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

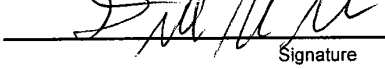
I am the

☐ applicant/inventor.

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

☑ attorney or agent of record. Registration Number 61,091

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



Signature

05/18/12

Date

Deborah M. Grove

Typed or printed name

610-407-0700

Telephone Number

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

☑ Total of 1 forms are submitted.

This collection of information is required by 37 CFR 1.136(a). The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 6 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	12812210
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/Edna Phelan
Filer Authorized By:	Deborah M. Grove
Attorney Docket Number:	PRS-118US (PERS-0014US)
Receipt Date:	18-MAY-2012
Filing Date:	22-JAN-2008
Time Stamp:	13:57:11
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$75
RAM confirmation Number	216
Deposit Account	180350
Authorized User	GROVE,DEBORAH

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 1.16 (National application filing, search, and examination fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)

Exhibit PAT-C

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Applicant Arguments/Remarks Made in an Amendment	amendment.pdf	511808 0776902170b6683004491c43e18554b242c97a9e	no	15
Warnings:					
Information:					
2	Specification	subspec.pdf	1008382 54466fb3167480c58628f53c62c5b1524dca0d8d	no	18
Warnings:					
Information:					
3	Specification	cleanspec.pdf	996369 23037e21d908d335332728c9cb8ab9e9734f180f	no	18
Warnings:					
Information:					
4	Drawings-only black and white line drawings	replacementdrawings.pdf	74065 4a0298ee54fd81aaa1b1f0a03f26797308653885	no	3
Warnings:					
Information:					
5	Extension of Time	extensionoftime.pdf	57108 dcca7894da6563c0367aeff38f0dc2b5f0b47a7a	no	1
Warnings:					
Information:					
6	Fee Worksheet (SB06)	fee-info.pdf	30185 fee47de41efce87bcbeabbdb3aa3ae36aa42a7ca	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	2677917		

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

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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 07/03/2012

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EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	
MAIL DATE	DELIVERY MODE
07/03/2012	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 *18 May 2012*.

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) *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 _____ 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

DETAILED ACTION

Response to Amendment

Applicant's amendment to the claims and the specification, filed on May 18th, 2012, is acknowledged. Entry of amendments are accepted and made of record.

Response to Arguments/Remarks

Applicant's response filed on May 18th, 2012 is acknowledged and is answered as follows.

Applicant's remarks, see pg. 10, with respect to the missing reference Anderson (US 5,721,783) in PTO-892 form has been fully considered. Therefore, a new PTO-892 form has been attached to this office action including the reference Anderson (US 5,721,783).

Applicant's argument, see pgs.10-11, with respect to the objections of claims 1 and 8 because of the phrase “the at least one ECR” has been fully considered and are persuasive. Therefore, the objections of claims 1 and 8 have been withdrawn.

Applicant's arguments, see pg. 11, with respect to the rejection of the claims 1, 8 and 13 under Double Patenting rejection have been fully considered and are persuasive. Since claims 1, 8 and 13 have been amended, the rejections of claims 1, 8 and 13 under Double Patenting have been withdrawn.

Applicant's arguments, see pgs. 11-12, with respect to the rejection of the claims 1-2, 4-5, 8-9 and 13-14 under 35 U.S.C. § 102(e) have been fully considered but they are not persuasive in view of the following reasons.

Applicant argues that Goldstein does not disclose or suggest detecting an on-set of an acute sound (having an abrupt high energy sound profile) when the ambient sound level is greater than a predetermined threshold; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1 (emphasis added). Instead, Goldstein monitors the target sounds for identifiable features, i.e., a sound signature, with pattern recognition approaches being applied to detect the target sounds. Thus, Goldstein does not include all of the features of claim 1. The examiner respectfully disagrees because 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]), the ambient sound containing an acute sound having an abrupt high energy sound profile (target sounds such as alarm, bell, emergency vehicle, siren and ambulance are abrupt high energy sound profile) (see [0028]); 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, the processor being configured to monitor a change in an ambient sound level of the ambient sound, detect an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold (ambient sounds that exceed a predetermined power level are elevated for target sounds) (see [0050]), upon detecting the on-set of the acute sound, reproduce the acute sound within the ear canal via the ECR (see Fig.

2 and [0028]). Therefore, the Examiner submits that Goldstein still discloses all the limitation in the amended claims 1, 8 and 13.

In view of the foregoing reasons, the examiner believes that all Applicant's arguments and remarks are addressed. The examiner has determined that the previous Office Action is still proper based on the above responses. Therefore, the rejections are maintained

Claim Rejections - 35 USC § 102

1. 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.

2. 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]), the ambient sound containing an acute sound having an abrupt high energy sound profile (target sounds such as alarm, bell, emergency vehicle, siren and ambulance are abrupt high energy sound profile) (see [0028]); 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, the processor being configured to monitor a change in an ambient sound level of the ambient sound, detect an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold (ambient sounds that exceed a predetermined power level are elevated for target sounds) (see [0050]), upon detecting the on-set of the acute sound, reproduce 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]), the ambient sound containing an acute sound having an abrupt high energy sound profile (target sounds such as alarm, bell, emergency vehicle, siren and ambulance are abrupt high energy sound profile) (see [0028]); 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, the processor being configured to monitor a change in an ambient sound level of the ambient sound when the ambient sound level is greater than a predetermined threshold (ambient sounds that exceed a predetermined power level are

elevated for target sounds) (see [0050]), adjust an audio content level (ACL) of the audio content delivered to the ear canal, and upon detecting the on-set of the acute sound, reproduce the acute sound 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) of an ambient sound external (by ASM 110) to an ear at least partially occluded by an earpiece (see Fig. 2 and [0025]), the ambient sound containing an acute sound having an abrupt high energy sound profile (target sounds such as alarm, bell, emergency vehicle, siren and ambulance are abrupt high energy sound profile) (see [0028]); monitoring a change in the xASL of the ambient sound (sound signatures for previously learned target sounds) (see [0028]); detecting an on-set of the acute sound when the ambient sound level is greater than a predetermined threshold (ambient sounds that exceed a predetermined power level are elevated for target sounds) (see [0050]); and upon detecting the on-set of the acute sound, reproducing the acute sound within an 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

3. 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.

4. 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 ear canal, the audio content being direct via an ear canal receiver (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]).

5. 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

Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

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

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-5,721,783 A	02-1998	Anderson, James C.	381/328
	B	US-			
	C	US-			
	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.

Exhibit PAT-C

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
T.C./A.U.:	2818
Examiner:	Cuong B. Nguyen
Confirmation No.:	7135
Docket No.:	PRS-118US

AMENDMENT UNDER 37 C.F.R. § 1.116

Expedited Procedure

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

Sir:

Responsive to the Final Office Action dated July 3, 2012, please amend the above-identified application as follows:

☐ **Amendments to the Specification** begin on page of this paper.

☒ **Amendments to the Claims** are reflected in the listing of claims which begins on page 2 of this paper.

☐ **Amendments to the Drawings** begin on page of this paper and include an attached replacement sheet(s).

☐ **Amendments to the Abstract** are on page of this paper. A clean version of the Abstract is on page of this paper.

☒ **Remarks/Arguments** begin on page 8 of this paper.

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, the processor being configured to:

monitor a change in an ambient sound level of the ambient sound,

detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period, and

upon detecting the on-set of the acute sound, reproduce the acute sound within the ear canal via the ECR.

2. (Previously Presented) 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. (Previously Presented) 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 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. (Previously Presented) The earpiece of claim 5, wherein the processor measures a voltage level of 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, the processor being configured to:

monitor a change in an ambient sound level of the ambient sound,

detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period,

adjust an audio content level (ACL) of the audio content delivered to the ear canal, and

upon detecting the on-set of the acute sound, 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 of the ambient sound;

detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period; and

upon detecting the on-set of the acute sound, reproducing the acute sound within 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. (Previously Presented) 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 an internal ambient sound level (iASL) measured within the ear canal.

16. (Previously Presented) The method of claim 15, wherein the ACL is determined by

measuring a voltage level of the audio content directed to the ear canal, the audio content being directed via an Ear Canal Receiver(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.

26. (New) The earpiece of claim 1, wherein the processor detects the on-set of the acute sound without identifying the acute sound.

27. (New) The earpiece of claim 1, wherein the ASM is located internal to the earpiece.

28. (New) The earpiece of claim 1, wherein the ASM is located external to the earpiece on a remote device.

29. (New) The earpiece of claim 28, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

30. (New) The earpiece of claim 29, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

31. (New) The earpiece of claim 29, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

32. (New) The earpiece of claim 8, wherein the processor detects the on-set of the acute sound without identifying the acute sound.

33. (New) The earpiece of claim 8, wherein the ASM is located internal to the earpiece.

34. (New) The earpiece of claim 8, wherein the ASM is located external to the earpiece on a remote device.

35. (New) The earpiece of claim 34, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

36. (New) The earpiece of claim 35, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

37. (New) The earpiece of claim 35, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

38. (New) The method of claim 13, wherein the step of detecting the on-set includes detecting the on-set of the acute sound without identifying the acute sound.

39. (New) The method of claim 13, wherein the xASL is measured by an ambient sound microphone located internal to the earpiece.

40. (New) The method of claim 13, wherein the xASL is measured by an ambient sound microphone located external to the earpiece on a remote device.

41. (New) The method of claim 40, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

42. (New) The method of claim 41, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

43. (New) The method of claim 41, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

Exhibit PAT-C

Remarks/Arguments:

Claims 1, 8 and 13 have been amended. Claims 26-43 have been added. No new matter is introduced herein. Of pending claims 1-43, claims 21-25 have been withdrawn.

Claim Amendments

Claim 1 has been amended to clarify that the on-set of the acute sound is detected when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period. Claims 8 and 13 have been amended similarly to claim 1. No new matter is introduced herein. Support for the amendment may be found, for example, at Paragraph [0047] of the original specification.

§§102,103 Rejections

Claims 1, 2, 4, 5, 8, 9, 13 and 14 have been rejected under 35 U.S.C. §102(e) as being anticipated by Goldstein et al. (US 2008/0240458, referred to herein as Goldstein). Claims 1-6 and 8-20 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Svean et al. (US 6,754,359, referred to herein as Svean) in view of Rothschild (US 2006/0083388). It is respectfully submitted, however, that these claims are patentable over the cited art for the reasons set forth below.

> Claim 1 includes features neither disclosed nor suggested by the cited art, namely:
>
> monitor a change in an ambient sound level of the ambient sound,
>
> detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period, and
>
> upon detecting the on-set of the acute sound, reproduce the acute sound within the ear canal via the ECR.

Goldstein discloses, in Fig. 3, a method for earpiece monitoring and target detection which detects target sounds (such as an alarm, a horn, a voice or a noise) within the environment based on sound signatures (steps 302 and 304). The sound signatures relate to identifiable features that characterize the sound, such as statistical properties or parametric properties of the target sound. Feature extraction techniques are applied to the ambient sound to generate the sound signatures. The processor can adjust sound delivered to the ear canal in view of a detected target sound (step 306). (Paragraphs [0034 – 0036].) At Paragraph [0050], Goldstein discloses analyzing potentially harmful audio artifacts such as high level

onsets and mitigating the audio artifacts to reduce timbral distortion effects. At Paragraph [0051], Goldstein teaches evaluating ambient sounds that exceed a predetermined power level for target sounds.

Goldstein, however, does not disclose or suggest: detecting an on-set of an acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1 (emphasis added). Instead, Goldstein relates to monitoring target sounds for identifiable features (i.e., a sound signature), with pattern recognition approaches being applied to detect the target sounds.

Goldstein only teaches detecting high level onsets of potentially harmful audio artifacts (Paragraph [0050]) and evaluating ambient sounds that exceed a predetermined power level for target sounds (Paragraph [0051].) Goldstein is silent regarding detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period, as required by claim 1.

Furthermore, Applicants note that Goldstein discloses, at Paragraph [0050], analyzing potentially harmful audio artifacts such as high level onsets and mitigating the potentially harmful artifacts. Accordingly, the skilled person would understand that Goldstein would not reproduce detected potentially harmful artifacts. Thus, Paragraph [0050] of Goldstein teaches away from Applicants' claimed feature of reproducing the acute sound upon detecting an on-set of the acute sound.

In summary, Goldstein does not disclose or suggest: detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period; and reproducing the acute sound within the ear canal upon detecting the on-set of the acute sound, as required by claim 1. Thus, Goldstein does not include all of the features of claim 1.

Applicants note that, although the Examiner indicated that claims 1-6 and 8-20 were rejected under 35 U.S.C. §103(a) as being unpatentable over Svean in view of Rothschild, the Examiner has not addressed where all of the features of these claims are disclosed in the cited art. Namely, the features of detecting an on-set of the acute sound and reproducing the acute sound upon detecting the on-set (in claims 1, 8 and 13) have not been addressed. Applicants respectfully point out that because the Examiner has failed to address all of the features cited in these indicated claims, the rejection of these claims is improper.

Applicants also submit, for the reasons set forth below, that neither Svean nor Rothschild, either alone or in combination, disclose or suggest detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period and reproducing the acute sound within the ear canal upon detecting the on-set of the acute sound, as required by claim 1.

Svean discloses, in Fig. 1, an earplug including sealing section 2, microphone M1 (for picking up external sound), microphone M2 (for picking the acoustic signal in meatus 3) and sound generator SG. (Col. 4, lines 29-34 and col. 5, lines 29-51.) Svean discloses, in Fig. 2, electric circuitry 11 of the earplug including processing unit E3. (Col. 6, lines 45-51.)

As acknowledged by the Examiner on page 8 of the Office Action, Svean does not disclose or suggest that the processor is configured to monitor a change in an ambient sound level for detecting an acute sound from the change and reproducing the acute sound within the ear canal via the ECR upon detecting the acute sound. Accordingly, Svean cannot disclose or suggest detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period and reproducing the acute sound within the ear canal upon detecting the on-set of the acute sound, as required by claim 1.

Rothschild discloses, in Fig. 1, system 100 for selectably switching between a plurality of audio channels. System 100 includes audio content device (ACD) 102, audio signal processing control unit 104, microphone input device (MID) 122 and audio output (AOD) device 106. Control unit 104 listens for predetermined sounds (such as from MID 122) and, if necessary, interrupts the audio content from ACD 102 to play the externally generated sound to the user. Control unit 104 includes switching mechanism 128 to select which channel of audio is played to the user. (Paragraphs [0019], [0023] and [0024].) Microprocessor 124 of control unit 104 monitors ACD 102 and MID 122 inputs for preselected sound patterns (such as human voice patterns, noise patterns or other sound patterns). The patterns can be recognized such as by speech recognition algorithms. (Paragraphs [0027] and [0028].) Control unit 104 can also lower the volume of the audio content from ACD 102, instead of completely switching between channels. (Paragraph [0032].)

Rothschild, however, does not disclose or suggest detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period; and reproducing the acute sound upon detecting the on-set of the

acute sound, as required by claim 1 (emphasis added). Instead, Rothschild monitors audio inputs for preselected sound patterns, where the sound patterns can be recognized by speech recognition algorithms. Thus, Rothschild does not include all of the features of claim 1 nor does Rothschild make up for the deficiencies of Goldstein or Svean with respect to claim 1.

In summary, none of the cited art disclose or suggest detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1. These features are absent from the cited art. Accordingly, allowance of claim 1 is respectfully requested.

Claims 8 and 13, although not identical to claim 1, include features similar to claim 1 which are neither disclosed nor suggested by the cited art. Accordingly, these claims are also patentable over the cited art for at least the same reasons as claim 1.

Claims 2-6, 9-12 and 14-20 include all of the features of respective claims 1, 8 and 13 from which they depend. Accordingly, claims 2-6, 9-12 and 14-20 are also patentable over the cited art for at least the same reasons as respective claims 1, 8 and 13.

Claim 7 has been rejected under 35 U.S.C. §103(a) as being unpatentable over Svean in view of Rothschild and further in view of Anderson (US 5,721,783). This claim, however, includes all of the features of claim 1 from which it depends and is patentable over Svean and Rothschild for at least the same reasons as claim 1. Anderson does not make up for the deficiencies of Svean and Rothschild with respect to claim 1. Accordingly, claim 7 which depends from claim 1, is also patentable over Svean and Rothschild combined with Anderson for at least the same reasons as claim 1.

New Claims

Claims 26-43 have been added. No new matter is introduced herein. Claims 26-43 include all of the features of respective claims 1, 8 and 13 from which they depend. Claims 26-43 include the additional features of: detecting the on-set of the acute sound without identifying the acute sound (claims 26, 32 and 38); the ambient sound microphone is located internal to the earpiece (claims 27, 33 and 39); the ambient sound microphone is located external to the earpiece on a remote device (claims 28, 34 and 40); the remote device includes at least one of a further earpiece, a cellphone, a media player, a portable computing device or a personal digital assistant (claims 29, 35, 41) the earpiece and the further earpiece are won by a same individual (claims 30, 36 and 42); and the earpiece and the further earpiece are

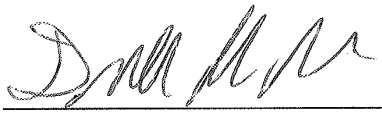
Exhibit PAT-C

configured to be worn by different individuals (claims 31, 37 and 43). No new matter is introduced herein. Support for claims 26-43 may be found, for example, at Paragraph [0047] (claims 26, 32 and 38); Paragraph [0035] and Fig. 1 (claims 27, 33 and 39); and Paragraphs [0037], [0040 – 0041], [0058] and [0060] and Figs. 2 and 5 (claims 28-31, 34-37 and 40-43). Claims 26-43 are patentable over the cited art for at least the same reasons as respective claims 1, 8 and 13.

Conclusion

In view of the amendments and arguments set forth above, the above-identified application is in condition for allowance which action is respectfully requested.

Respectfully submitted,



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Deborah M. Grove, Reg. No. 61,091
Patent Agent for Applicants

JHS/DMG/emp

Dated: October 3, 2012

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The Director is hereby authorized to charge or credit Deposit Account No. **18-0350** for any additional fees, or any underpayment or credit for overpayment in connection herewith.

1652292_1

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	13897704
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
Filer Authorized By:	
Attorney Docket Number:	PRS-118US (PERS-0014US)
Receipt Date:	03-OCT-2012
Filing Date:	22-JAN-2008
Time Stamp:	11:10:14
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$540
RAM confirmation Number	12311
Deposit Account	180350
Authorized User	GROVE, DEBORAH

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 1.16 (National application filing, search, and examination fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)

Exhibit PAT-C

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Amendment After Final	amendmentafterfinal.pdf	574934 f261fe15e05979f3631e45f8596f9c6023e7f288	no	12
Warnings:					
Information:					
2	Fee Worksheet (SB06)	fee-info.pdf	30157 bd2f086b64810129c091cfa380df151783d042a4	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	605091		

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

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

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

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 10/23/2012

RatnerPrestia
P.O. Box 980
Valley Forge, PA 19482

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	
MAIL DATE	DELIVERY MODE
10/23/2012	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

Advisory Action Before the Filing of an Appeal Brief	**Application No.** 12/017,878	**Applicant(s)** GOLDSTEIN ET AL.
	Examiner CUONG NGUYEN	**Art Unit** 2818

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

THE REPLY FILED 03 October 2012 FAILS TO PLACE THIS APPLICATION IN CONDITION FOR ALLOWANCE.

NO NOTICE OF APPEAL FILED

1. ☒ The reply was filed after a final rejection. No Notice of Appeal has been filed. To avoid abandonment of this application, applicant must timely file one of the following replies: (1) an amendment, affidavit, or other evidence, which places the application in condition for allowance;

(2) a Notice of Appeal (with appeal fee) in compliance with 37 CFR 41.31; or (3) a Request for Continued Examination (RCE) in compliance with 37 CFR 1.114 if this is a utility or plant application. Note that RCEs are not permitted in design applications. The reply must be filed within one of the following time periods:

a) ☒ The period for reply expires *3* months from the mailing date of the final rejection.

b) ☐ The period for reply expires on: (1) the mailing date of this Advisory Action; or (2) the date set forth in the final rejection, whichever is later. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of the final rejection.

c) ☐ A prior Advisory Action was mailed more than 3 months after the mailing date of the final rejection in response to a first after-final reply filed within 2 months of the mailing date of the final rejection. The current period for reply expires months from the mailing date of *the prior Advisory Action* or SIX MONTHS from the mailing date of the final rejection, whichever is earlier.

Examiner Note: If box 1 is checked, check either box (a), (b) or (c). ONLY CHECK BOX (b) WHEN THIS ADVISORY ACTION IS THE FIRST RESPONSE TO APPLICANT'S FIRST AFTER-FINAL REPLY WHICH WAS FILED WITHIN TWO MONTHS OF THE FINAL REJECTION. ONLY CHECK BOX (c) IN THE LIMITED SITUATION SET FORTH UNDER BOX (c). See MPEP 706.07(f).

Extensions of time may be obtained under 37 CFR 1.136(a). The date on which the petition under 37 CFR 1.136(a) and the appropriate extension fee have been filed is the date for purposes of determining the period of extension and the corresponding amount of the fee. The appropriate extension fee under 37 CFR 1.17(a) is calculated from: (1) the expiration date of the shortened statutory period for reply originally set in the final Office action; or (2) as set forth in (b) or (c) above, if checked. Any reply received by the Office later than three months after the mailing date of the final rejection, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

NOTICE OF APPEAL

2. ☐ The Notice of Appeal was filed on _____. A brief in compliance with 37 CFR 41.37 must be filed within two months of the date of filing the Notice of Appeal (37 CFR 41.37(a)), or any extension thereof (37 CFR 41.37(e)), to avoid dismissal of the appeal. Since a Notice of Appeal has been filed, any reply must be filed within the time period set forth in 37 CFR 41.37(a).

AMENDMENTS

3. ☒ The proposed amendments filed after a final rejection, but prior to the date of filing a brief, will not be entered because

a) ☒ They raise new issues that would require further consideration and/or search (see NOTE below);

b) ☐ They raise the issue of new matter (see NOTE below);

c) ☐ They are not deemed to place the application in better form for appeal by materially reducing or simplifying the issues for appeal; and/or

d) ☐ They present additional claims without canceling a corresponding number of finally rejected claims.

NOTE: *See Continuation Sheet*. (See 37 CFR 1.116 and 41.33(a)).

4. ☐ The amendments are not in compliance with 37 CFR 1.121. See attached Notice of Non-Compliant Amendment (PTOL-324).

5. ☐ Applicant's reply has overcome the following rejection(s): _____.

6. ☐ Newly proposed or amended claim(s) _____ would be allowable if submitted in a separate, timely filed amendment canceling the non-allowable claim(s).

7. ☒ For purposes of appeal, the proposed amendment(s): (a) ☒ will not be entered, or (b) ☐ will be entered, and an explanation of how the new or amended claims would be rejected is provided below or appended.

AFFIDAVIT OR OTHER EVIDENCE

8. ☐ The affidavit or other evidence filed after final action, but before or on the date of filing a Notice of Appeal will not be entered because applicant failed to provide a showing of good and sufficient reasons why the affidavit or other evidence is necessary and was not earlier presented. See 37 CFR 1.116(e).

9. ☐ The affidavit or other evidence filed after the date of filing the Notice of Appeal, but prior to the date of filing a brief, will not be entered because the affidavit or other evidence failed to overcome all rejections under appeal and/or appellant fails to provide a showing of good and sufficient reasons why it is necessary and was not earlier presented. See 37 CFR 41.33(d)(1).

10. ☐ The affidavit or other evidence is entered. An explanation of the status of the claims after entry is below or attached.

REQUEST FOR RECONSIDERATION/OTHER

11. ☐ The request for reconsideration has been considered but does NOT place the application in condition for allowance because: _____.

12. ☐ Note the attached Information *Disclosure Statement*(s). (PTO/SB/08) Paper No(s). _____

13. ☐ Other: _____.

STATUS OF CLAIMS

14. The status of the claim(s) is (or will be) as follows:

Claim(s) allowed: None.
Claim(s) objected to: None.
Claim(s) rejected: 1-20.
Claim(s) withdrawn from consideration: 21-25.

/STEVEN LOKE/ Supervisory Patent Examiner, Art Unit 2818	

Exhibit PAT-C

Continuation of 3. NOTE: The amended claims 1, 8, 13 and new claims 26-43 contain new limitation and would require further consideration and/or search.

Exhibit PAT-C

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
T.C./A.U.:	2818
Examiner:	Cuong B. Nguyen
Confirmation No.:	7135 DO NOT ENTER: /C.N./
Docket No.:	PRS-118US

AMENDMENT UNDER 37 C.F.R. § 1.116

Expedited Procedure

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

Sir:

Responsive to the Final Office Action dated July 3, 2012, please amend the above-identified application as follows:

☐ **Amendments to the Specification** begin on page of this paper.

☒ **Amendments to the Claims** are reflected in the listing of claims which begins on page 2 of this paper.

☐ **Amendments to the Drawings** begin on page of this paper and include an attached replacement sheet(s).

☐ **Amendments to the Abstract** are on page of this paper. A clean version of the Abstract is on page of this paper.

☒ **Remarks/Arguments** begin on page 8 of this paper.

Exhibit PAT-C

PTO/SB/30 (09-08)
Approved for use through 10/31/2006. OMB 0651-0031
U.S. Patent and Trademark Office: U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it display a valid OMB control number.

Request for Continued Examination (RCE) Transmittal

Address to:
Mail Stop RCE
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Application Number	12/017,878
Filing Date	Janaury 22, 2008
First Named Inventor	Steven W. Goldstein
Art Unit	2818
Examiner Name	Cuong B. Nguyen
Attorney Docket No.	PRS-118US

This is a Request for Continued Examination (RCE) under 37 CFR 1.114 of the above-identified application.
Request for Continued Examination (RCE) practice under 37 CFR 1.114 does not apply to any utility or plant application filed prior to June 8, 1995, or to any design application. See Instruction Sheet for RCEs (not to be submitted to the USPTO) on page 2.

1. Submission required under 37 CFR 1.114 Note: If the RCE is proper, any previously filed unentered amendments and amendments enclosed with the RCE will be entered in the order in which they were filed unless applicant instructs otherwise. If applicant does not wish to have any previously filed unentered amendment(s) entered, applicant **must** request non-entry of such amendment(s).
   - a. ☒ Previously submitted. If a final Office Action is outstanding, any amendments filed after the final Office Action may be considered as a submission even if this box is not checked.
     - i. ☐ Consider the arguments in the Appeal Brief or Reply Brief previously filed on _____.
     - ii. ☐ Other _____.
   - b. ☒ Enclosed
     - i. ☐ Amendment/Reply
     - ii. ☐ Affidavit(s)/Declaration(s)
     - iii. ☐ Information Disclosure Statement (IDS)
     - iv. ☒ Other Extension of Time.

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

3. Fees The RCE fee under 37 CFR 1.17(e) is required by 37 CFR 1.114 when the RCE is filed.
   - a. ☒ The Director is hereby authorized to charge any underpayment of fees, or credit any overpayments, to Deposit Account No. 18-0350. I have enclosed a duplicate copy of this sheet.
     - i. ☒ RCE fee required under 37 CFR 1.17(e).
     - ii. ☒ Extension of Time fee (37 CFR 1.136 and 1.17).
     - iii. ☐ Other _____.
   - b. ☐ Check in the amount of $ _____ is enclosed.
   - c. ☒ Payment by credit card (Form PTO-2038 enclosed)

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.

SIGNATURE OF APPLICANT, ATTORNEY, OR AGENT REQUIRED

Signature	Deborah M. Grove	Date	11/09/12
Name (Print/Type)	Deborah M. Grove	Registration No. (Attorney/Agent)	61,091

CERTIFICATE OF MAILING OR TRANSMISSION

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as First Class Mail in an envelope addressed to: Mail Stop RCE, Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450, or facsimile transmitted to the U.S. Patent and Trademark Office on the date shown below.

Signature			
Name (Print/Type)		Date	

This collection of information is required by 37 CFR 1.114. This information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Mail Stop RCE, Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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

Exhibit PAT-C

PTO/SB/22 (08-08)
Approved for use through 09/30/2008. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a) **FY 2008** *(Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818))*	Docket Number (Optional) PRS-118US
Application Number 12/017,878	Filed January 22, 2008
For METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION	
Art Unit 2818	Examiner Cuong B. Nguyen

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

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

		Fee	Small Entity Fee	
☐ One month	(37 CFR 1.17(a)(1))	$150	$75	$_____
☒ Two months	(37 CFR 1.17(a)(2))	$570	$285	$285
☐ Three months	(37 CFR 1.17(a)(3))	$1290	$645	$____
☐ Four months	(37 CFR 1.17(a)(4))	$2010	$1005	$_____
☐ Five months	(37 CFR 1.17(a)(5))	$2730	$1365	$_____

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

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

☒ Payment by credit card.

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

☒ The Director is hereby authorized to charge any additional fees which may be required, or credit any overpayment, to Deposit Account Number 18-0350.

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

I am the ☐ applicant/inventor.

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

☒ attorney or agent of record. Registration Number: 61,091.

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

Signature	Date
	11/09/12
Deborah M. Grove	***610-407-0700***
Typed or Printed Name	***Telephone Number***

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

☐ Total of _____ forms are submitted.

This collection of information is required by 37 CFR 1.136(a). The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 6 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	14189919
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/Edna Phelan
Filer Authorized By:	Deborah M. Grove
Attorney Docket Number:	PRS-118US (PERS-0014US)
Receipt Date:	09-NOV-2012
Filing Date:	22-JAN-2008
Time Stamp:	13:32:17
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$750
RAM confirmation Number	10642
Deposit Account	180350
Authorized User	GROVE, DEBORAH

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 1.16 (National application filing, search, and examination fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)

Exhibit PAT-C

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Request for Continued Examination (RCE)	RCE.pdf	92735 1976203726332d78f1a4fa6990526583ac88e7e5	no	1
Warnings:					
This is not a USPTO supplied RCE SB30 form.					
Information:					
2	Extension of Time	extensionfotime.pdf	70945 2b160951c23cadfa0bed1db697ad8483dbca133b	no	1
Warnings:					
Information:					
3	Fee Worksheet (SB06)	fee-info.pdf	32271 2c3f5ddee65a41eb8ac7bfefee56528cb80e2a49	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	195951		

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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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

117822 7590 01/29/2014

Personics Holdings Inc.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	
MAIL DATE	DELIVERY MODE
01/29/2014	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. 12/017,878	Applicant(s) GOLDSTEIN ET AL.	
	Examiner CUONG B. NGUYEN	Art Unit 2818	AIA (First Inventor to File) Status No

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

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTHS FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *10/03/2012*.

☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on _____.

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

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

4)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims*

5)☒ Claim(s) *1-43* 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 and 26-43* is/are rejected.

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

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

* If any claims have been determined allowable, you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see http://www.uspto.gov/patents/init_events/pph/index.jsp or send an inquiry to PPHfeedback@uspto.gov.

Application Papers

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

11)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

Priority under 35 U.S.C. § 119

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

Certified copies:

a)☐ All b)☐ Some** c)☐ None of the:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

** See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Information Disclosure Statement(s) (PTO/SB/08a and/or PTO/SB/08b) Paper No(s)/Mail Date _____.

3) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

4) ☐ Other: _____.

Exhibit PAT-C

DETAILED ACTION

Notice of Pre-AIA or AIA Status

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

Continued Examination Under 37 CFR 1.114

A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on October 3rd, 2012 has been entered.

Response to Amendment

Applicant's amendment to the claims, filed on October 3rd, 2012, is acknowledged. Entry of amendment is accepted and made of record.

Response to Arguments/Remarks

Applicant's response filed on October 3rd, 2012 is acknowledged and is answered as follows.

Applicant's arguments, see pgs. 8-11, with respect to the rejections of claims 1, 2, 4, 5,8-9 and 13-14 under 35 U.S.C 102(e) and claims 1-6 and 8-20 under 35 U.S.C 103(a) have been fully considered but are moot in view of the new ground(s) of rejection.

Claim Rejections - 35 USC § 112

1. The following is a quotation of the first paragraph of pre-AIA 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same, and shall set forth the best mode contemplated by the inventor of carrying out his invention.

2. Claims 1-7, 8-20 and 26-43 are rejected under 35 U.S.C. 112 (pre-AIA), first paragraph, as failing to comply with the written description requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to reasonably convey to one skilled in the relevant art that the inventor or a joint inventor, or for pre-AIA the inventor(s), at the time the application was filed, had possession of the claimed invention.

Claim 1 recites a limitation of “detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period” in lines 9-10 which is not being described in the original disclosure. According to [0047] of Applicant’s specification, the processor 206 looks for an abrupt change in the sound pressure level of the ambient sound across a small time period or the processor 206 can search for on-sets of the acute sound. Furthermore, [0055-0056] of Applicant's specification describes the processor 206 determines if the

sound pressure level of the ambient sound is greater than the predetermined threshold for the ambient sound signal is reproduced with the ECR 125. However, the Examiner is unable to find any description from Applicant's specification for the limitation of "detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period". Therefore, the limitation is considered as new matter.

Claims 2-7 and 26-31 are rejected for being depended on the rejected claim 1.

Claim 8 recites a limitation of "detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period" in lines 9-10 which is not being described in the original disclosure. According to [0047] of Applicant's specification, the processor 206 looks for an abrupt change in the sound pressure level of the ambient sound across a small time period or the processor 206 can search for on-sets of the acute sound. Furthermore, [0055-0056] of Applicant's specification describes the processor 206 determines if the sound pressure level of the ambient sound is greater than the predetermined threshold for the ambient sound signal is reproduced with the ECR 125. However, the Examiner is unable to find any description from Applicant's specification for the limitation of "detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period". Therefore, the limitation is considered as new matter.

Claims 9-12 and 32-37 are rejected for being depended on the rejected claim 8.

Claim 13 recites a limitation of "detecting an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period" in lines 7-8 which is not being described in the original disclosure. According to [0047] of Applicant's specification, the processor 206 looks for an abrupt change in the sound pressure level of the ambient sound across a small time period or the processor 206 can search for on-sets of the acute sound. Furthermore, [0055-0056] of Applicant's specification describes the processor 206 determines if the sound pressure level of the ambient sound is greater than the predetermined threshold for the ambient sound signal is reproduced with the ECR 125. However, the Examiner is unable to find any description from Applicant's specification for the limitation of "detect an on-set of the acute sound when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period". Therefore, the limitation is considered as new matter.

Claims 14-20 and 38-43 are rejected for being depended on the rejected claim 13.

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 8:30 AM-5:00 PM.

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.

/C. B. N./
Examiner, Art Unit 2818

/STEVEN LOKE/
Supervisory Patent Examiner, Art Unit 2818

Exhibit PAT-C

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
T.C./A.U.:	2818
Examiner:	Cuong B. Nguyen
Confirmation No.:	7135
Docket No.:	PRS-118US

RESPONSE TO NON-FINAL OFFICE ACTION

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

Sir:

Responsive to the Non-Final Office Action dated January 29, 2014, please amend the above-identified application as follows:

☐ **Amendments to the Specification** begin on page of this paper.

☒ **Amendments to the Claims** are reflected in the listing of claims which begins on page **2** of this paper.

☐ **Amendments to the Drawings** begin on page of this paper and include an attached replacement sheet(s).

☐ **Amendments to the Abstract** are on page of this paper. A clean version of the Abstract is on page of this paper.

☒ **Remarks/Arguments** begin on page 9 of this paper.

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, the processor being configured to:

monitor a change in an ambient sound level of the ambient sound,

detect an on-set of a fast rising amplitude wave-front of the acute sound ~~when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period~~, and

upon detecting the on-set of the acute sound, reproduce the acute sound within the ear canal via the ECR.

2. (Previously Presented) 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. (Previously Presented) 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 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. (Previously Presented) The earpiece of claim 5, wherein the processor measures a voltage level of 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, the processor being configured to:

monitor a change in an ambient sound level of the ambient sound,

detect an on-set of a fast rising amplitude wave-front of the acute sound ~~when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period~~,

adjust an audio content level (ACL) of the audio content delivered to the ear canal, and

upon detecting the on-set of the acute sound, 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 of the ambient sound;

detecting an on-set of a fast rising amplitude wave-front of the acute sound ~~when the change in the ambient sound level is greater than a predetermined threshold over a predetermined time period~~; and

upon detecting the on-set of the acute sound, reproducing the acute sound within 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. (Previously Presented) 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 an internal ambient sound level (iASL) measured within the ear canal.

16. (Previously Presented) The method of claim 15, wherein the ACL is determined by

measuring a voltage level of the audio content directed to the ear canal, the audio content being directed via an Ear Canal Receiver(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.

26. (Previously Presented) The earpiece of claim 1, wherein the processor detects the on-set of the acute sound without identifying the acute sound.

27. (Previously Presented) The earpiece of claim 1, wherein the ASM is located internal to the earpiece.

28. (Previously Presented) The earpiece of claim 1, wherein the ASM is located external to the earpiece on a remote device.

29. (Previously Presented) The earpiece of claim 28, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

30. (Previously Presented) The earpiece of claim 29, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

31. (Previously Presented) The earpiece of claim 29, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

32. (Previously Presented) The earpiece of claim 8, wherein the processor detects the on-set of the acute sound without identifying the acute sound.

33. (Previously Presented) The earpiece of claim 8, wherein the ASM is located internal to the earpiece.

34. (Previously Presented) The earpiece of claim 8, wherein the ASM is located external to the earpiece on a remote device.

35. (Previously Presented) The earpiece of claim 34, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

36. (Previously Presented) The earpiece of claim 35, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

37. (Previously Presented) The earpiece of claim 35, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

38. (Previously Presented) The method of claim 13, wherein the step of detecting the on-set ~~includes detecting the on-set of the acute sound without identifying the acute sound~~comprises detecting an abrupt magnitude change across frequency sub-bands.

39. (Previously Presented) The method of claim 13, wherein the xASL is measured by an ambient sound microphone located internal to the earpiece.

40. (Previously Presented) The method of claim 13, wherein the xASL is measured by an ambient sound microphone located external to the earpiece on a remote device.

41. (Previously Presented) The method of claim 40, wherein the remote device includes at least one of a further earpiece, a cell phone, a media player, a portable computing device or a personal digital assistant.

42. (Previously Presented) The method of claim 41, wherein the earpiece and the further earpiece are configured to be worn by a same individual.

43. (Previously Presented) The method of claim 41, wherein the earpiece and the further earpiece are configured to be worn by different individuals.

Exhibit PAT-C

Remarks/Arguments:

Claims 1, 8, 13, and 38 have been amended. Claims 26-43 were previously added. No new matter is introduced herein. Of pending claims 1-43, claims 21-25 have been previously withdrawn without prejudice or disclaimer.

Claim Amendments

Claims 1-7, 8-20, and 26-43 were rejected under 35 USC Section 112, first paragraph, as failing to comply with the written description requirement. Claim 1 has been amended to clarify that the processor is configured to detect the on-set of a fast rising amplitude wave-front of the acute sound. Claims 8 and 13 have been amended similarly to claim 1. The claim contains subject matter which is clearly described in the specification to reasonably convey to one skilled in the art that the inventor(s) had possession of the claimed invention. Claim 38 was amended to clarify that detecting the on-set can comprise detecting an abrupt magnitude change across frequency sub-bands. No new matter is introduced herein. Support for the amendment may be found, for example, at Paragraph [0047] of the original specification.

> [0047] 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 presence of acute sounds before identifying the type of sound source.

Accordingly, Claims 1-7, 8-20, and 26-43 overcome the rejection under 35 USC Section 112, first paragraph and therefore claims 1-7, 8-20, and 26-43 are allowable.

Further, it is respectfully submitted that these claims are patentable over the previously cited art for the reasons set forth below.

Claim 1 includes features neither disclosed nor suggested by the cited art, namely:

monitor a change in an ambient sound level of the ambient sound,

detect an on-set of a fast rising amplitude wave-front of the acute sound, and

Exhibit PAT-C

upon detecting the on-set of the acute sound, reproduce the acute sound within the ear canal via the ECR.

In summary, none of the cited art disclose or suggest detecting an on-set of a fast rising amplitude wave-front of the acute sound; and reproducing the acute sound upon detecting the on-set of the acute sound, as required by claim 1. These features are absent from the cited art. Accordingly, allowance of claim 1 is respectfully requested.

Claims 8 and 13, although not identical to claim 1, include features similar to claim 1 which are neither disclosed nor suggested by the cited art. Accordingly, these claims are also patentable over the cited art for at least the same reasons as claim 1.

Claims 2-6, 9-12 and 14-20 include all of the features of respective claims 1, 8 and 13 from which they depend. Accordingly, claims 2-6, 9-12 and 14-20 are also patentable over the cited art for at least the same reasons as respective claims 1, 8 and 13.

Conclusion

In view of the amendments and arguments set forth above, the above-identified application is in condition for allowance which action is respectfully requested.

Respectfully submitted,

/pablo meles/

_____

Pablo Meles, Reg. No. 33739
Attorney for Applicants
954-410-1342

The Director is hereby authorized to charge or credit Deposit Account No. 504284for any additional fees, or any underpayment or credit for overpayment in connection herewith.

Exhibit PAT-C

Electronic Acknowledgement Receipt

EFS ID:	19461447
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:	117822
Filer:	Pablo Meles
Filer Authorized By:	
Attorney Docket Number:	PRS-118US (PERS-0014US)
Receipt Date:	30-JUN-2014
Filing Date:	22-JAN-2008
Time Stamp:	22:40:55
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	Deposit Account
Payment was successfully received in RAM	$300
RAM confirmation Number	10476
Deposit Account	504284
Authorized User	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)

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1		PRS-118US_NFOA_30June2014.pdf	119537 fc2747bf67999552b5daf1446d9d90d6846074fe	yes	10

	Multipart Description/PDF files in .zip description		
	Document Description	**Start**	**End**
	Amendment/Req. Reconsideration-After Non-Final Reject	1	1
	Claims	2	8
	Applicant Arguments/Remarks Made in an Amendment	9	10

Warnings:

Information:

2	Fee Worksheet (SB06)	fee-info.pdf	30176 7f5144657a85d52958c394d0f9c39c991a061e98	no	2

Warnings:

Information:

Total Files Size (in bytes):	149713

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

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

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

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United States Patent and Trademark Office
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www.uspto.gov

NOTICE OF ALLOWANCE AND FEE(S) DUE

117822 7590 09/02/2014

Personics Holdings, LLC.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	

DATE MAILED: 09/02/2014

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

TITLE OF INVENTION: METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0	$0	$480	12/02/2014

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

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

HOW TO REPLY TO THIS NOTICE:

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

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

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

For purposes of this notice, small entity fees are 1/2 the amount of undiscounted fees, and micro entity fees are 1/2 the amount of small entity fees.

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

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

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

PTOL-85 (Rev. 02/11)

Exhibit PAT-C PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), to: Mail **Mail Stop ISSUE FEE**
Commissioner for Patents
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Alexandria, Virginia 22313-1450
or Fax (571)-273-2885

INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE and PUBLICATION FEE (if required). Blocks 1 through 5 should be completed where appropriate. All further correspondence including the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

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117822 7590 09/02/2014

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Note: A certificate of mailing can only be used for domestic mailings of the Fee(s) Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing or transmission.

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I hereby certify that this Fee(s) Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Mail Stop ISSUE FEE address above, or being facsimile transmitted to the USPTO (571) 273-2885, on the date indicated below.

(Depositor's name)
(Signature)
(Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
12/017,878	01/22/2008	Steven Wayne Goldstein	PRS-118US (PERS-0014US)	7135

TITLE OF INVENTION: METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0	$0	$480	12/02/2014

EXAMINER	ART UNIT	CLASS-SUBCLASS
NGUYEN, CUONG B	2818	381-312000

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363).

❑ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.

❑ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47; Rev 03-02 or more recent) attached. **Use of a Customer Number is required.**

2. For printing on the patent front page, list

(1) The names of up to 3 registered patent attorneys or agents OR, alternatively,

(2) The name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.

1 _____

2 _____

3 _____

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)

PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. If an assignee is identified below, the document has been filed for recordation as set forth in 37 CFR 3.11. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE

(B) RESIDENCE: (CITY and STATE OR COUNTRY)

Please check the appropriate assignee category or categories (will not be printed on the patent) : ❑ Individual ❑ Corporation or other private group entity ❑ Government

4a. The following fee(s) are submitted:

❑ Issue Fee

❑ Publication Fee (No small entity discount permitted)

❑ Advance Order - # of Copies _____

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❑ A check is enclosed.

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

❑ The Director is hereby authorized to charge the required fee(s), any deficiency, or credits any overpayment, to Deposit Account Number _____ (enclose an extra copy of this form).

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

❑ Applicant certifying micro entity status. See 37 CFR 1.29

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

❑ Applicant asserting small entity status. See 37 CFR 1.27

NOTE: If the application was previously under micro entity status, checking this box will be taken to be a notification of loss of entitlement to micro entity status.

❑ Applicant changing to regular undiscounted fee status.

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

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

Authorized Signature _____ Date _____

Typed or printed name _____ Registration No. _____

PTOL-85 Part B (10-13) Approved for use through 10/31/2013. OMB 0651-0033 U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

Exhibit PAT-C

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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

117822 7590 09/02/2014

Personics Holdings, LLC.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	

DATE MAILED: 09/02/2014

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

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

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

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

Exhibit PAT-C

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

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

The information collected by PTOL-85 Part B is required by 37 CFR 1.311. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, Virginia 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, Virginia 22313-1450. Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

Privacy Act Statement

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

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

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

Exhibit PAT-C

Notice of Allowability	Application No. 12/017,878	Applicant(s) GOLDSTEIN ET AL.	
	Examiner CUONG B. NGUYEN	Art Unit 2818	AIA (First Inventor to File) Status No

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

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

1. ☒ This communication is responsive to *amendment filed on 06/30/2014*.
   ☐ A declaration(s)/affidavit(s) under **37 CFR 1.130(b)** was/were filed on_____.
2. ☐ An election was made by the applicant in response to a restriction requirement set forth during the interview on _____; the restriction requirement and election have been incorporated into this action.
3. ☒ The allowed claim(s) is/are *1-20 and 26-43*. As a result of the allowed claim(s), you may be eligible to benefit from the **Patent Prosecution Highway** program at a participating intellectual property office for the corresponding application. For more information, please see http://www.uspto.gov/patents/init_events/pph/index.jsp or send an inquiry to PPHfeedback@uspto.gov .
4. ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

   Certified copies:

   a) ☐ All b) ☐ Some *c) ☐ None of the:
      1. ☐ Certified copies of the priority documents have been received.
      2. ☐ Certified copies of the priority documents have been received in Application No. _____ .
      3. ☐ Copies of the certified copies of the priority documents have been received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

   * Certified copies not received: _____.

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

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

   Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the drawings in the front (not the back) of each sheet. Replacement sheet(s) should be labeled as such in the header according to 37 CFR 1.121(d).
6. ☐ DEPOSIT OF and/or INFORMATION about the deposit of BIOLOGICAL MATERIAL must be submitted. Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Attachment(s)

1. ☐ Notice of References Cited (PTO-892)
2. ☐ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date _____
3. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
4. ☒ Interview Summary (PTO-413), Paper No./Mail Date *20140814* .
5. ☒ Examiner's Amendment/Comment
6. ☒ Examiner's Statement of Reasons for Allowance
7. ☐ Other _____.

/DAVID VU/
Primary Examiner, Art Unit 2818

Exhibit PAT-C

DETAILED ACTION

Response to Amendment

Applicant's amendment to the claims, filed on June 30th, 2014, is acknowledged. Entry of amendments are accepted and made of record.

Response to Arguments/Remarks

Applicant's response filed on June 30th, 2014 is acknowledged and is answered as follows.

Applicant's arguments, see pgs. 9-10, with respect to the rejections of the claims 1-7, 8-20 and 26-43 under 35 U.S.C. § 112, first paragraph have been fully considered and are persuasive. However, Applicants' attorney agreed to amend claims 1, 8 and 13 to resolve issues of minor informalities and cancel claims 21-25 of non-elected species as discussed during the telephone interview on August 14th, 2014 (see Examiner's Amendment).

EXAMINER'S AMENDMENT

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

Exhibit PAT-C

Authorization for this examiner's amendment was given in a telephone interview with Pablo Meles (Reg. No. 33,739) on August 14th, 2014.

The application has been amended as follows:

In the claim:

2. In claim 1, line 12, after "on-set of" insert "the fast rising amplitude wave-front of".

3. In claim 8, line 14, after "on-set of" insert "the fast rising amplitude wave-front of".

4. In claim 13, line 10, after "on-set of" insert "the fast rising amplitude wave-front of".

5. Cancel claims 21-25.

Allowable Subject Matter

6. Claims 1-20 and 26-43 are allowed over prior art of record.

7. The following is an examiner's statement of reason for allowance: the prior art made of record does not teach or fairly suggest the following: the earpiece comprising: the processor being configured to detect an on-set of a fast rising amplitude wave-front of the acute sound and upon detecting the on-set of the fast rising amplitude wave-front, reproduce the acute sound within the ear canal via ECR as recited in claims 1 and 8. Claims 2-7, 26-31 and 9-12, 32-36 depend on claims 1 and 9, and therefore also include said claimed limitation.

Furthermore, the prior art made of record does not teach or fairly suggest the following: the method comprising: detecting an on-set of a fast rising amplitude wave-front of the acute sound and upon detecting the on-set of the fast rising amplitude wave-front, reproducing the acute sound within the ear canal via ECR as recited in claim 13.

Claims 15-20 and 38-43 depend on claim 13, and therefore also include said claimed limitation.

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

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to 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.

/C. B. N./
Examiner, Art Unit 2818

/DAVID VU/
Primary Examiner, Art Unit 2818

Exhibit PAT-C

Examiner-Initiated Interview Summary	**Application No.** 12/017,878	**Applicant(s)** GOLDSTEIN ET AL.	
	Examiner CUONG B. NGUYEN	**Art Unit** 2818	

All participants (applicant, applicant's representative, PTO personnel):

(1) *CUONG B. NGUYEN*. (3)_____.

(2) *Pablo Meles (Reg. No. 33,739)*. (4)_____.

Date of Interview: *14 August 2014*.

Type: ☒ Telephonic ☐ Video Conference
☐ Personal [copy given to: ☐ applicant ☐ applicant's representative]

Exhibit shown or demonstration conducted: ☐ Yes ☒ No.
If Yes, brief description: _____.

Issues Discussed ☐101 ☐112 ☐102 ☐103 ☒Others
(For each of the checked box(es) above, please describe below the issue and detailed description of the discussion)

Claim(s) discussed: *1,8,13 and 21-25*.

Identification of prior art discussed: _____.

Substance of Interview
(For each issue discussed, provide a detailed description and indicate if agreement was reached. Some topics may include: identification or clarification of a reference or a portion thereof, claim interpretation, proposed amendments, arguments of any applied references etc...)

Examiner initiated the interview to propose amendments to claims 1, 8, 13 and cancel claims 21-25 of non-elected species in order to bring the application into condition for allowance. Applicant's Attorney authorized the Examiner to do the Examiner Amendment with the proprosed amendments to the claims (see Examiner Amendment).

Applicant recordation instructions: It is not necessary for applicant to provide a separate record of the substance of interview.

Examiner recordation instructions: Examiners must summarize the substance of any interview of record. A complete and proper recordation of the substance of an interview should include the items listed in MPEP 713.04 for complete and proper recordation including the identification of the general thrust of each argument or issue discussed, a general indication of any other pertinent matters discussed regarding patentability and the general results or outcome of the interview, to include an indication as to whether or not agreement was reached on the issues raised.

☐ Attachment

/DAVID VU/ Primary Examiner, Art Unit 2818	

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

NOTICE OF ALLOWANCE AND FEE(S) DUE

117822 7590 09/02/2014

Personics Holdings, LLC.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

EXAMINER	
NGUYEN, CUONG B	
ART UNIT	PAPER NUMBER
2818	

DATE MAILED: 09/02/2014

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

TITLE OF INVENTION: METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0	$0	$480	12/02/2014

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

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

HOW TO REPLY TO THIS NOTICE:

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

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

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

For purposes of this notice, small entity fees are 1/2 the amount of undiscounted fees, and micro entity fees are 1/2 the amount of small entity fees.

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

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

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

PTOL-85 (Rev. 02/11)

Exhibit PAT-C PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), to: Mail **Mail Stop ISSUE FEE**
Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313-1450
or Fax (571)-273-2885

INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE and PUBLICATION FEE (if required). Blocks 1 through 5 should be completed where appropriate. All further correspondence including the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

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

117822 7590 09/02/2014

Personics Holdings, LLC.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

Note: A certificate of mailing can only be used for domestic mailings of the Fee(s) Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing or transmission.

Certificate of Mailing or Transmission

I hereby certify that this Fee(s) Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Mail Stop ISSUE FEE address above, or being facsimile transmitted to the USPTO (571) 273-2885, on the date indicated below.

(Depositor's name)
(Signature)
(Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
12/017,878	01/22/2008	Steven Wayne Goldstein	PRS-118US (PERS-0014US)	7135

TITLE OF INVENTION: METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0	$0	$480	12/02/2014

EXAMINER	ART UNIT	CLASS-SUBCLASS
NGUYEN, CUONG B	2818	381-312000

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363).

☐ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.

☐ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47; Rev 03-02 or more recent) attached. **Use of a Customer Number is required.**

2. For printing on the patent front page, list

(1) The names of up to 3 registered patent attorneys or agents OR, alternatively,

(2) The name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.

1 Pablo Meles

2

3

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)

PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. If an assignee is identified below, the document has been filed for recordation as set forth in 37 CFR 3.11. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE: Personics Holdings, LLC.

(B) RESIDENCE: (CITY and STATE OR COUNTRY): Boca Raton, Florida

Please check the appropriate assignee category or categories (will not be printed on the patent) : ☐ Individual ☒ Corporation or other private group entity ☐ Government

4a. The following fee(s) are submitted:

☒ Issue Fee

☐ Publication Fee (No small entity discount permitted)

☐ Advance Order - # of Copies _____

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

☐ A check is enclosed.

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

☐ The Director is hereby authorized to charge the required fee(s), any deficiency, or credits any overpayment, to Deposit Account Number 504284 (enclose an extra copy of this form).

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

☐ Applicant certifying micro entity status. See 37 CFR 1.29

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

☐ Applicant asserting small entity status. See 37 CFR 1.27

NOTE: If the application was previously under micro entity status, checking this box will be taken to be a notification of loss of entitlement to micro entity status.

☐ Applicant changing to regular undiscounted fee status.

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

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

Authorized Signature /pablo meles/ Date _____

Typed or printed name Pablo Meles Registration No. 33739

PTOL-85 Part B (10-13) Approved for use through 10/31/2013. OMB 0651-0033 U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
12/017,878	12/23/2014	8917894	PRS-118US (PERS-0014US)	7135

117822 7590 12/03/2014

Personics Holdings, LLC.
4800 N Federal Hwy. Ste A205
Boca Raton, FL 33431

ISSUE NOTIFICATION

The projected patent number and issue date are specified above.

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

The Patent Term Adjustment is 1636 day(s). Any patent to issue from the above-identified application will include an indication of the adjustment on the front page.

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

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

Any questions regarding the Patent Term Extension or Adjustment determination should be directed to the Office of Patent Legal Administration at (571)-272-7702. Questions relating to issue and publication fee payments should be directed to the Application Assistance Unit (AAU) of the Office of Data Management (ODM) at (571)-272-4200.

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

Steven Wayne Goldstein, Delray Beach, FL;
Marc Andre Boillot, Plantation, FL;
John Usher, Montreal, CANADA;

The United States represents the largest, most dynamic marketplace in the world and is an unparalleled location for business investment, innovation, and commercialization of new technologies. The USA offers tremendous resources and advantages for those who invest and manufacture goods here. Through SelectUSA, our nation works to encourage and facilitate business investment. To learn more about why the USA is the best country in the world to develop technology, manufacture products, and grow your business, visit SelectUSA.gov.

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

REQUEST FOR *EX PARTE* REEXAMINATION OF U.S. PATENT NO. 11,244,666

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TABLE OF EXHIBITS

The exhibits in this Request are arranged in three groups: prior art ("PA"); relevant patent, patent family, and prosecution file histories ("PAT"); and other ("OTH").

A. Prior Art (PA)

- PA-1449 USPTO Form 1449
- 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")

B. Relevant Patent Materials (PAT)

- PAT-A U.S. Patent No. 11,244,666 ("the '666 patent")
- PAT-B File Wrapper for the '666 patent
- PAT-C Excerpts from the File Wrapper for U.S. Patent No. 8,917,894
- PAT-D Excerpts from the File Wrapper for U.S. Patent No. 10,134,377
- PAT-E Excerpts from the File Wrapper for U.S. Patent No. 10,535,334
- PAT-F Excerpts from the File Wrapper for U.S. Patent No. 10,810,989
- PAT-G U.S. Patent No. 8,917,894
- PAT-H U.S. Patent No. 10,134,377
- PAT-I U.S. Patent No. 10,535,334
- PAT-J U.S. Patent No. 10,810,989
- PAT-K File Wrapper for U.S. Provisional Application No. 60/885,917

C. Other Documents (OTH)

- OTH-A Declaration of Nathaniel Polish, Ph.D. ("Exp. Decl.")
- OTH-B *Curriculum Vitae* of Nathaniel Polish, Ph.D.
- OTH-C Petition for *Inter Partes* Review of U.S. Pat. No. 11,244,666 Challenging Claims 1-2 and 8-11, *Samsung Electronics Co., Ltd. v.*

Staton Techiya, LLC, IPR2022-01099, Paper No. 3 (PTAB June 9, 2022)

- OTH-D Patent Owner's Preliminary Response, *Samsung Electronics Co., Ltd. v. Staton Techiya, LLC*, IPR2022-01099, Paper No. 7 (PTAB Oct. 7, 2022)

- OTH-E Decision Regarding Institution, *Samsung Electronics Co., Ltd. v. Staton Techiya, LLC*, IPR2022-01099, Paper No. 10 (PTAB Jan. 3, 2023)

I. INTRODUCTION

Pursuant to 35 U.S.C. §§ 302-307 and 37 C.F.R. § 1.510, Samsung Electronics Co., Ltd. and Samsung Electronics America, Inc. ("Requesters") respectfully request reexamination of claims 1-11 of U.S. Patent No. 11,244,666 ("the '666 patent"), purportedly owned by Staton Techiya, LLC ("Patent Owner") through an assignment recorded with the USPTO at Reel/Frame 054756/0494. Reexamination is requested in view of the substantial new questions ("SNQ") of patentability presented below.

The prior art relied on in this Request discloses non-cumulative technological teachings that either were not previously presented or considered by the Office or were previously presented in different combinations than as presented herein, including in the petition for *inter partes* review ("IPR") that challenged the claims of the '666 patent. The prior art references and combinations relied upon in this Request raise SNQs that render claims 1-11 of the '666 patent invalid. Had the Office considered the new, non-cumulative teachings relied on in this Request, claims 1-11 ("the Challenged Claims") would not have issued.

Requesters reserve all rights and defenses available including, without limitation, defenses as to invalidity and non-infringement. By filing this Request, Requesters do not represent, agree, or concur that the '666 patent is otherwise valid and enforceable.

II. REQUESTING PARTIES, INTERESTED PARTIES, AND LITIGATION HISTORY OF THE '666 PATENT

A. The Requesting Parties

The Requesters of this *ex parte* reexamination ("Request") are as follows:

Samsung Electronics Co. Ltd., which is a corporation organized and existing under the laws of the Republic of Korea, with its headquarters at the following address:

129 Samsung-Ro
Yeongtong-Gu, Suwon
Gyeonggi-Do, Republic of Korea

Samsung Electronics, America, Inc., which is a New York corporation with a principal place of business at the following address:

85 Challenger Road
Ridgefield Park, New Jersey 07660

Communications regarding this application should be through Requesters' counsel:

Ali R. Sharifahmadian
Arnold & Porter Kaye Scholer LLP
601 Massachusetts Ave., NW
Washington, DC 20001-3743
Reg. No. 48,202

B. The Co-Pending Litigation Involving the '666 Patent Requires Treatment with Special Dispatch and Priority Over All Other Cases

The '666 patent is presently at issue in the following litigation:

- *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.).

Pursuant to 35 U.S.C. § 305, Requesters respectfully urge the U.S. Patent and Trademark Office ("USPTO" or "the Office") to grant this Request. Requesters also request that the reexamination be conducted with "special dispatch" and with "priority over all other cases" in accordance with MPEP § 2261 due to the ongoing nature of the underlying litigation.

III. REQUIREMENTS UNDER 37 C.F.R. § 1.510

As this Request satisfies each requirement of 37 C.F.R. § 1.510, Requesters request that the USPTO grant this reexamination a filing date pursuant to 37 C.F.R. § 1.510(d).

A. 37 C.F.R. § 1.510(a)—Fees

The Commissioner is hereby authorized to charge the fee of $12,600, as set forth in 37 C.F.R. § 1.20(c)(2), to the Arnold & Porter Kaye Scholer LLP ("Arnold & Porter") credit card on file with the USPTO referencing matter number 0027071.00133. 37 C.F.R. § 1.510(a). Any additional fees may also be charged to Arnold & Porter's Deposit Account No. 50-2387 referencing matter number 0027071.00133.

B. 37 C.F.R. § 1.510(b)—*Ex Parte* Requirements

Pursuant to 37 C.F.R. § 1.510(b), Requesters hereby provide statement of substantial new questions of patentability ("SNQ"), an identification of the claims for which reexamination is requested, copies of the prior art, a copy of the '666 patent, certification that Patent Owner has been served, and certification that Requesters are not estopped from filing this Request under 35 U.S.C. §§ 315(e)(1) or 325(e)(1). 37 C.F.R. § 1.510(b).

1. 37 C.F.R. § 1.510(b)(1)—Statement Pointing Out Each Substantial New Question of Patentability

The following table identifies the SNQs addressed in this Request.

SNQ #	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

In addition to the prior art references themselves, the SNQs are supported by a Declaration from Requesters' technical expert, Dr. Nathaniel Polish, and numerous supporting pieces of evidence. (*See generally* Exhibit OTH-A.)

a. SNQs 1-3 based on Nemirovski and Rast

SNQs 1-3 are principally based on the combination of Nemirovski and Rast. Nemirovski discloses an earpiece including the claimed ambient microphone, ear canal microphone, and speaker. Rast is exemplary of a prior art headset with an integrated DSP and ambient microphone providing an ambient sound hearthrough feature that can be automatically controlled in response to detected acoustic events.

As explained further below, although Nemirovski does not expressly disclose a hearthrough feature, a person having ordinary skill in the art would have been motivated to combine Nemirovski and Rast in a manner that renders claims 1-11 obvious. A POSITA would have been motivated to further combine the Nemirovski-Rast device with Dijkstra or Van Doorn, which additionally disclose the "microphone port" limitations of the independent claims.

Nemirovski and Van Doorn were not of record during prosecution of the '666 patent or any of its parent applications. Accordingly, neither the Office nor the applicant considered, raised, or discussed the obviousness of any of the Challenged Claims in view of Nemirovski in combination with Rast; Nemirovski in combination with Rast and Dijkstra; or Nemirovski in combination with Rast and Van Doorn.

Although Rast was considered and discussed during prosecution of one of the '666 patent's parents (U.S. Patent No. 10,134,377 (the "'377 patent")), applicant made multiple amendments to the '377 patent claims to overcome Rast that are not present in the claims of the '666 patent. For example, the claims of the '377 patent require certain specific techniques for detecting an "acute sound," estimating a proximity of the "acute sound," and detecting a time lag of the "acute sound." More particularly, the '377 patent claims require:

- "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."
- "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 the microphone for measuring the external ambient sound level and identifying a peak in the cross correlation."
- "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"

None of those limitations are present in the '666 patent. Moreover, applicant did not resubmit the Rast reference to the Office during prosecution of the '666 patent.

During prosecution of the '666 patent, the Office took the position that Dijkstra's corresponding U.S. Patent Application Publication discloses the "first microphone port" and "second microphone port" limitations. (PAT-B, Aug. 16, 2021 Non-Final Office Action.). Notably, the applicant did not contest or otherwise dispute the Office's characterization of Dijkstra (*Id.*, Oct. 25, 2021 Response to Office Action), and Dijkstra is being presented in SNQs 1-2 for

the same undisputed reason. Moreover, both Rast and Dijkstra are being presented here in a new light not previously considered by the Office or applicant.

b. SNQs 4-7 based on Victorian and Beasley

SNQs 4-7 are principally based on Victorian in combination with Beasley. Victorian discloses an earpiece including the claimed ambient microphone, ear canal microphone, processor, and speaker. Victorian also discloses such an earpiece with an ambient sound passthrough feature. Beasley is exemplary of a prior art headset with an integrated DSP and ambient microphone providing an ambient sound hearthrough feature that can be automatically controlled in response to detected acoustic events. Moreover, Beasley expressly discloses adjusting the mix of a noise-cancellation signal, an ambient sound signal, and audio content in response to a user's speech command.

As explained further below, a POSITA would have been motivated to combine Victorian and Beasley in a manner that renders claims 1, 2, and 8-11 obvious. A POSITA would have been motivated to further combine the Victorian-Beasley device with Dijkstra or Van Doorn, which disclose the "microphone port" limitations of the independent claims. And a POSITA would have been motivated to implement Andrea's adaptive feedback noise-cancellation technique in the Victorian-Beasley combination in a manner that further renders claim 10 obvious.

Beasley, Van Doorn and Andrea were not of record during prosecution of the '666 patent or any of its parent applications. Accordingly, neither the Office nor the applicant considered, raised, or discussed the obviousness of any of the Challenged Claims in view of Victorian in combination with Dijkstra and Beasley; Victorian in combination with Dijkstra, Beasley and Andrea; Victorian in combination with Van Doorn and Beasley; or Victorian in combination with Van Doorn, Beasley and Andrea.

Although Victorian's corresponding U.S. Patent Application Publication was considered and discussed during prosecution of the '377 patent, the applicant made multiple amendments to the '377 patent claims to overcome Victorian that are not present in the claims of the '666 patent. As detailed above in Section III.B.1.a, the claims of the '377 patent require certain specific techniques for detecting an "acute sound," estimating a proximity of the "acute sound," and detecting a time lag on the "acute sound"—none of which are in the '666 patent. Moreover, applicant did not resubmit the Victorian reference discussed during prosecution of the '377 patent to the Office during prosecution of the '666 patent. And, again, Dijkstra is being presented for its teachings of the "first microphone port" and "second microphone port" that applicant did not dispute during prosecution of the '666 patent. Moreover, both Victorian and Dijkstra are being presented here in a new light not previously considered by the Office or the applicant.

c. SNQs 8-11 based on Dijkstra and Boersma

SNQs 8-11 are principally based on the combination of Dijkstra and Boersma. Dijkstra discloses an earpiece including the claimed ambient microphone, ear canal microphone, digital signal processor, memory, and speaker. Dijkstra also expressly discloses such an earpiece with an ambient sound passthrough feature. Boersma is exemplary of a prior art headset including a processor that provides the user with the ability to control the relative mix of ambient sound and audio content for different environments, as well as automatically maintaining the ratio of the selected mix in response to changes in the volume of the audio content. As discussed above, Beasley is exemplary of a prior art headset with an integrated DSP and ambient microphone providing an ambient sound hearthrough feature that can automatically adjust the mix of ambient sound and audio content in response to user speech commands.

Beasley and Van Doorn were not of record during prosecution of the '666 patent or any of its parent applications. Accordingly, neither the Office nor the applicant considered, raised, or discussed the obviousness of the Challenged Claims in view of Dijkstra in combination with Boersma and Beasley (SNQ 9); Dijkstra in combination with Boersma and Van Doorn (SNQ 10); or Dijkstra in combination with Boersma, Beasley and Van Doorn (SNQ 11).

SNQ 8 presents a combination of Dijkstra and Boersma, both of which were before the Office during prosecution of the '666 patent. To decide whether a reference that was previously considered by the PTO creates a SNQ, the Office "should evaluate the context in which the reference was previously considered and the scope of the prior consideration and determine whether he reference is now being considered for a substantially different purpose." (*In re Swanson*, 540 F.3d 1368, 1380 (Fed. Cir. 2008); MPEP 2216 ("The substantial new question of patentability may be based on art previously considered by the Office if the reference is presented in a new light or a different way that escaped review during earlier examination."); MPEP 2258.01 ("[A] SNQ may be based solely on [art previously considered by the Office] where the [previously considered] art is being presented/viewed in a new light, or in a different way, as compared with its use in the earlier concluded examination(s), in view of a material new argument or interpretation presented in the request.").)

Neither the Office nor the applicant expressly considered, raised, or discussed the obviousness of claims 1 and 11 in view of Dijkstra in combination with Boersma. Previously the Office applied Dijkstra only with respect to its teachings regarding the first and second microphone ports. Here, on the other hand, Requesters contend that Dijkstra discloses or suggests the entirety of each of the "first microphone configured to …" and "second microphone configured to …"

limitations, as well as the "speaker," "memory," "processor," "receiving a first microphone signal," and "receiving a second microphone signal" limitations.

Requesters also note that Boersma's disclosure is substantially similar to U.S. Patent Publication 2008/0165988 ("Terlizzi"), which the Office used as a primary reference to reject claim 1 during prosecution of the '666 patent. (PAT-B, June 1, 2021 Non-Final Office Action; Aug. 16, 2021 Final Office Action.) For example, both Terlizzi and Boersma disclose a device that allows the user to adjust the volume of received external sounds relative to generated audio signals. *Compare* Terlizzi, Abstract with Boersma, Abstract.

To traverse the Terlizzi-based rejections, applicant first amended the claims to add the "microphone port" limitations to distinguish their claims over Terlizzi, and when the Office maintained the rejection by adding Dijkstra to the combination to meet the "microphone port" limitations, applicant overcame the Terlizzi-based grounds by submitting a declaration under 37 C.F.R. § 1.131 to antedate Terlizzi. (PAT-B, Oct. 25, 2021 Response to Office Action.) The declaration averred a date of invention "at a date prior to 5 Jan 2007" and included as evidence U.S. Patent Application Publication No. 2016/0249128 ("Goldstein"), with an alleged priority date of November 18, 2006. (PAT-B, Declaration of Prior Invention in a WTO Member Country to Overcome a Cited Patent Pursuant to 37 C.F.R. § 1.131.) Boersma was filed March 8, 2006, about 10 months earlier than Terlizzi's filing date that the § 131 declaration swore behind, and about 8 months earlier than the earliest alleged priority date of the Goldstein reference.

Additionally, the specific challenges relied upon in this Request were not raised in the '666 IPR. (*See* Section IV.C.) Specifically, none of Nemirovski, Rast, Dijkstra, Van Doorn, Beasley, Andrea, or Boersma were part of the grounds presented in the IPR petition directed to the '666 patent. Although the IPR petition challenged the claims of the '666 patent on the basis of Victorian

in combination with Inanaga, and Victorian in combination with Inanaga and Kvaløy, none of the claims were challenged on the basis of Victorian in combination with Dijkstra and Beasley; Victorian in combination with Dijkstra, Beasley and Andrea; Victorian in combination with Van Doorn and Beasley; or Victorian in combination with Van Doorn, Beasley and Andrea. Requesters thus present arguments that were not presented during the '666 IPR and, accordingly were not considered by the PTAB or decided on the merits by the PTAB. Moreover, Requesters expressly present arguments here demonstrating that the "first microphone port" and "second microphone port" limitations were known in the art (e.g., based on Dijkstra or Van Doorn) and the claims were obvious in view of the SNQs presented herein.

2. 37 C.F.R. § 1.510(b)(2)—Identification of Every Claim for Which Reexamination is Requested

Requesters request reexamination of claims 1-11 of the '666 patent. 37 C.F.R. § 1.510(b)(2).

3. 37 C.F.R. 1.510(b)(2)—Detailed Explanation of the Pertinency and Manner of Applying the Prior Art

A detailed explanation of the pertinency and manner of applying the prior art—Nemirovski, Rast, Dijkstra, Van Doorn, Victorian, Beasley, Andrea, and Boersma—to every claim for which reexamination is requested in accordance with 37 C.F.R. § 1.510(b)(2) is provided below in Section VII.

4. 37 C.F.R. § 1.510(b)(3)—Copy of Every Patent or Printed Publication Relied upon or Referred to

Requesters provide a copy of every patent or printed publication relied upon or referred to in this Request as exhibits PA-A through PA-W. For convenience, the cited patents and printed publications are listed on a Form PTO-1449, which is submitted as Exhibit PA-1449.

5. 37 C.F.R. § 1.510(b)(4)—Copy of the Entire Patent

A copy of the entire '666 patent—including the front face, drawings, and specification/claims, in double column format—is provided herewith as Exhibit PAT-A in accordance with 37 C.F.R. § 1.510(b)(4). The Requesters are not aware of any prior reexamination certificates or certificates of correction for the '666 patent.

6. 37 C.F.R. § 1.510(b)(5)—Certification of Service

In accordance with 37 C.F.R. § 1.510(b)(5), Requesters provide below in Section IX a certification that a copy of this Request has been served in its entirety on Patent Owner's counsel of record at the address provided for in 37 C.F.R. § 1.33(c).

7. 37 C.F.R. § 1.510(b)(6)—Certification by the Third-Party Requester

In accordance with 37 C.F.R. § 1.510(b)(6), Requesters certify that the statutory estoppel provisions of 35 U.S.C. §§ 315(e)(1) and 325(e)(1) do not prohibit Requesters from filing this Request.

IV. THE '666 PATENT

A. Overview of the '666 Patent

The '666 patent is titled "METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION." (PAT-A, Cover.) The '666 patent issued on February 8, 2022, from U.S. Patent Application No. 16/987,396. (*Id.*) The '666 patent lists on its face Steven Wayne Goldstein, John Usher, and Marc Andre Boillot as the named inventors. (*Id.*) As mentioned above, the '666 patent claims priority through a series of applications to U.S. Provisional Application No. 60/885,917, filed on January 22, 2007. (*Id.*) For purposes of this Request only, Requesters assume an effective filing date of January 22, 2007.

The '666 patent is generally directed to devices and methods for detecting and reproducing acute sounds. (*Id.*, Title, Abstract, 1:23-28.) The '666 patent explains in the Background section that "[e]nvironmental noise is constantly presented in industrialized societies given the ubiquity of external sound intrusions." (*Id.*, 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.

(*Id.*, 1:55-2:5.) The '666 patent contends that "[a] need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties." (*Id.*, 2:19-21.)

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.

(*Id.*, 1:23-28.) The '666 patent claims recite an embodiment of this earpiece. The earpiece includes 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 monitors a change in the ambient sound level to detect an acute

sound from the change. The acute sound is reproduced within the ear canal via the ECR responsive to detecting the acute sound. (*Id.*, 2:28-36.) In some embodiments, an internal microphone to detect sounds within the ear canal, (i.e., an ear canal microphone (ECM)) is also used, for example, to detect the voice of the earpiece user or for other functions. (*Id.*, 2:49-51, 10:8-40.)

B. Prosecution History of the '666 Patent[1]

The applicant filed U.S. Patent Application No. 16/987,396 ("'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.)

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.)

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.

[1] The examination of the '666 patent and its parent applications was conducted under the pre-AIA version of Title 35 of the United States Code. Accordingly, all references in this section are to the pre-AIA version of Title 35.

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.)

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.)

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).

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.

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.

C. IPR Proceedings

On June 9, 2022, Requesters filed a petition for *inter partes* review of claims 1-2 and 8-11 of the '666 patent. (IPR2022-01099, Paper 1 ("Petition") (OTH-C).) The Petition raised seven obviousness grounds, organized into three groups based on three different primary references. Grounds 1A-1C relied on U.S. Paten Publication 2010/0119077 ("Platz") as the primary reference, Grounds 2A-2B relied on European Patent Publication EP1519625A2 ("Victorian") as the primary reference, and Grounds 3A-3B relied on U.S. Patent 8,477,955 ("Inanaga") as the primary reference. (*Id.*, 2).

On October 7, 2022, the Patent Owner filed a preliminary response. (IPR2022-01099, Paper 7 (OTH-D).) Notably, Patent Owner proposed that "all claim terms be afforded their plain and ordinary meaning," and did not otherwise elaborate on the meaning of the term "microphone port." (*Id.*, 11-12.) Nor did Patent Owner argue that any of the grounds advanced by the Petition failed to disclose a "first microphone port" or a "second microphone port." (*See generally id.*)[2]

On January 3, 2023, the Board denied institution. (IPR2022-01099, Paper 10 (OTH-E).) The Board found that none of Platz, Victorian or Inanaga, or any of the other references relied on in the Grounds disclosed a "first microphone port that is configured to face away from a user when the earphone is inserted" and a "second microphone port that is configured to face toward the user when the earphone is inserted." (OTH-E, 16-19, 24-25, 29-30.) The Board noted that the Petition did "not provide a construction or otherwise explain what either 'first microphone port' or the

[2] To be sure, Patent Owner did argue that Inanaga (Ground 3A) did not disclose that the second microphone port was oriented towards the user. (OTH-D, 16-19.) But there Patent Owner was taking issue with the orientation of the second microphone port, and not contesting that Inanaga failed to disclose or suggest a second microphone port.

'second microphone port' encompasses," and that the "record does not help [the Board] ascertain the plain and ordinary mean of 'microphone port.'" (*Id.*, 18.)

This Request demonstrates that a POSITA would have understood the term "microphone port" or "port" to refer to one or more related structures. As discussed in more detail in Section VII.A.2.b, *infra*, a POSITA would have understood the term "port"—and "microphone port" specifically—to refer to refer either to (1) an opening in the housing of a device in which the microphone is mounted, or (2) an opening in a microphone package itself. The Request also demonstrates that microphone ports were well-known in the art by January 22, 2007 (i.e., the'666 patent's earlies possible effective filing date). As explained in more detail herein, all limitations of the Challenged Claims, including the "first microphone port" and "second microphone port" limitations, would have been obvious to a POSITA based on the combination of references presented herein.

D. Level of Ordinary Skill in the Art

As explained by Requesters' expert, Dr. Nathaniel Polish, after considering (1) the type of problems encountered in the art, (2) prior art solutions to those problems, (3) the rapidity with which innovations are made, (4) sophistication of the technology, and (5) the educational level of active workers in the field, a person of ordinary skill in the art ("POSITA") as of the date of the purported invention of the claims of the '666 patent would have been a person possessing a bachelor's degree in electrical engineering, computer science, or a similar field and two years of experience in the design of digital audio systems and associated signal processing. (OTH-A, Exp. Decl., ¶¶ 21-23.)

E. Claim Construction

The '666 patent is not expired. Accordingly, the broadest reasonable interpretation standard should be applied in this reexamination. (*In re Swanson*, 540 F.3d 1368, 1377-78 (Fed. Cir. 2008).) Under this standard, "words of the claim must be given their plain meaning, unless such meaning is inconsistent with the specification." (MPEP § 2111.01.) "The broadest reasonable interpretation of a claim term may be the same as or broader than the construction of a term under the *Phillips* standard." (*Facebook, Inc. v. Pragmatus AV, LLC*, 582 F. App'x 864, 869 (Fed. Cir. 2014).)

V. DIRECTOR'S DISCRETION UNDER 35 U.S.C. § 325(d)

Requesters submit that this Request should be granted and reexamination of the '666 patent ordered in view of the discretionary factors set forth in 35 U.S.C. § 325(d). Specifically, Requesters submit herewith non-cumulative prior art that was either not previously presented or considered by the Office or was previously presented in different combinations than as presented herein, and, furthermore, present arguments regarding the patentability of claims 1-11 of the '666 patent which were not previously presented to the Office during prosecution of the '666 patent or any of its parent applications, nor during the '666 IPR.

Requesters submit that the substantial questions of patentability raised in this Request were not previously considered by the Office, are not currently being considered by the Office, and have not been decided on the merits by the Office or a district court, and are thus "new" under 35 U.S.C. § 325(d). (*In re Vivint, Inc.*, 14 F.4th 1342, 1349 (Fed. Cir. 2021).) In view of the substantial new questions of patentability raised herein, Requesters respectfully request that the Director exercise her discretion under § 325(d) to grant this Request and order Reexamination of the '666 patent.

Requesters submit that discretionarily denying this Request is not warranted. For instance, the present Request stands in stark contrast to the request at issue in *Vivint*. In *Vivint*, the Federal

Circuit held that ordering an *ex parte* reexamination based on a request that was nearly identical to a previously denied *inter partes* review petition, where that IPR petition was discretionarily denied due to a pattern of serially challenging the same patent, was an abuse of discretion under § 325(d). (*Vivint*, 14 F.4th at 1354.) Here, by contrast, Requesters have not engaged in any similar serial challenges to the '666 patent.

Furthermore, unlike in *Vivint*, ordering reexamination here would not be arbitrary, capricious, or an abuse of discretion. *Vivint* involved a non-merits-based discretionary denial of an IPR petition followed by the grant of a substantially similar *ex parte* reexamination request. *Id.* In this case, the '666 IPR's non-institution decision was based, not on an exercise of the Board's discretion, but, rather, on the Board's finding that the Petition, as filed, did not demonstrate a reasonable likelihood of prevailing on at least one challenged claim. (OTH-E.) Because the present Request and the Petition in the '666 IPR present different prior art, different combinations of prior art, and different arguments regarding the patentability of the claims of the '666 patent, the abuse of discretion issues raised by *Vivint* are absent here.

Therefore, for at least the foregoing reasons, Requesters respectfully request Reexamination of the '666 patent ordered in view of the Director's discretion under 35 U.S.C. § 325(d).

VI. THE PRIOR ART[3]

SNQs for claims 1-11 of the '666 patent exist based upon Nemirovski, Rast, Dijkstra, Van Doorn, Victorian, Beasley, Andrea, and Boersma.

[3] The examination of the '666 patent and its parent applications was conducted under the pre-AIA version of Title 35 of the United States Code. Accordingly, all references in this section are to the pre-AIA version of Title 35.

As discussed above, the '666 patent claims priority to U.S. Provisional Application No. 60/885,917, filed January 22, 2007. The prior art references cited here are prior art under pre-AIA 35 U.S.C. §§ 102(a), (b) and/or (e). A brief summary of the prior art is provided below.

A complete listing of the prior art and references discussed in this Request is provided in the attached Form PTO-1449, together with a complete copy of each listed document.

Sections VII.A-K discuss in detail how each prior art reference, or combination of references, presents new, non-cumulative technological teachings that were not previously considered and discussed on the record during the prosecution of the '666 patent. MPEP § 2216.

A. U.S. Patent 6,647,368 ("Nemirovski") (Ex. PA-A)

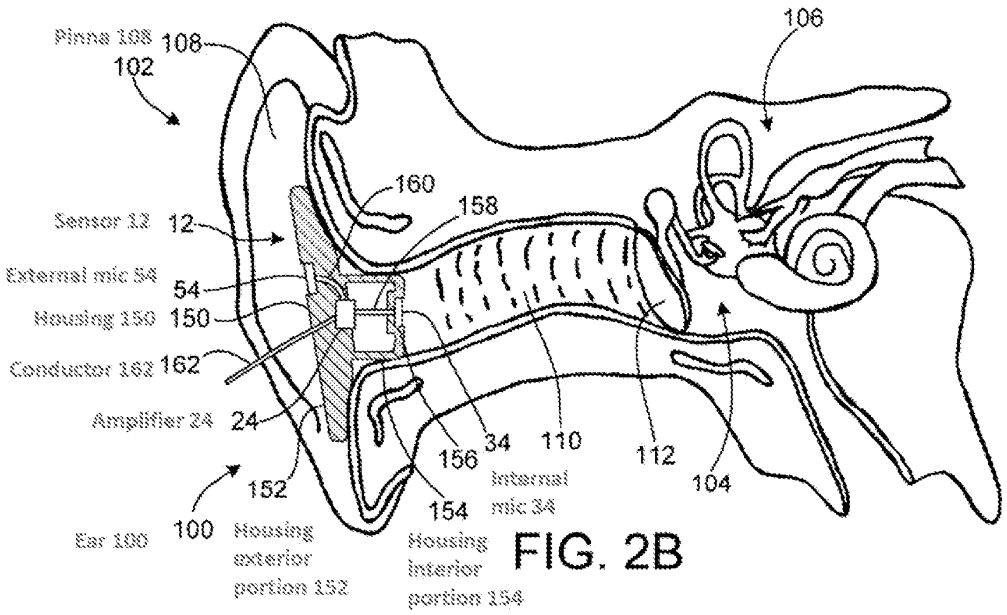
Nemirovski issued on November 11, 2003, and is entitled "SENSOR PAIR FOR DETECTING CHANGES WITHIN A HUMAN EAR AND PRODUCING A SIGNAL CORRESPONDING TO THOUGHT, MOVEMENT, BIOLOGICAL FUNCTION AND/OR SPEECH." (PA-A, Cover.) Accordingly, Nemirovski is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

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.)

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.*)

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):



Nemirovski, FIG. 2B (annotated)[4]

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.)

B. U.S. Patent Publication 2001/0046304A1 ("Rast") (Ex. PA-B)

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." (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).

[4] All bold/italics/color emphases and annotations are added unless noted otherwise.

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].)

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. European Patent Application Publication EP1640972A1 ("Dijkstra") (Ex. PA-C)

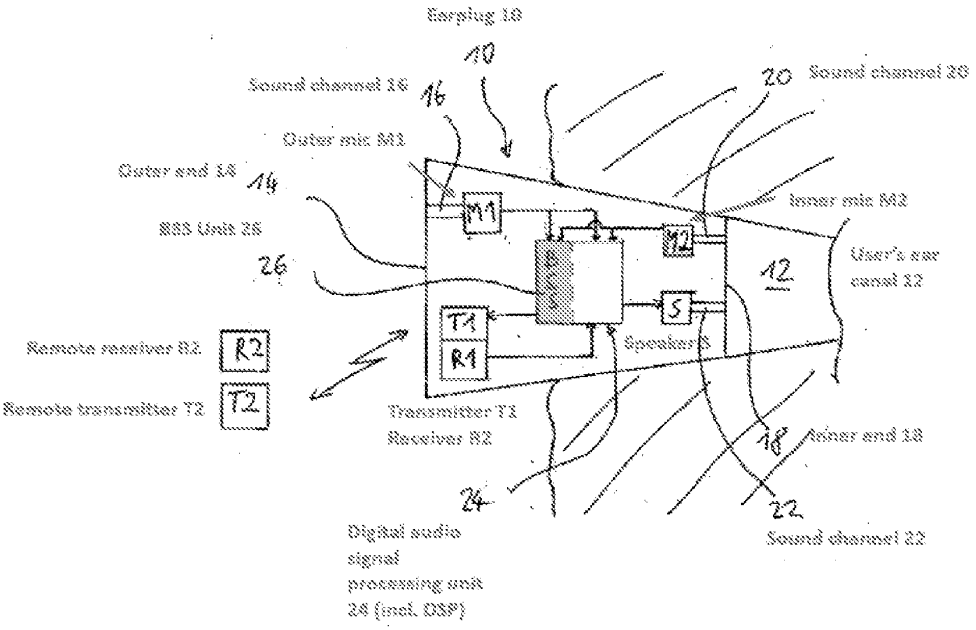
Dijkstra is a publication of European Patent Application No. 05028366.2, entitled "SYSTEM AND METHOD FOR SEPARATION OF A USERS VOICE FROM AMBIENT SOUND." (PA-C, Cover.) Dijkstra was published on March 29, 2006. (*Id.*) Accordingly, Dijkstra is prior art to the '666 patent under at least 35 U.S.C. § 102(a).

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].)

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].)

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)

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.*,

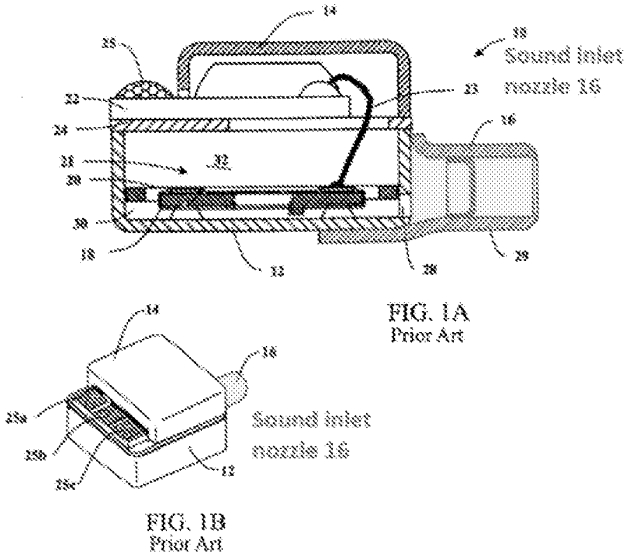
¶ [0032].) Having undergone BSS, the output signal contains the user's voice separated from background noise, resulting in "good intelligibility of speech." (*Id.*, ¶¶ [0007]-[0008]; *see also id.*, ¶¶ [0009]-[0012].) "Such a system could be used by any persons who need to communicate in a noisy environment, such as workers, soldiers, firemen, etc." (*Id.*, ¶ [0032].)

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. U.S. Patent Application Publication 2004/0047486A1 ("Van Doorn") (Ex. PA-D)

Van Doorn is a publication of U.S. Patent Application No. 10/236,649, entitled "MICROPHONE WITH IMPROVED SOUND INLET PORT." (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).

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.*).



Van Doorn, FIGs. 1A-B (annotated)

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].)

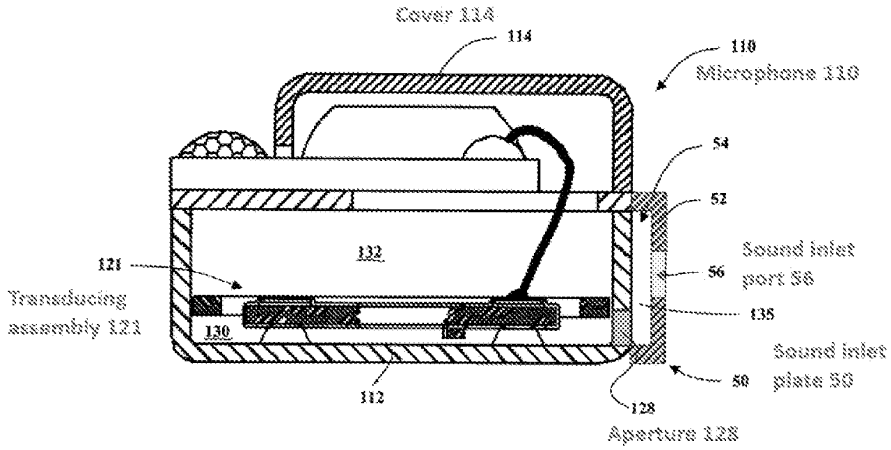


FIG. 3

Van Doorn, FIG. 3 (annotated)

"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. European Patent Application Publication EP1519625A2 ("Victorian") (Ex. PA-E)

Victorian is a publication of European Patent Application No. 04255520.1, entitled "EXTERNAL EAR CANAL VOICE DETECTION." (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).

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:

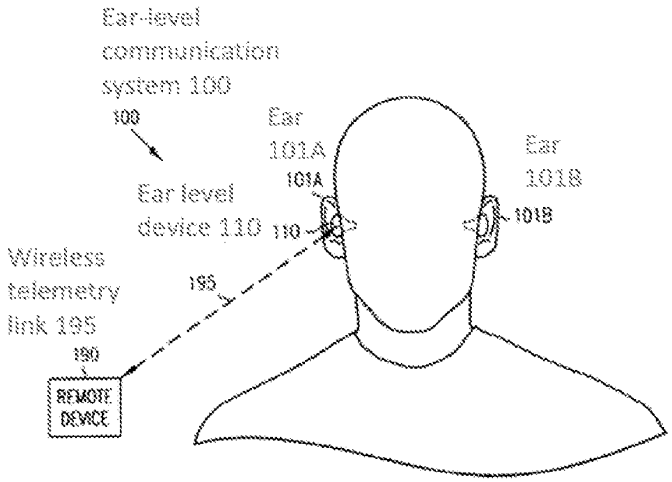
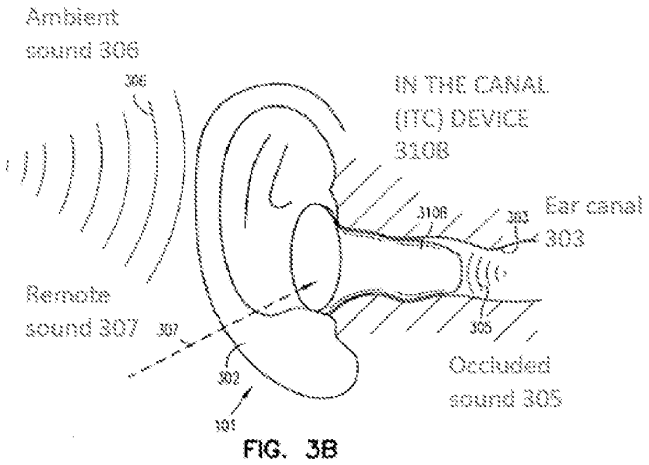


FIG. 1

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:



Victorian, FIG. 3B (annotated)

By reference to FIG. 4A, which is reproduced below, Victorian discloses the circuit of the single ear-level device 110. *Id.*, ¶ [0029].

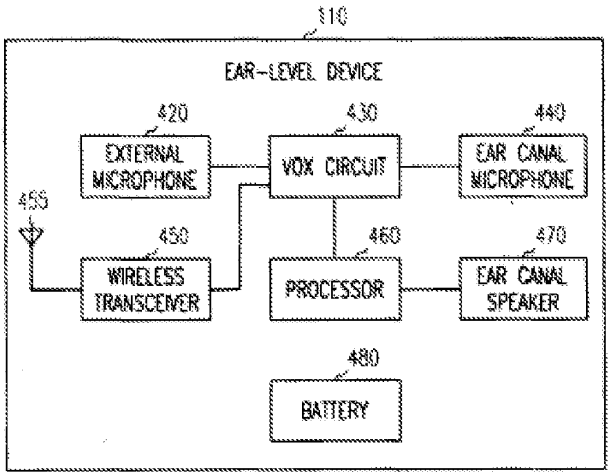


FIG. 4A

Victorian, FIG. 4A

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.*)

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].)

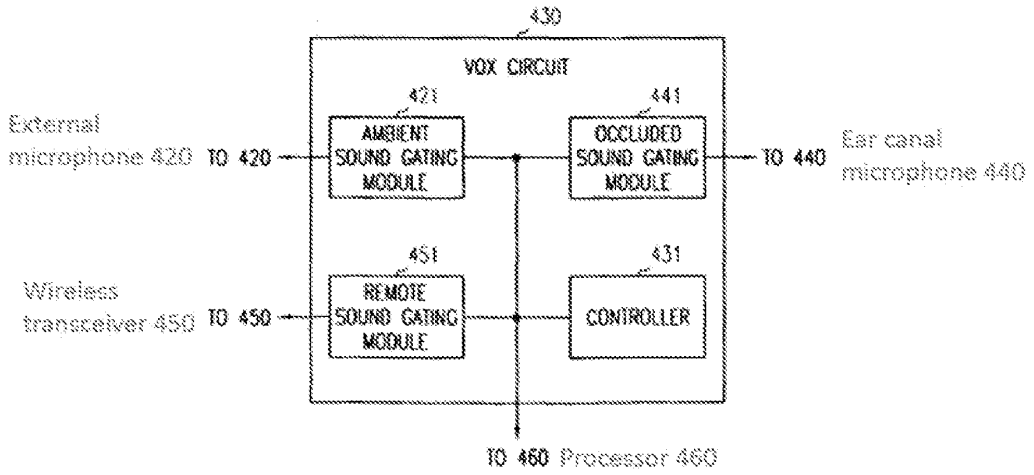


FIG. 4B

Victorian, FIG. 4B (annotated)

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. U.S. Patent Application Publication 2006/0153394A1 ("Beasley") (Ex. PA-F)

Beasley is a publication of U.S. Patent Application No. 11/328,890, entitled "HEADSET AUDIO BYPASS APPARATUS AND METHOD." (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).

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.*)

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. U.S. Patent 5,251,263 ("Andrea") (Ex. PA-G)

Andrea issued on October 5, 1993, and is entitled "ADAPTIVE NOISE CANCELLATION AND SPEECH ENHANCEMENT SYSTEM AND APPARATUS THEREFOR." (PA-G, Cover.) Accordingly, Andrea is prior art to the '666 patent under at least 35 U.S.C. § 102(b).

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.)

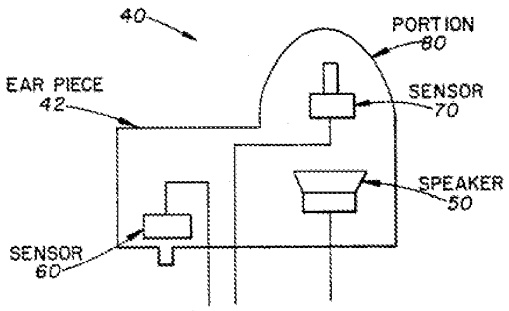
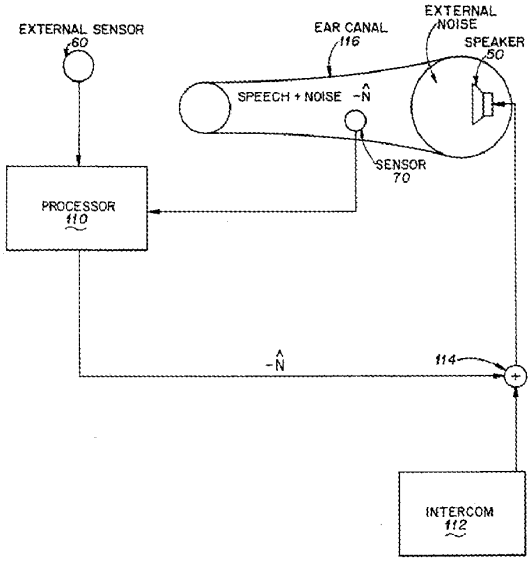


Fig. 4

Andrea, FIG. 4

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. U.S. Patent 7,903,826 ("Boersma") (Ex. PA-H)

Boersma issued on March 8, 2011, and is entitled "HEADSET WITH AMBIENT SOUND." (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).

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.)

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.)

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).

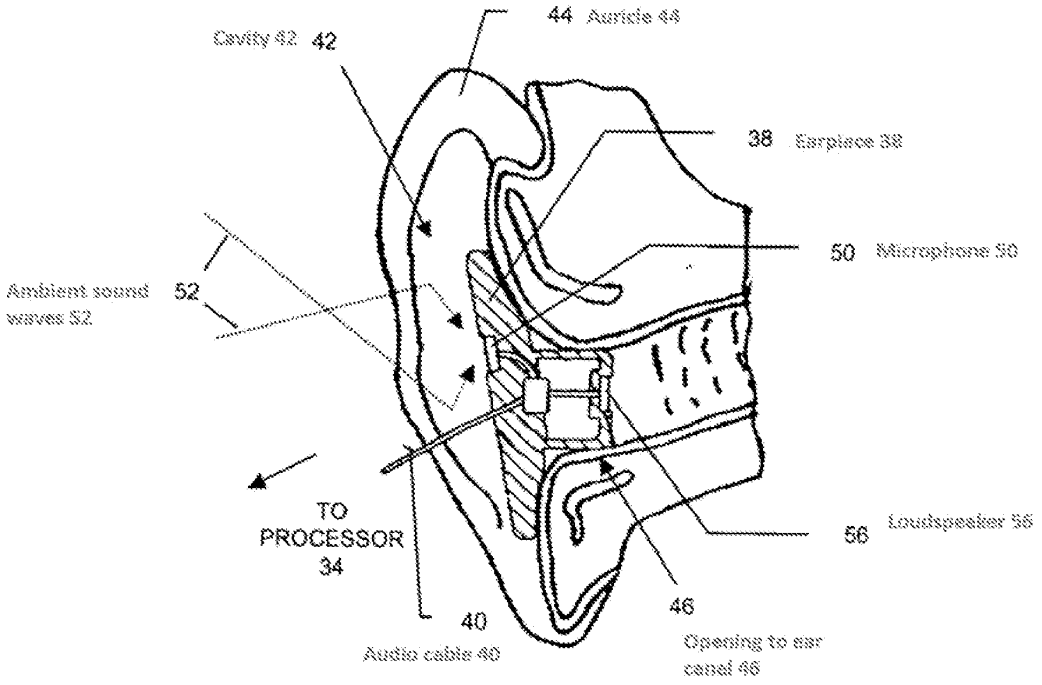


FIG. 2

Boersma, FIG. 2 (annotated)

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.)

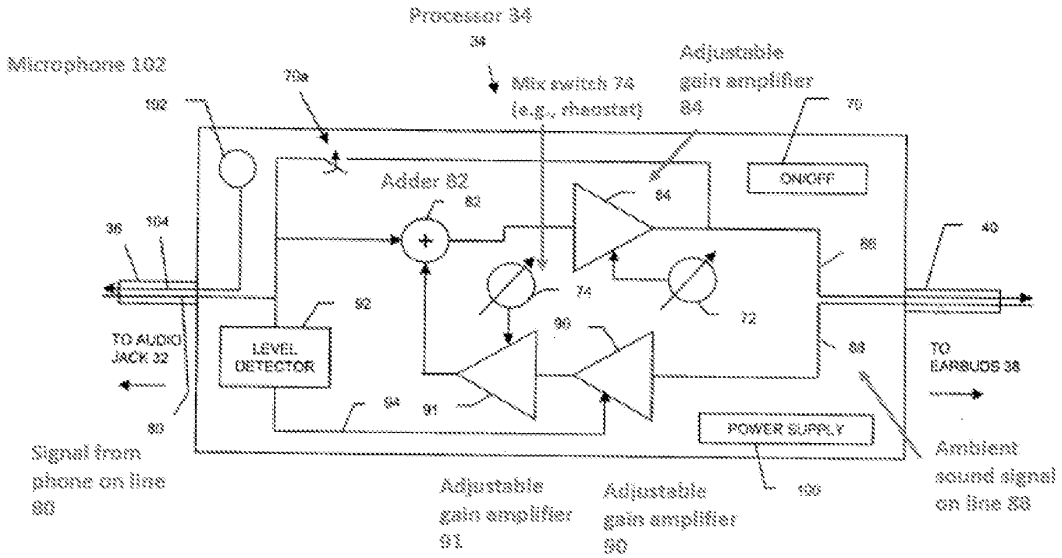


FIG. 4

Boersma, FIG. 4 (annotated)

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.)

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.)

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.)

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.

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.

VII. DETAILED EXPLANATION OF THE SUBSTANTIAL NEW QUESTIONS OF PATENTABILITY

A. SNQ 1: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast

Requesters respectfully submit that Nemirovski in combination with Rast would have rendered obvious claims 1-11 of the '666 patent under 35 U.S.C. § 103 for the reasons discussed below.

1. A POSITA Would Have Been Motivated to Combine Nemirovski and Rast

As explained further below, a POSITA would have been motivated to combine the teachings of Nemirovski and Rast and would have had a reasonable expectation of success in doing so. (OTH-A, Exp. Decl., ¶¶ 96-111.)

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. (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.)

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 POSITA would have understood that this is due to the acoustic isolation created by sensor 12 preventing environmental, or ambient, sound from reaching the user. (OTH-A, Exp. Decl., ¶¶ 98-99 (discussing U.S. Patent Application Publication No. 2003/0035551A1 ("Light") (PA-S), ¶ [0003]; U.S. Patent Application Publication No.

2006/0083388A1 ("Rothschild") (PA-T), ¶ [0005]).)[5] In view of these disclosures, a POSITA 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). (OTH-A, Exp. Decl., ¶ 100.) In so looking, a POSITA would have readily found Rast. (*Id.*)

Rast also recognizes that using noise-canceling headsets in the circumstances described by Nemirovski may be dangerous, but it proposes a solution to that problem. That is, 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. (PA-B, ¶¶ [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] (explaining that, with the "hearthrough" functionality, "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").)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 103.) For example, with Rast's solution implemented in

[5] Here, Light (PA-S) and Rothschild (PA-T) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.

Nemirovski, users could more often employ Nemirovski's superior two-sensor embodiment with less risk associated with being unable to effectively hear their surroundings. (*Id.*)

A POSITA 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). (*Id.*, ¶ 104.) The combination would also have been obvious to a POSITA because it would have 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). (*Id.*, ¶ 105.) 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). (*Id.*, ¶ 106.)

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). (*Id.*, ¶ 107.)

There is ample evidence that a POSITA 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. (*Id.*, ¶ 108.) Moreover, Nemirovski already discloses using "feature extraction and analysis using a neural net trained to recognize respective signals/features of such signals." (PA-A, 17:36-38, 18:27-32, 19:31-20:9.)

Other earpieces with a similar structural configuration to Nemirovski disclose a similar technique for playing ambient sound in the user's ear. (OTH-A, Exp. Decl., ¶¶ 109-110 (discussing U.S. Patent No. 6,754,359 ("Svean") (PA-I), 2:22-27, 4:51-55).)[6] 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 the prior art shows that there were solutions to that issue that a POSITA would have been able to implement. (OTH-A, Exp. Decl., ¶ 111 (discussing U.S. Patent No. 6,671,379 ("Nemirovski '379") (PA-U), Victorian (PA-E), and Svean (PA-I)).)[7]

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

To the extent the preamble is limiting, Nemirovski discloses or suggests an earphone. (OTH-A, Exp. Decl., ¶¶ 112-114.) Nemirovski discloses "a system 52" including a "sensor 12" with a housing 150 "to be inserted at least part way into the ear of a person." (PA-A, 9:3-28.) FIG.

[6] Here, Svean (PA-I) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.
[7] Here, Nemirovski '379 (PA-U), Victorian (PA-E), and Svean (PA-I) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.

2B, reproduced and annotated below, "illustrates sensor 12 inserted at least partially into the ear 100 of a person." (*Id.*, 11:63-67.)

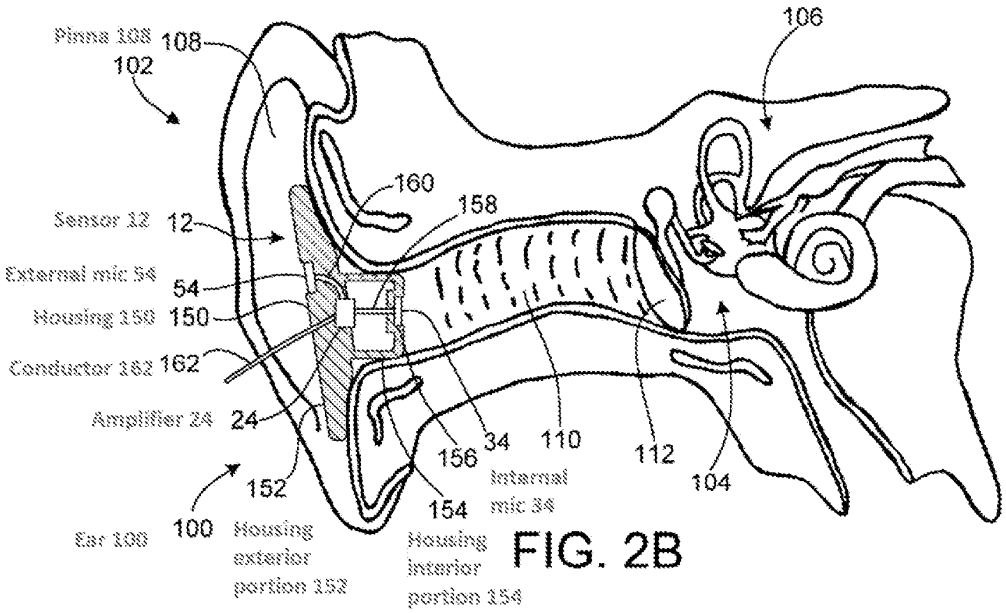


FIG. 2B

Nemirovski, FIG. 2B (annotated)

A POSITA would have understood Nemirovski's system 52 including sensor 12 to comprise "[a]n earphone" at least because it is of the same general construction as the "earphone" (or "earpiece") described in the '666 patent. (PAT-A, FIG. 1, 4:52-65; OTH-A, Exp. Decl., ¶ 114.)

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;"

Nemirovski discloses or suggests these features. (OTH-A, Exp. Decl., ¶¶ 115-133.)

Addressing element 1[b] first, Nemirovski discloses system 52 including an "internal microphone 34" that is disposed on housing 150. (PA-A, 9:10-15; *see also id.*, FIGs. 1E & 2B.) Nemirovski explains that "internal microphone 34 detects air pressure changes" within the ear. (*Id.*, 9:15-17.)

Nemirovski explains that the disclosure uses the term "changes in air pressure" to characterize "the parameter ***being measured***." (*Id.*, 11:20-23.)[8] Nemirovski further describes "[c]hanges in air pressure" as "sound waves." (*Id.*, 11:23-24.) Thus, a POSITA 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. (OTH-A, Exp. Decl., ¶ 117.) This is consistent with how the '666 patent describes this feature. (PAT-A, 4:66-5:3, 5:22-28; OTH-A, Exp. Decl., ¶ 118.)

Regarding the "second microphone port," the '666 patent does not define the term "microphone port." The "first microphone port" and "second microphone port" limitations were added by amendment during prosecution. (PAT-B, Aug. 4, 2021 Response to Office Action.) The applicant explained that Figures 1 and 2 supported the amendment and that it was made to "further distinguish the first and second microphone orientations (i.e., the associated microphone port through which measurements are taken)." (*Id.*, 6.)

A POSITA would have understood the term "microphone port" or "port" to refer to one or more related structures. (OTH-A, Exp. Decl., ¶¶ 120-126.) 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. (*See, e.g.*, U.S. Patent No. 7,983,433 ("Nemirovski '433") (PA-

[8] All emphases are added unless noted otherwise.

J), 7:13-23 (describing an earset with "a microphone port" as an opening in the housing of the earset); U.S. Patent No. 7,502,484 ("Ngia") (PA-K), 9:18-27 (describing 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"); *see also* U.S. Patent No. 8,351,634 ("Khenkin") (PA-L), 6:26-50 (defining "microphone port 110" as "a hole through which sound enters the housing 104").)[9]

The prior art also uses the term "port" to refer to an opening in a microphone package itself. (*See, e.g.*, Van Doorn (PA-D), Abstract, 1:12-27, 3:23-29 (describing an improved "sound inlet port" for a miniature microphone); U.S. Patent No. 7,953,241 ("Jorgensen") (PA-M), 6:28-45 (describing a "sound inlet port 5"); U.S. Patent No. 4,533,795 ("Baumhauer") (PA-N), 9:18-27 (describing an "acoustic port" in a capacitive microphone).)[10]

Thus, 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. (OTH-A, Exp. Decl., ¶ 126.)

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." (PA-A, 12:64-13:2.) A POSITA would have understood

[9] Here, Nemirovski '433 (PA-J), Ngia (PA-K), and Khenkin (PA-L) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.
[10] Here, Van Doorn (PA-D), Jorgensen (PA-M), and Baumhauer (PA-N) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.

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. (OTH-A, Exp. Decl., ¶ 127.)

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"). (OTH-A, Exp. Decl., ¶ 128.)

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." (PA-A, 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.)

For the same reasons as discussed with respect to element 1[b], a POSITA 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. (OTH-A, Exp. Decl., ¶ 130.) This is consistent with how the '666 patent describes this feature. (PAT-A, 4:66-5:3, 5:28-34 6:38-46; OTH-A, Exp. Decl., ¶ 130.)

Regarding the first microphone port, a POSITA 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 POSITA 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. For example, a POSITA 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. (OTH-A, Exp. Decl., ¶ 132.) In doing so, a POSITA 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. (*Id.*)

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"). (OTH-A, Exp. Decl., ¶ 133.)

c. 1[c] "a speaker configured to play an audio signal;"

Nemirovski discloses or suggests this feature. (OTH-A, Exp. Decl., ¶ 134.) 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." (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. (OTH-A, Exp. Decl., ¶ 134.)

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:"

Nemirovski in combination with Rast discloses or suggests these features. (OTH-A, Exp. Decl., ¶¶ 135-152.) Nemirovski discloses system 52 including "processing circuitry 14"." (PA-A, 9:29-40.) FIG. 1E, which depicts system 52 and includes processing circuitry 14" is reproduced below:

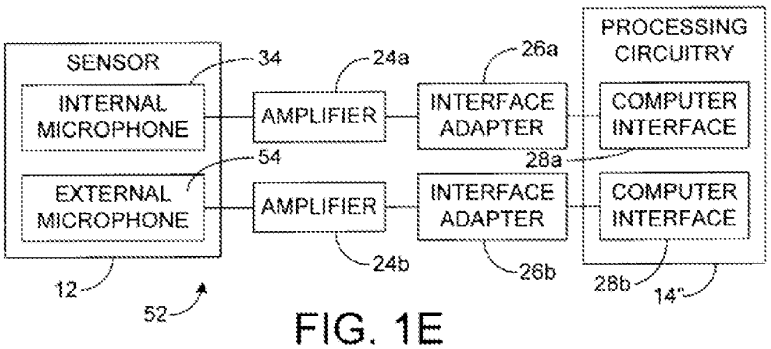


FIG. 1E

Nemirovski, FIG. 1E

A POSITA would have found it obvious to implement Nemirovski's processing circuitry 14" as a processor or signal processing unit. (OTH-A, Exp. Decl., ¶ 136.) 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. (PA-A, 6:46-51, 7:35-36.) Nemirovski also notes that "like reference numerals will refer to like elements throughout." (*Id.*, 3:42-45.) As such, a POSITA 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. (OTH-A, Exp. Decl., ¶ 136.)

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 POSITA 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 POSITA. (OTH-A, Exp. Decl., ¶ 137.)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 form factors, including earbuds. (*Id.*, ¶ [0013].)

For the reasons discussed in Section VII.A.1, a POSITA would have been motivated to modify Nemirovski's processing circuitry 14" 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. (OTH-A, Exp. Decl., ¶ 139.) A POSITA would also have been motivated to include processing circuitry 14" (implemented as a processor or signal processing unit) 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. (*Id.*, ¶¶ 139-140.)

For example, there was a recognized preference in the art for devices that were hands-free. (*See, e.g.*, U.S. Patent No. 8,014,553 ("Radivojevic") (PA-O), 1:58-63; OTH-A, Exp. Decl., ¶ 140.)[11] Additionally, a POSITA would have understood that implementing Rast's functionality within the headset would have provided the earpiece with interoperability advantages because the earpiece would not be reliant on processing power derived from a separate device. (*See, e.g.*, PA-H, 1:57-67; OTH-A, Exp. Decl., ¶¶ 140-141.)[12] 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 being listened to—are ***preferably implemented within the headset***." (PA-B, ¶ [0059].)

[11] Here, Radivojevic (PA-O) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.
[12] Here, Boersma (PA-H) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.

Moreover, a POSITA 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. (OTH-A, Exp. Decl., ¶¶ 142-143 (discussing U.S. Patent No. 6,728,385 ("Kvaløy") (PA-P) and Radivojevic (PA-O)).)[13] 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.) Thus, any expected tradeoffs associated with a smaller (or larger) form factor would not have dissuaded a POSITA from making the modifications necessary to achieve the claimed invention. (OTH-A, Exp. Decl., ¶¶ 144-145.)

Accordingly, in the combined Nemirovski-Rast device, it would have been obvious to a POSITA 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. (*Id.*, ¶ 146.)

Nemirovski discloses that each of internal microphone 34 and external microphone 54 are connected to processing circuitry 14". (PA-A, 9:29-41.) As explained in Section VII.A.2.a, Nemirovski also describes a speaker 38 included within the housing 150. Although Nemirovski does not explicitly disclose processing circuitry 14" coupled to speaker 38, it would have been obvious to a POSITA 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

[13] Here, Kvaløy (PA-P) and Radivojevic (PA-O) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.

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). (OTH-A, Exp. Decl., ¶¶ 147-148.) 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. (PA-B, ¶ [0063].)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 149.) As such, a POSITA would have found it obvious to couple Nemirovski's processing circuitry 14" to each of internal microphone 34, external microphone 54, and speaker 38.

A POSITA 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. (*Id.*, ¶ 150.) 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)." (PA-A, 10:18-25.) A POSITA 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. (OTH-A, Exp. Decl., ¶ 151.) At the very least, it would have been obvious in view of the knowledge of a POSITA 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. (OTH-A, Exp. Decl.,

¶ 151 (discussing PA-I, 6:45-58; PA-C, ¶ [0029]).)[14] Moreover, when implementing Nemirovski's processing circuitry 14" as a processor, or signal processing unit, located within housing 150, it would have been obvious to a POSITA 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. (PA-B, ¶ [0053], OTH-A, Exp. Decl., ¶ 151.)

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"). (OTH-A, Exp. Decl., ¶ 152.) 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;"

Nemirovski discloses or suggests these features. (OTH-A, Exp. Decl., ¶ 153.) 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." (PA-A, 15:18-20.) Nemirovski further describes

[14] Here, Svean (PA-I) and Dijkstra (PA-C) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 1.

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"). (OTH-A, Exp. Decl., ¶ 153.)

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;"

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 154-163.) As discussed in Section VII.A.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. (PA-A, 15:48-16:7; OTH-A, Exp. Decl., ¶ 155.) More particularly, Nemirovski discloses "filter[ing]" or "mathematically remov[ing]" the noise detected by external microphone 54 from the internal signal to isolate the initiating action. (PA-A, 15:48-67; OTH-A, Exp. Decl., ¶ 155.) Nemirovski discloses or suggests similarly 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." (PA-A, 16:1-7; OTH-A, Exp. Decl., ¶ 156.)

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. (*See, e.g.*, '666 patent, 9:3-4 ("At step 402, the earpiece 100 ***captures***

ambient sound signals from the ASM 111."); 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."); OTH-A, Exp. Decl., ¶¶ 157-160.) The claims 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.) The applicant did not expressly describe the written description support for the amendment.

In view of the above, a POSITA 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. (OTH-A, Exp. Decl., ¶ 161.) Such an interpretation is consistent with the description of the method of Figure 9, which describes 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 POSITA 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." (OTH-A, Exp. Decl., ¶ 162.) Moreover, a POSITA 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." (*Id.*)

Accordingly, a POSITA 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. (OTH-A, Exp. Decl., ¶ 163.)

g. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 164-169.) Rast discloses that in "hearthrough mode," the "acoustics of the environment" are routed to the headset wearer at a "user selected amplification level." (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.)

In the Nemirovski-Rast device, a POSITA 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. (OTH-A, Exp. Decl., ¶ 165.) 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. (*Id.*) Indeed, Rast itself suggests using a noise-reduced signal for its hearthrough feature. (PA-B, ¶ [0014]

(implementing hearthrough by "amplifying the ambient sounds ***after extracting spurious noises***").)

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 70*a*, 70*b* to the separate channels of the earpiece speakers 56*a*, 56*b*." (*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 "simplify[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 POSITA would have considered it obvious to implement the functionality of Rast's amplifier circuit 68 and amplifiers 70 using processing unit 14" implemented as a processor or signal processing unit, and a POSITA would have been motivated to do so. (OTH-A, Exp. Decl., ¶¶ 166-167 (discussing Mulgrew, Bernard *et al.*, Digital Signal Processing: Concepts and Applications. (2nd ed. 2002) ("Mulgrew") (PA-W)).)[15] A POSITA 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. (OTH-A, Exp. Decl., ¶ 168).

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

[15] Here, Mulgrew (PA-W) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.

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"). (OTH-A, Exp. Decl., ¶ 169.)

h. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 170-173.) For example, Rast discloses that a headset may receive "audio signals" such as music programming or communications. (PA-B, ¶¶ [0011], [0022], [0025], [0055], [0059], [0061].) Nemirovski similarly discloses broadcasting received sounds from a cell phone to the user. (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").)

As explained in Section VII.A.2.g, 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." (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].)

As also explained in Section VII.A.2.g, in the Nemirovski-Rast device, a POSITA 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. (OTH-A, Exp. Decl., ¶ 172.) And as explained in Section VII.D.2.g, a POSITA would have understood that although Rast

describes an embodiment in which the mixing is performed in an analog circuit, a POSITA would have been motivated to implement the system digitally such that a processor digitally mixes the ambient sound signal and the incoming received sounds. (*Id.*)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 173.)

i. 1[k] "sending the mixed audio signal to the speaker."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 174-175.) 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." (PA-B, ¶ [0022]; *see also id.*, ¶¶ [0011], [0012], [0014], [0023], [0025], [0051], [0058], [0059], [0060], [0061], Table 1.)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 175.)

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"

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 176-181.) 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. (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.").)

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].) Rast describes conducting a sound bite-by-sound bite comparison to stored sounds. (*Id.*) 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. (OTH-A, Exp. Decl., ¶¶ 177-178.)

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 POSITA would have understood Rast's sound bite-by sound bite routine to be consistent with searching for the on-set of a programmed sound. (OTH-A, Exp. Decl., ¶ 179.)

As discussed in Section VII.A.1, a POSITA 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. (*Id.*, ¶ 180.) 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 VII.A.1, a POSITA would have found it obvious to modify Nemirovski's system to utilize the sound bite-by-sound bite correlation 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. (*Id.*) A POSITA 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). (*Id.*)

Accordingly, a POSITA 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 …."). (*Id.*, ¶ 181.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 182-184.) As explained in Section VII.A.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." (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.").)

Accordingly, a POSITA 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 …."). (OTH-A, Exp. Decl., ¶ 183.)

For the reasons discussed in Sections VII.A.1 and VII.A.3.a, a POSITA 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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 185-189.) As discussed in Section VII.A.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. (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]. 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].)

Rast further discloses that 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 POSITA would have understood discerning sounds based on stored sound signatures to include performing a spectrum analysis. (OTH-A, Exp. Decl., ¶ 187 (discussing U.S. Patent No. 9,135,797 ("Couper") (PA-V), Abstract, 4:60-5:4, 5:18).)[16]

In view of the above disclosures, a POSITA would have understood Rast to disclose or suggest performing a spectrum analysis to match a detected sound with the stored sound signature for a siren commonly in use. (OTH-A, Exp. Decl., ¶ 188.) A POSITA would have thus 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. (*Id.*)

Accordingly, a POSITA 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 ….") (*Id.*, ¶ 189.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 190-191.) 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. (PA-B, ¶ [0051].) As explained in Sections VII.A.3 and VII.A.4, the programmed

[16] Here, Couper (PA-V) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.

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].)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 191.)

6. Claim 5

a. "5. The earphone according to claim 4, where the operations further comprise: sending a notification signal to the speaker."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 192-193.) 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." (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 POSITA 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. (OTH-A, Exp. Decl., ¶ 192.)

Accordingly, a POSITA 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. (*Id.*, ¶ 193.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 194-195.) Rast discloses detecting a correlation between environmental sounds from microphones 58a, 58b and stored sound parameters. (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].)

Accordingly, a POSITA 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 ...."). (OTH-A, Exp. Decl., ¶ 195.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 196-197.) Rast discloses detecting a correlation between environmental sounds from microphones 58a, 58b and stored sound parameters. (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 POSITA 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. (OTH-A, Exp. Decl., ¶ 196.) Rast also describes the processing unit identifying a horn or screeching brakes. (PA-B, ¶ [0021].)

Accordingly, a POSITA 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 ...."). (OTH-A, Exp. Decl., ¶ 197.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 198-200.) As discussed in Section VII.A.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. 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." (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.*)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 200.)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶ 201.) Indeed, for the same reasons discussed above in Section VII.A.9, a POSITA 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."). (*Id.*)

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."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 202-205.) As explained above in Section VII.A.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 VII.A.2.f, in the Nemirovski-Rast device, a POSITA 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.

A POSITA would have understood 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. (OTH-A, Exp. Decl., ¶ 203.) Indeed, a POSITA 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. (*Id.* (citing PA-F, ¶¶[0031], [0050]-[0051]).)[17] As such, a POSITA 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. (*Id.*)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 204.) Therefore, a POSITA would have understood the inverse signal of the noise detected by internal microphone 34 to comprise "a noise reduction signal." (*Id.*)

Accordingly, a POSITA 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

[17] Here, Beasley (PA-F) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 1.

microphone 34 ("includes a noise reduction signal derived from the second microphone signal"). (*Id.*, ¶ 205.)

12. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

Nemirovski in combination with Rast discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 206-207.) 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." (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.)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 207.)

B. SNQ 2: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Dijkstra

As discussed in Section VII.A.2.b, 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. This SNQ 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.

Although Requesters do not concede it, to the extent that Patent Owner asserts or the Office finds that Nemirovski does not already disclose or suggest the "first microphone port" and "second microphone port" limitations under this interpretation of the term "microphone port" (*see supra* Section VII.A.2.b), Requesters respectfully submit that Nemirovski in combination with Rast and Dijkstra 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

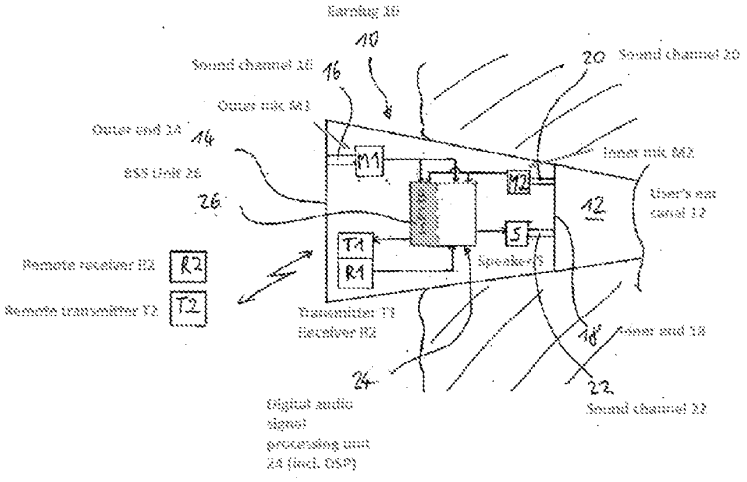
With respect to 1[pre] and 1[c]-1[k], Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section VII.A, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section VII.A.2.a.
1[c]	*See supra* Section VII.A.2.c.
1[d]	*See supra* Section VII.A.2.d.
1[e]	*See supra* Section VII.A.2.d.
1[f]	*See supra* Section VII.A.2.e.
1[g]	*See supra* Section VII.A.2.e.
1[h]	*See supra* Section VII.A.2.f.
1[i]	*See supra* Section VII.A.2.g.
1[j]	*See supra* Section VII.A.2.h.
1[k]	*See supra* Section VII.A.2.i.

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 VII.A.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 POSITA would have been motivated to combine Dijkstra with Nemirovski and Rast with a reasonable expectation of success.

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. (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)

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.*)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 213.) 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. 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 POSITA would have understood that sound channel 16 and sound channel 20 terminate at openings in earplug 10. (*Id.*, ¶ 214 (discussing PA-K, 9:18-27).)[18]

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 is inserted" and "wherein the second microphone has a second microphone port that is configured to face toward the user when the

[18] Here, Ngia (PA-K) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 2.

earphone is inserted." (*See supra* Section IV.B.) 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.)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 216.) A POSITA would have recognized that disposing external microphone 54 and internal microphone 34 within housing 150 would better protect the microphones from physical damage. (*Id.*, ¶ 217.) A POSITA 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. (*Id.*, ¶¶ 217-218 (discussing U.S. Patent No. 8,150,084 ("Jessen") (PA-R), 2:55-61).)[19]

A POSITA 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

[19] Here, Jessen (PA-R) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 2.

Dijkstra in the combined Nemirovski-Rast device. (*Id.*, ¶ 219.) A POSITA 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. (*Id.*) Nemirovski also already describes an embodiment in which internal microphone 34 is disposed within a hollow cavity defined by the housing 150. (PA-A, 12:64-13:2.) A POSITA 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). (OTH-A, Exp. Decl., ¶ 220.)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 221.)

2. <u>Dependent Claims 2-11</u>

With respect to dependent claims 2-11, Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section VII.A, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section VII.A.3.
3	*See supra* Section VII.A.4.
4	*See supra* Section VII.A.5.
5	*See supra* Section VII.A.6.
6	*See supra* Section VII.A.7.
7	*See supra* Section VII.A.8.
8	*See supra* Section VII.A.9.
9	*See supra* Section VII.A.10.
10	*See supra* Section VII.A.11.
11	*See supra* Section VII.A.12.

Accordingly, for at least the same reasons, these claims would have been obvious over Nemirovski, Rast, and Dijkstra.

C. SNQ 3: Claims 1-11 are Obvious in View of Nemirovski in Combination with Rast and Van Doorn

As discussed in Section VII.A.2.b, 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. This SNQ 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, Requesters respectfully submit 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

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 VII.A, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section VII.A.2.a.
1[c]	*See supra* Section VII.A.2.c.
1[d]	*See supra* Section VII.A.2.d.
1[e]	*See supra* Section VII.A.2.d.
1[f]	*See supra* Section VII.A.2.e.
1[g]	*See supra* Section VII.A.2.e.
1[h]	*See supra* Section VII.A.2.f.
1[i]	*See supra* Section VII.A.2.g.
1[j]	*See supra* Section VII.A.2.h.
1[k]	*See supra* Section VII.A.2.i.

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 VII.A.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 POSITA would have been motivated to combine Van Doorn with Nemirovski and Rast with a reasonable expectation of success.

Van Doorn is directed to "a microphone or listening device with an improved sound inlet port." (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:

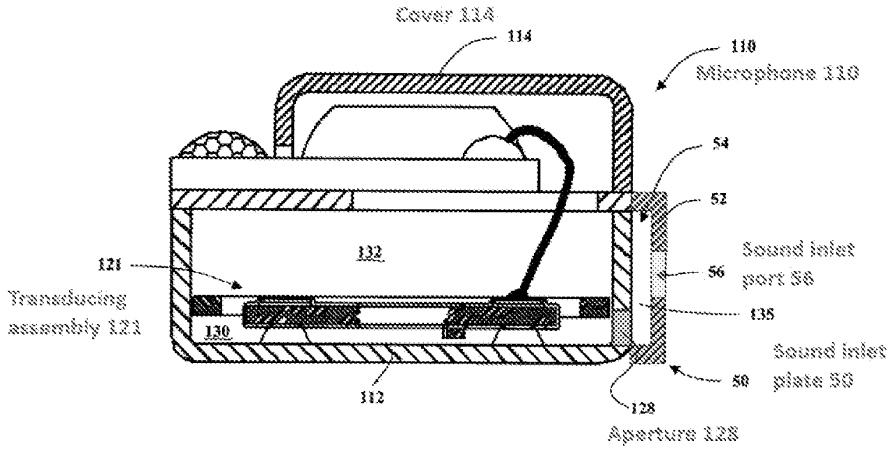


FIG. 3

Van Doorn, FIG. 3 (annotated)

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].)

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").)

A POSITA would have understood Van Doorn's sound port 56 to comprise an opening in microphone 110 through which measurements are taken. (OTH-A, Exp. Decl., ¶ 230.) 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 POSITA would have understood Van Doorn's sound port 56 to comprise a "microphone port." (*Id.*)

As discussed above in Section VII.A.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. (*Id.*, ¶ 231.) As such, a POSITA would have been motivated for look for microphones for use in sensor 12 as external microphone 54 and internal microphone 34. (*Id.*) In so looking, a POSITA would have readily found Van Doorn.

A POSITA 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. (*Id.*, ¶ 232.) Indeed, a POSITA 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 (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]). (OTH-A, Exp. Decl., ¶ 232.) A POSITA 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. (*Id.*, ¶ 233 (citing PA-D, ¶[0006]).) A POSITA 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). (*Id.*, ¶ 234.)

Accordingly, a POSITA 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"). (*Id.*, ¶ 235.)

2. Dependent Claims 2-11

With respect to dependent claims 2-11, Nemirovski in combination with Rast meets the limitations in the same way discussed above in Section VII.A, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section VII.A.3.
3	*See supra* Section VII.A.4.
4	*See supra* Section VII.A.5.
5	*See supra* Section VII.A.6.
6	*See supra* Section VII.A.7.
7	*See supra* Section VII.A.8.
8	*See supra* Section VII.A.9.
9	*See supra* Section VII.A.10.
10	*See supra* Section VII.A.11.
11	*See supra* Section VII.A.12.

Accordingly, for at least the same reasons, these claims would have been obvious over Nemirovski, Rast, and Van Doorn.

D. SNQ 4: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Dijkstra and Beasley

As discussed in Section VII.A.2.b, 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. This SNQ 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, Requesters respectfully submit 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 POSITA Would Have Been Motivated to Combine Victorian and Beasley

As explained further below, a POSITA would have been motivated to combine the teachings of Victorian and Beasley and would have had a reasonable expectation of success in doing so. (OTH-A, Exp. Decl., ¶¶ 238-246.)

Victorian discloses ear-level device 110 including an "external microphone 420 [that] picks up ambient sound" and an "[e]ar canal microphone 440 [that] detects sound from the ear canal," that is primarily the wearer's speech. (PA-E, ¶ [0030].) Victorian discloses implementing ear-level device 110 with an earmold that fully occludes the ear to maximize audio efficiency. (*Id.*, Abstract, ¶ [0027].)

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].) 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 blanked or substantially attenuated." (*Id.*, ¶ [0032].) 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].)

Beasley discloses an audio bypass mode "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. (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. (*Id.*, ¶¶ [0037], [0061], claims 18, 19, 30, 31.)

A POSITA would have been motivated to modify Victorian in view of Beasley. (OTH-A, Exp. Decl., ¶ 242.) More particularly, a POSITA 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. 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. (*Id.*)

Accordingly, a POSITA 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). (*Id.*, ¶ 243.)

The combination would also have been obvious to a POSITA 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). (*Id.*, ¶ 244.)

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). (*Id.*)

As noted, 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. (*Id.*, ¶ 245.)

There is ample evidence that a POSITA 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. (OTH-A, Exp. Decl., ¶ 246.) 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." (PA-E, ¶ [0030].)

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

To the extent the preamble is limiting, Victorian discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 247-248.) Victorian discloses an ear-level device 110. (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 and used as an earpiece. (*Id.*, Abstract, ¶¶ [0008], [0024]-[0025].) 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"). (OTH-A, Exp. Decl., ¶ 248.)

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;"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 249-258.) Victorian discloses ear-level device 110 including an "external microphone 420 [that] picks up ambient sound" and an "[e]ar canal microphone 440 [that] detects sound from the ear canal." (PA-E, ¶[0030].)

A POSITA would have understood that ambient sound captured by external microphone 420 necessarily comprises a measure of the sound reaching that microphone. (OTH-A, Exp. Decl., ¶ 251.) 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." (PA-E, ¶ [0033].) Accordingly, a POSITA 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. (OTH-A, Exp. Decl., ¶¶ 252-253.)

Similarly, a POSITA 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. (*Id.*, ¶ 254.) 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." (PA-E, ¶¶ [0034], [0031].) Accordingly, a POSITA 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. (OTH-A, Exp. Decl., ¶¶ 254-255.)

It would have been obvious to a POSITA 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 VII.B.1. (*Id.*, ¶ 256.)

A POSITA 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. (*Id.*, ¶ 257.) Indeed, a POSITA 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. (*Id.*)

A POSITA 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). (*Id.*)

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"). (*Id.*, ¶ 258.)

c. 1[c] "a speaker configured to play an audio signal;"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 259-260.) 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." (PA-E, ¶¶ [0030], [0051].)

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"). (OTH-A, Exp. Decl., ¶ 260.)

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:"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 261-264.) Victorian discloses ear-level device 110 including "a voice operated exchange (VOX) circuit 430" and "a processor 460." (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.

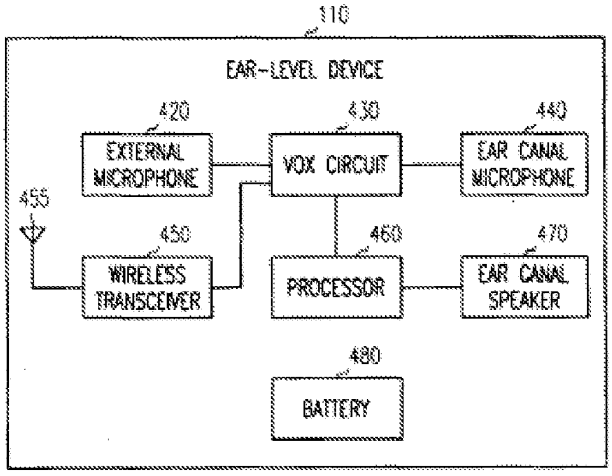


FIG. 4A

Victorian, FIG. 4A

A POSITA 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 in a more efficient manner. (OTH-A, Exp. Decl., ¶ 262 (discussing PA-I, 6:35-51; PA-Q)[20]; *see also DyStar Textilfarben GmbH & Co. Deutschland KG v. C.H. Patrick Co.*, 464 F.3d 1356, 1368 (Fed. Cir. 2006) (holding that that "an implicit motivation to combine exists not only when a suggestion may be gleaned from the prior art as a whole, but when the 'improvement' is technology-independent and the combination of references results in a product or process that is more desirable, for example because it is stronger, cheaper, cleaner, faster, lighter, smaller, more durable, ***or more efficient***.") (emphasis added).)

A POSITA would have known that a digital processor would operate by executing instructions and that these instructions would be stored in a memory. (OTH-A, Exp. Decl., ¶ 263.)

[20] Here, Svean (PA-I) and Oshana (PA-Q) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 4.

At the very least, it would have been obvious in view of the knowledge of a POSITA 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 more efficiently. (*Id.* (discussing PA-I, 6:52-58; PA-Q).)[21]

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"). (OTH-A, Exp. Decl., ¶ 264.) Victorian discloses or suggests processor 460 performing the claimed operations, as discussed below.

e. 1[f] "receiving a first microphone signal from the first microphone;"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 265-266.) 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." (PA-E, ¶ [0030].)

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. (OTH-A, Exp. Decl., ¶ 266.)

[21] Here, Svean (PA-I) and Oshana (PA-Q) are being used as evidence of a POSITA's knowledge and the state of the art; they are not part of the combination in SNQ 4.

f. 1[g] "receiving a second microphone signal from the second microphone;"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 267-268.) Victorian discloses processor 460 converting "the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190." (PA-E, ¶ [0030].)

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. (OTH-A, Exp. Decl., ¶ 268.)

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;"

Victorian in combination with Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 269-273.) 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].) As explained above in Section VII.A.2.f, a POSITA 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, including at least equalizing or applying a gain to the signal. A POSITA 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. (OTH-A, Exp. Decl., ¶ 271.)

Thus, a POSITA would have understood that "converting" and "enhancing" 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. (OTH-A, Exp. Decl., ¶ 272.) A POSITA 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. (*Id.*)

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 …"). (*Id.*, ¶ 273.)

h. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 274-277.) Beasley discloses implementing an automatic audio bypass module utilizing a digital signal processor (DSP). (PA-F, ¶ [0054].) An embodiment utilizing DSP technology is depicted in Beasley's FIG. 10, which is reproduced and annotated below:

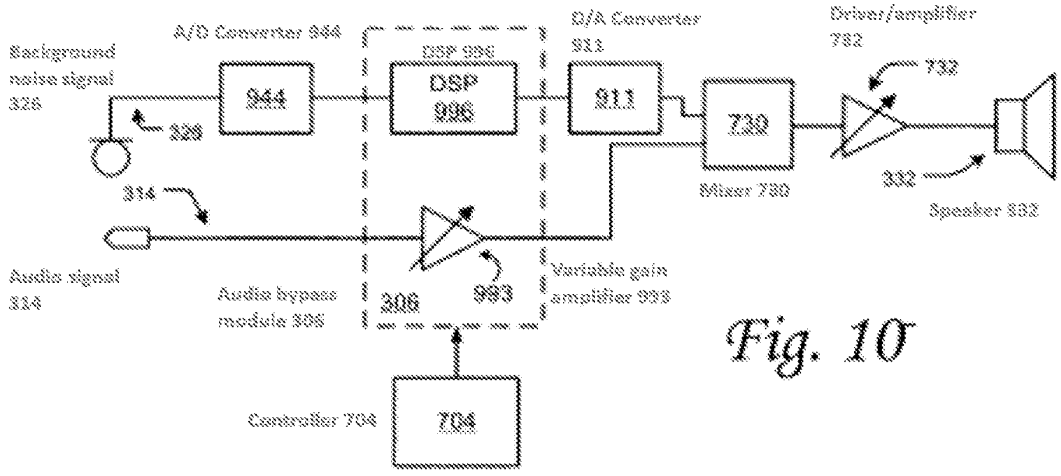


Fig. 10

Beasley, FIG. 10 (annotated)

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 POSITA 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"); OTH-A, Exp. Decl., ¶ 275.) Beasley also discloses utilizing a digital signal processor to amplify the speech signal. (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 POSITA would have understood utilizing a digital signal processor to amplify the speech signal to comprise DSP 996 amplifying background noise signal 326. (OTH-A, Exp. Decl., ¶ 275.)

A POSITA would thus have understood that Beasley discloses or suggests amplifying the ambient sound so that it can be better heard by the user. (*Id.*, ¶ 276.) A POSITA 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. (*Id.*) 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." (PA-F, ¶ [0051].) A POSITA 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. (OTH-A, Exp. Decl., ¶ 276.)

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"). (*Id.*, ¶ 277.)

i. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 278-280.) For example, Victorian discloses a gating rule that allows the transmission of both the remote sound and the ambient sound to the ear canal. (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 POSITA 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."); OTH-A, Exp. Decl., ¶ 278.) That is, because Victorian discloses that both the ambient sound and remote sound are provided to the speaker, a POSITA would have understood that the processor would have to mix the two signals. (OTH-A, Exp. Decl., ¶ 278.)

For the reasons discussed in Section VII.D.1, it would have been obvious to a POSITA 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 amplified background audio signal with the audio content from the remote device for playback by the ear canal speaker 470. (*Id.*, ¶ 279.)

Accordingly, a POSITA would have understood the combined Victorian-Dijkstra-Beasley device to disclose or suggest processor 460 mixing the amplified 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"). (*Id.*, ¶ 280.)

j. 1[k] "sending the mixed audio signal to the speaker."

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 281-282.) For example, Victorian discloses a gating rule that allows the transmission of both the remote sound and the ambient sound to the ear canal. (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 VII.D.2.i, a POSITA 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.

Accordingly, a POSITA 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. (OTH-A, Exp. Decl., ¶ 282.)

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"

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 283-286.) 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. (PA-F, ¶¶ [0037], [0061].)

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.*)

A POSITA 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 POSITA 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. (OTH-A, Exp. Decl., ¶ 284.) In the combined Victorian-Dijkstra-Beasley device, a POSITA would have understood that one or more of these techniques would be performed on signals picked up by Victorian's external microphone 420. (*Id.*)

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. (PAT-A, 5:40-55.)

Accordingly, a POSITA 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 …."). (OTH-A, Exp. Decl., ¶ 286.)

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."

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 287-290.) As discussed in Section VII.D.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 POSITA would have understood that each of the events detected by these techniques comprise the user's voice. (*Id.*, ¶ 287.) 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." (PA-F, ¶ [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 VII.D.3.a, a POSITA 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. (OTH-A, Exp. Decl., ¶ 287.)

Accordingly, a POSITA 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 …."). (*Id.*, ¶ 288.)

Although Requesters do not concede it, 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, a POSITA would have nonetheless found each of these features to be obvious in view of the teachings of Beasley. (OTH-A, Exp. Decl., ¶ 289.)

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." (PA-F, ¶ [0061].) However, a POSITA 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. (OTH-A, Exp. Decl., ¶ 289.) As such, a POSITA 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. (*Id.*) Indeed, Beasley's claims contemplate using both speech recognition and background audio level threshold detection. (PA-F, claim 19 ("[W]herein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection.").)

Accordingly, a POSITA 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 …."). (OTH-A, Exp. Decl., ¶ 290.)

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."

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 291-292.) As discussed in Section VII.D.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." (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 VII.D.2.f, a POSITA would have understood Beasley to disclose that "enhancing" the signal so that it can be "better heard" includes amplification of the signal. (OTH-A, Exp. Decl., ¶ 291.) As discussed in Section VII.D.2.i, Beasley discloses or at a minimum renders obvious implementing the functionality of variable gain amplifier 993 in software executed by DSP 996.

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 292.)

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."

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶ 293.) Indeed, for the same reasons discussed above in Section VII.D.4, a POSITA 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. (*Id.*)

6. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

Victorian in combination with Dijkstra and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 294-296.) Victorian discloses embodiments in which an earmold is used with the ear-level device. (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:

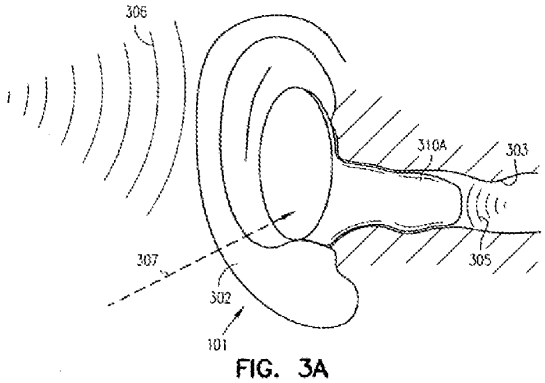


FIG. 3A

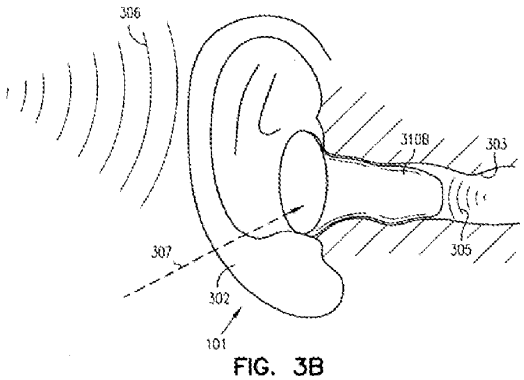


FIG. 3B

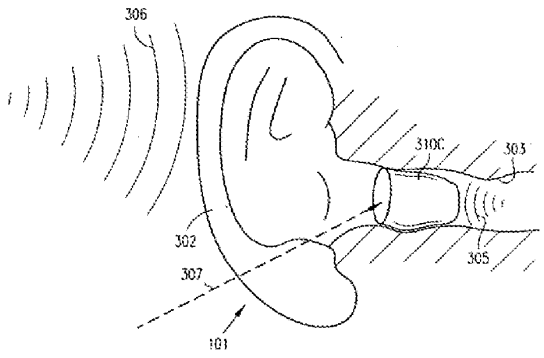


FIG. 3C

Victorian, FIGs. 3A-C

These embodiments include the same "ear-level device 110" discussed above with respect to claim 1. (*See id.*, [0029].) A POSITA 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.). (OTH-A, Exp. Decl., ¶ 295.) At a minimum a POSITA 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." (*Id.*, ¶ 295.)

Accordingly, a POSITA 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"). (*Id.*, ¶ 296.)

E. SNQ 5: Claim 10 is Obvious in View of Victorian in Combination with Dijkstra, Beasley, and Andrea

Requesters respectfully submit 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."

Victorian in combination with Dijkstra, Beasley, and Andrea discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 298-309.) 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. (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.

Andrea is directed to an adaptive noise cancellation (ANC) and speech enhancement system. (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:

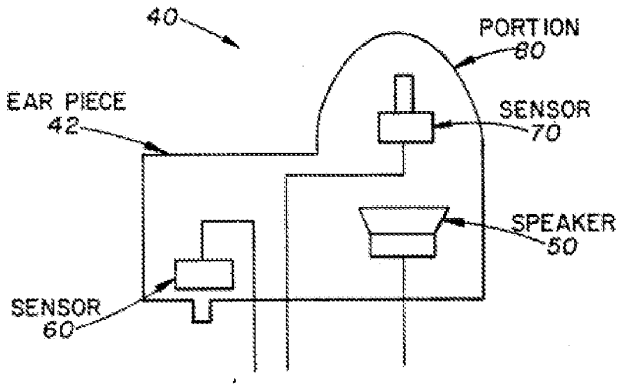
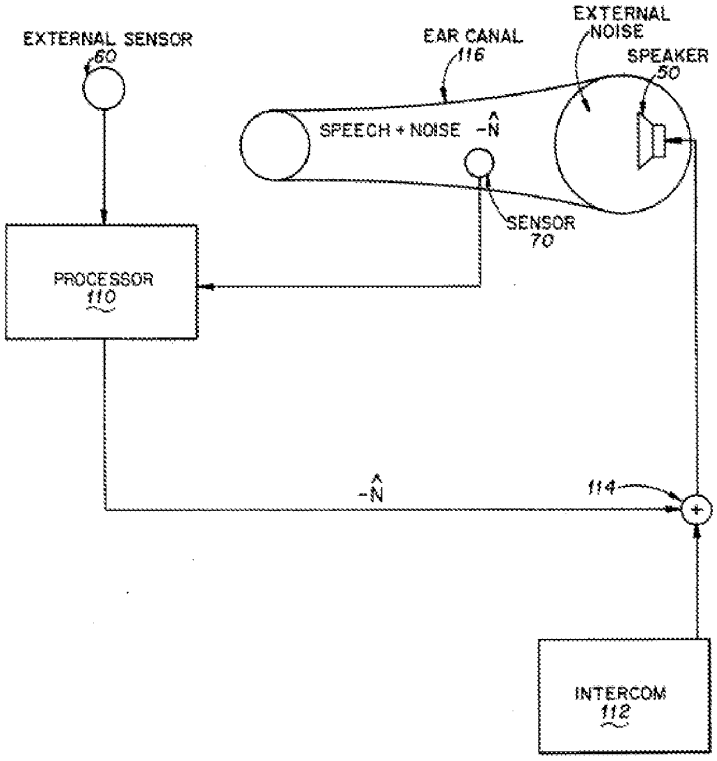


Fig. 4

Andrea, FIG. 4

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.)

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

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.)

A POSITA would have understood the output of Andrea's processor 110 representing anti-noise to comprise a noise reduction signal. (OTH-A, Exp. Decl., ¶ 303.) A POSITA would also have understood the output of Andrea's processor 110 generated from a received signal that

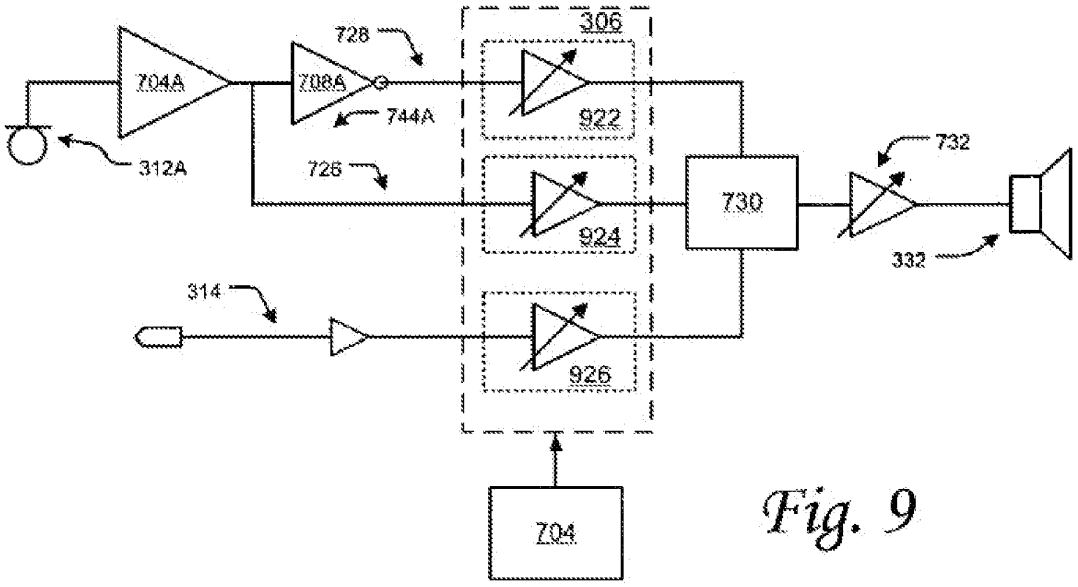
includes sound picked up by sensor 60 and a feedback or error signal from sensor 70 to comprise an output derived from a signal from sensor 70. (*Id.*) A POSITA would also have understood Andrea's microphone 70 to comprise an internal microphone configured to detect sound from the ear canal. Accordingly, a POSITA 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"). (*Id.*)

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 (PA-E, ¶ [0030]), a POSITA 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. (*Id.*, ¶ 304.) In so looking, a POSITA would have readily found Andrea. (*Id.*) Indeed, a POSITA 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. (*Id.*)

A POSITA 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. (*Id.*, ¶ 305.) A POSITA 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. (*Id.*) A POSITA would also have understood that Andrea describes implementing its noise reduction functionality with digital signal processing technology.

(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)."); OTH-A, Exp. Decl., ¶ 305.)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 306.) 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. (PA-F, ¶ [0053].) The foregoing embodiment is depicted in Beasley's FIG. 9, which is reproduced below:



Beasley, FIG. 9

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").)

A POSITA would have understood Beasley to disclose, in audio bypass mode (discussed above, for example, in Section VII.D.1), providing a mixed signal that includes a background audio signal, an audio signal, and a noise cancellation signal. (OTH-A, Exp. Decl., ¶ 307.) 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." (PA-F, ¶ [0051].) For example, "[t]his can be accomplished using inverters, compensation filters, and other known techniques." (*Id.*) As such, a POSITA would have understood that the functionality of Beasley's inverter 708 could be implemented instead by Andrea's noise reduction functionality. (OTH-A, Exp. Decl., ¶ 308.) Thus, a POSITA 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. (*Id.*)

Accordingly, when Beasley's "audio bypass mode" is implemented in the combined Victorian-Dijkstra-Beasley device, a POSITA would have considered it predictable to modify

Victorian's 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. (*Id.*, ¶ 309.)

F. SNQ 6: Claims 1-2, 8-9 and 11 are Obvious in View of Victorian in Combination with Van Doorn and Beasley

As discussed in Section VII.A.2.b, 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. This SNQ 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, Requesters respectfully submit 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

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 VII.D, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section VII.D.2.a.
1[c]	*See supra* Section VII.D.2.c.
1[d]	*See supra* Section VII.D.2.d.
1[e]	*See supra* Section VII.D.2.d.
1[f]	*See supra* Section VII.D.2.e.
1[g]	*See supra* Section VII.D.2.f.
1[h]	*See supra* Section VII.D.2.g.
1[i]	*See supra* Section VII.D.2.h.
1[j]	*See supra* Section VII.D.2.i.
1[k]	*See supra* Section VII.D.2.j.

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 VII.D.2.b. Additionally, for the reasons discussed in Section VII.C.1 a POSITA would have understood Van Doorn's microphone having a sound port 56 to disclose a "microphone port." (OTH-A, Exp. Decl., ¶ 312.)

A POSITA 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. (*Id*, ¶ 313). Indeed, a POSITA 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 (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]). (OTH-A, Exp. Decl., ¶ 313). A POSITA 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. (*Id.*, ¶ 314 (citing PA-D, ¶ [0006]).) A POSITA 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). (OTH-A, Exp. Decl., ¶ 315.)

Accordingly, a POSITA 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"). (*Id.*, ¶ 316.)

2. Dependent Claims 2, 8-9, and 11

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 VII.D, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section VII.D.3.
8	*See supra* Section VII.D.4.
9	*See supra* Section VII.D.5.
11	*See supra* Section VII.D.6.

Accordingly, for at least the same reasons, these claims would have been obvious over Victorian, Van Doorn, and Beasley.

G. SNQ 7: Claim 10 is Obvious in View of Victorian in Combination with Van Doorn, Beasley, and Andrea

Requesters respectfully submit 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."

It would have been obvious for a POSITA to incorporate Andrea with the Victorian-Van Doorn-Beasley device for the same reasons and in the same way as discussed above in Section

VII.E. Accordingly, for at least the same reasons, claim 10 would have been obvious over Victorian, Van Doorn, Beasley, and Andrea. (OTH-A, Exp. Decl., ¶ 320.)

H. SNQ 8: Claims 1 and 11 are Obvious in view of Dijkstra in combination with Boersma

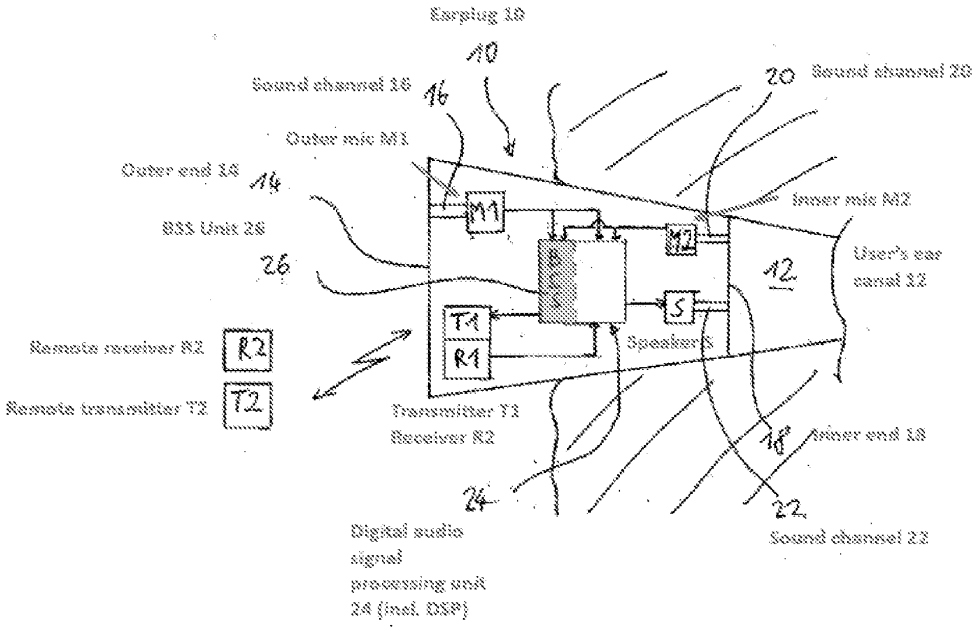
As discussed in Section VII.A.2.b, under the broadest reasonable interpretation of claim 1, any opening associated with a microphone "through which measurements are taken" qualifies as a "microphone port" within the meaning of the '666 patent. This SNQ addresses the circumstance in which Patent Owner 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, Requesters respectfully submit 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 POSITA Would Have Been Motivated to Combine Dijkstra and Boersma

As explained further below, a POSITA would have been motivated to combine the teachings of Dijkstra and Boersma and would have had a reasonable expectation of success in doing so. (OTH-A, Exp. Decl., ¶¶ 322-333.)

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. (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)

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].)

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.*)

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).

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.)

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).

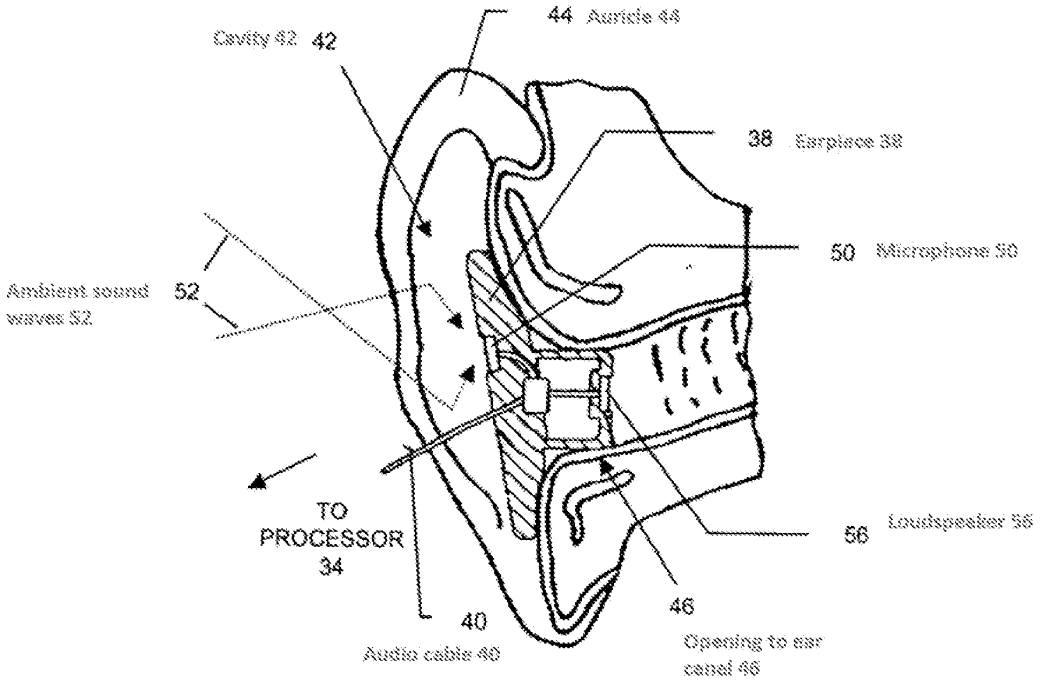


FIG. 2

Boersma, FIG. 2 (annotated)

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).

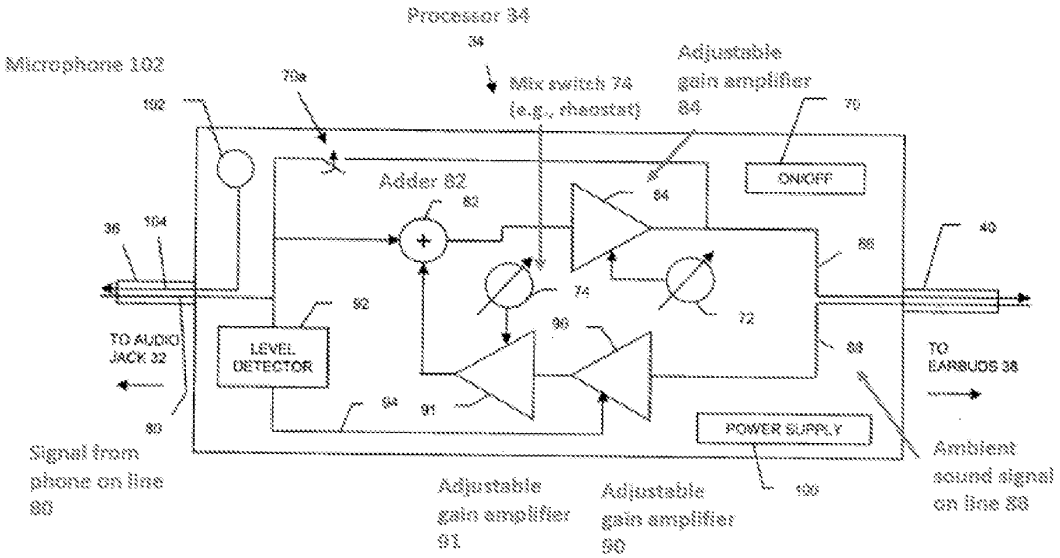


FIG. 4

Boersma, FIG. 4 (annotated)

A POSITA would have recognized that Boersma's and Dijkstra's teachings complement each other in a manner that would have motivated a POSITA 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. (OTH-A, Exp. Decl., ¶ 330.)

For example, a POSITA 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 POSITA 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. (*Id.*, ¶ 331.) A POSITA 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 in some circumstances. (PA-H, 1:43-50.)

Accordingly, a POSITA 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 passthrough and mixing). (OTH-A, Exp. Decl., ¶ 332.)

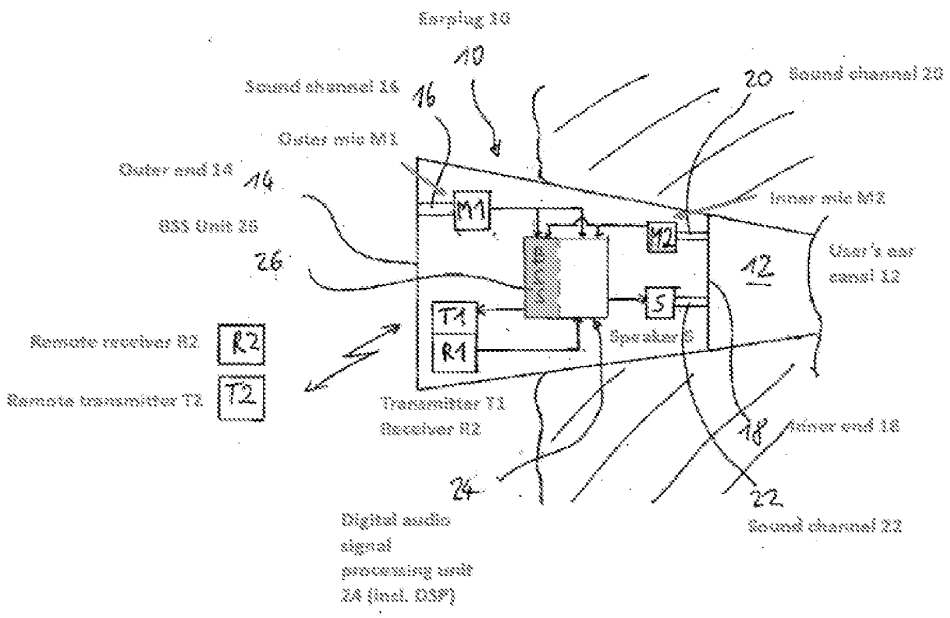
There is ample evidence that a POSITA would have had a reasonable expectation of success in making the proposed combination. (*Id.*, ¶ 333.) For example, Boersma itself suggests implementing processor 34 with digital circuitry in a digital design, and implementing it within the housing of an earpiece. (PA-H, 7:47-61, 9:9-15.) Moreover, it was well-known to use DSPs for mixing audio signals with ambient sound. (OTH-A, Exp. Decl., ¶ 333.) (discussing U.S. Patent Application Publication No. 2003/0035551A1 ("Light") (PA-S)).)[22] Furthermore, Boersma discloses that a POSITA 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. (OTH-A, Exp. Decl., ¶ 333.)

2. Claim 1

a. 1[pre] "1. An earphone comprising:"

To the extent the preamble is limiting, Dijkstra discloses or suggests an earphone. (OTH-A, Exp. Decl., ¶¶ 334-336.) For example, Dijkstra discloses earplug 10 that is inserted into the user's ear. (PA-C, ¶¶ [0013], [0026]-[0030], [0032], [0034]-[0043], FIGs. 1-6.)

[22] Here, Light (PA-S) is being used as evidence of a POSITA's knowledge and the state of the art; it is not part of the combination in SNQ 8.



Dijkstra, FIG. 1 (annotated)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 335.) 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; OTH-A, Exp. Decl., ¶ 335.)

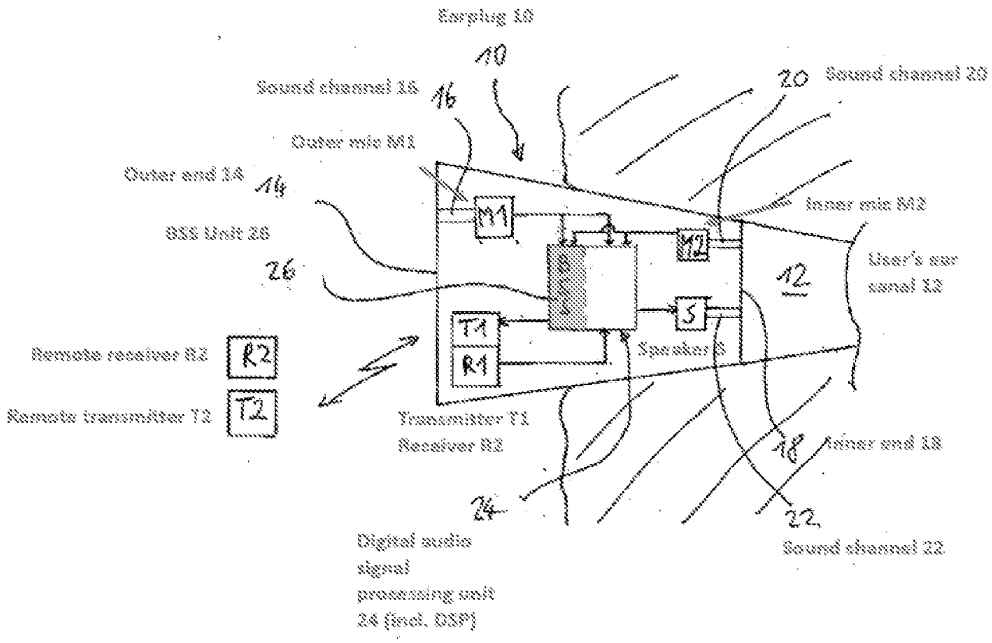
Thus, a POSITA would have understood Dijkstra to disclose or suggest an earplug ("an earphone"). (OTH-A, Exp. Decl., ¶ 336.)

b. **<u>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;"</u>**

<u>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</u>

microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;"

Dijkstra discloses or suggests these features. (OTH-A, Exp. Decl., ¶¶ 337-340.) 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)

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.*)

A POSITA 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 VII.B.1.

Accordingly, a POSITA 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 POSITA 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"). (OTH-A, Exp. Decl., ¶ 340.)

c. 1[c] "a speaker configured to play an audio signal;"

Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 341-342). 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. (PA-C, ¶¶ [0017], [0029]-[0030], [0032], [0034].) A POSITA would have understood that a remote audio signal is an "audio signal." (OTH-A, Exp. Decl., ¶ 341). 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).)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 342.)

d. 1[d] "a memory that stores instructions; and"

Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 343-344.) 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." (PA-C, ¶ [0029].) A POSITA would have understood that a "program" refers to software instructions. (OTH-A, Exp. Decl., ¶ 343.)

Accordingly, a POSITA 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"). (*Id.*, ¶ 344.)

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:"

Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 345-346.) Dijkstra discloses that the earplug 10 includes an audio signal processing unit including a digital signal processor that has a program memory component. (PA-C, ¶ [0029].) A POSITA would have understood that a processor is a component specifically designed to execute instructions stored in memory. (OTH-A, Exp. Decl., ¶ 345.) Dijkstra explains that the audio signal processing unit receives signals from microphones M1 and M2 and provides them to the speaker. (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.)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 346.)

f. 1[f] "receiving a first microphone signal from the first microphone;"

Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶ 347.) Dijkstra explains that the audio signal processing unit receives signals from outer microphone M1. (PA-C, ¶¶ [0029], [0031], [0034], FIG. 1.) Thus, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 347.)

g. 1[g] "receiving a second microphone signal from the second microphone;"

Dijkstra discloses or suggests this feature. (OTH-A, Exp. Decl., ¶ 348.) Dijkstra explains that the audio signal processing unit receives signals from inner microphone M2. (PA-C, ¶¶ [0029], [0031], [0034], FIG. 1.) Thus, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 348.)

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;"

Dijkstra in combination with Boersma discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 349-353.)

Boersma discloses that the mix switch 74 provides the user with control over the mix of ambient sound signal and the audio signal. (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. (*Id.*, 8:33-46.)

As reflected in the annotated Figure 4 below, a POSITA would have understood that applying a gain to the signal received from microphone 50 generates an ambient sound signal. (OTH-A, Exp. Decl., ¶ 351.)

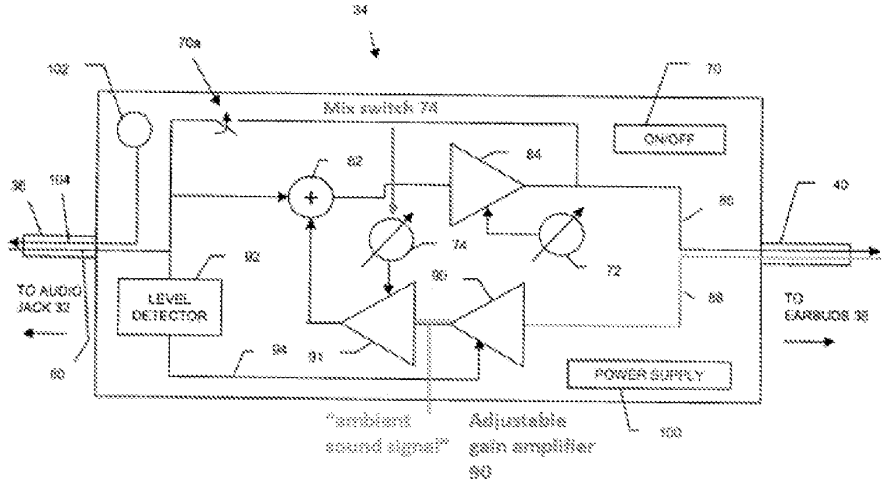


FIG. 4

Boersma, FIG. 4 (annotated)

This is consistent with how the '666 patent discloses this feature. As explained above in §VII.A.2.f, a POSITA 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. (OTH-A, Exp. Decl., ¶ 352.) 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) (emphasis added). Thus, a POSITA would have understood that applying a gain to a signal from the ambient sound signal encompasses "generating an ambient sound signal." (OTH-A, Exp. Decl., ¶ 352.)

As discussed above in Section VII.H.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 POSITA

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 …"). (*Id.*, ¶ 353.)

i. 1[i] "applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;"

Dijkstra in combination with Boersma discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 354-356.) 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.)

As reflected in annotated Figure 4 below, a POSITA 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); OTH-A, Exp. Decl., ¶ 355.)

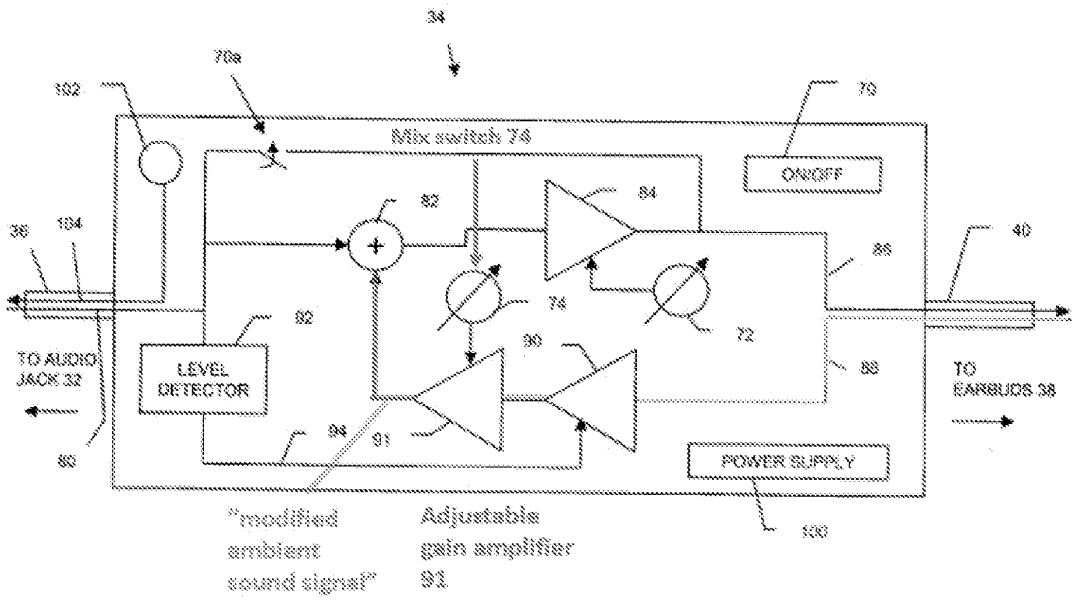


FIG. 4

Boersma, FIG. 4 (annotated)

Accordingly, a POSITA 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"). (*Id.*, ¶ 356.)

j. 1[j] "mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and"

Dijkstra in combination with Boersma discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 357-358.) Boersma discloses that adder 82 "functions to combine the ambient sound signal with the audio signal from the mobile phone 10." (PA-H, 8:19-21.. As reflected in the annotated FIG. 4 below, a POSITA 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); OTH-A, Exp. Decl., ¶ 357.)

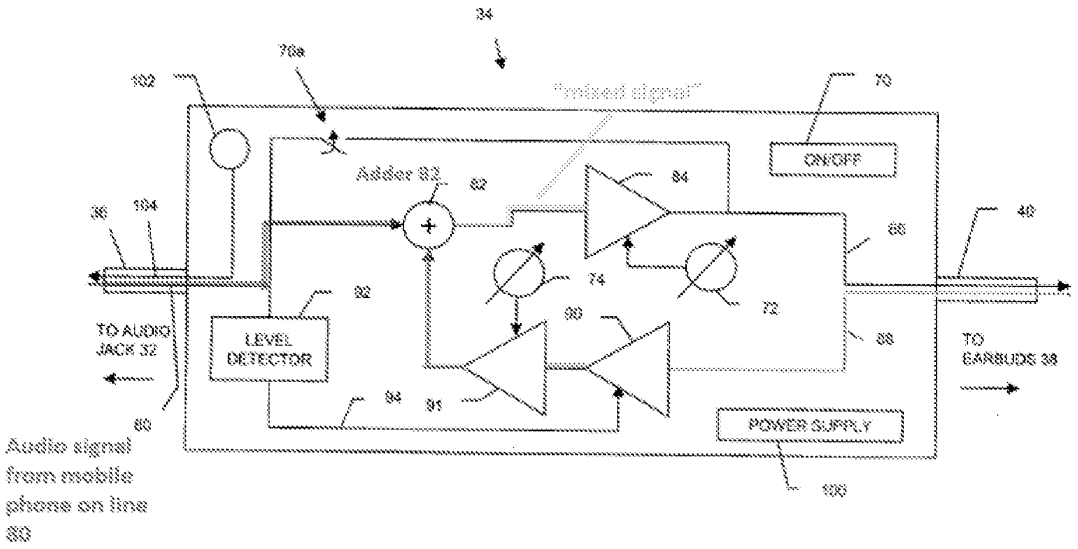


FIG. 4

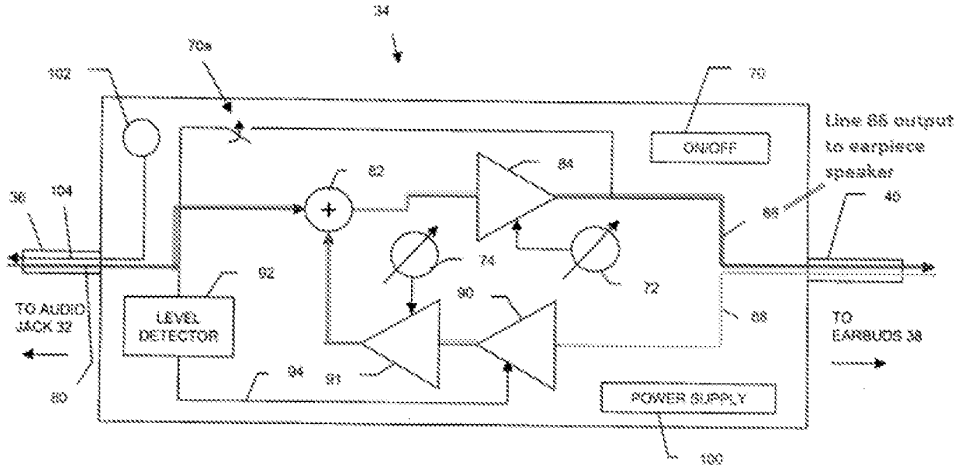
Boersma, FIG. 4 (annotated)

Accordingly, a POSITA 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"). (*Id.*, ¶ 358.)

k. 1[k] "sending the mixed audio signal to the speaker."

Dijkstra in combination with Boersma discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 359-360.) Boersma discloses that the output of the amplifier 84 is provided to the

earpieces 38 on line 86 to be output through the speaker. (PA-H, 8:6-7, 2:21-26, 2:65-3:5, 6:62-63.)



Boersma, FIG. 4 (annotated)

Accordingly, a POSITA 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"). (OTH-A, Exp. Decl., ¶ 360.)

3. Claim 11

a. "11. The earphone according to claim 1, wherein the earphone includes a sealing section."

Dijkstra in combination with Boersma discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 361-362.) Dijkstra discloses a "hearing protection device" that attenuates the ambient sound that reaches the inner microphone. (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; OTH-A, Exp. Decl., ¶ 361.)

Accordingly, a POSITA 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"). (*Id.*, ¶ 362.)

I. SNQ 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

Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 363-371.) 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.)

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. (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. (*Id.*, ¶¶ [0037], [0061], claims 18, 19, 30, 31.)

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].)

A POSITA 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 POSITA 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. (OTH-A, Exp. Decl., ¶ 367.)

A POSITA 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. (OTH-A, Exp. Decl., ¶ 368.) 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 POSITA 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. (*Id.*) 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.

Thus, a POSITA 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). (OTH-A, Exp. Decl., ¶ 369.)

The Federal Circuit has held that automating prior art manual processes is obvious. (*Cimline, Inc. v. Crafco, Inc.*, 413 F. App'x 240, 246–47 (Fed. Cir. 2011) ("The automation of mechanical equipment once operated manually is commonplace and reasonably obvious to one of ordinary skill.") (citing *Leapfrog Enters. v. Fisher–Price, Inc.*, 485 F.3d 1157, 1161 (Fed.Cir.2007) ("Accommodating a prior art mechanical device ... to modern electronics would have been reasonably obvious to one of ordinary skill ...." and "has been commonplace in recent years."))).)

Moreover, the Federal Circuit has held that "an implicit motivation to combine exists not only when a suggestion may be gleaned from the prior art as a whole, but when the 'improvement' is technology-independent and the combination of references results in a product or process that is more desirable, for example because it is stronger, cheaper, cleaner, faster, lighter, smaller, more durable, or more efficient." (*DyStar*, 464 F.3d at 1368.) The Federal Circuit noted that "[b]ecause the desire to enhance commercial opportunities by improving a product or process is universal—and even common-sensical—we have held that there exists in these situations a motivation to combine prior art references even absent any hint of suggestion in the references themselves. In such situations, the proper question is whether the ordinary artisan possesses knowledge and skills rendering him capable of combining the prior art references." (*Id.*)

There is ample evidence that a POSITA would have had a reasonable expectation of success in making the combination. (OTH-A, Exp. Decl., ¶ 370.) For example, Beasley itself

discloses implementing the audio bypass feature using a DSP and programming it to use speech recognition. (PA-F, ¶¶ [0054], [0061].)

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 POSITA 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..."). (OTH-A, Exp. Decl., ¶ 371.)

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."

Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 372-375.) As discussed in Section VII.I.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 POSITA would have understood that each of the events detected by these techniques comprise the user's voice. (OTH-A, Exp. Decl., ¶ 372.) 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 VII.I.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. (OTH-A, Exp. Decl., ¶ 372.)

Accordingly, a POSITA 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…"). (*Id.*, ¶ 373.)

Although Requesters do not concede it, 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, a POSITA would have nonetheless found each of these features to be obvious in view of the teachings of Beasley. (*Id.*, ¶ 374.) 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." (PA-F, ¶ [0061].) However, a POSITA 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. (OTH-A, Exp. Decl., ¶ 374.) As such, a POSITA 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. (*Id.*) 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. (PA-F, claims 19 ("[W]herein the means for automatically engaging comprises voice recognition, speech recognition, and background audio level threshold detection.").)

Accordingly, a POSITA 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 …."). (OTH-A, Exp. Decl., ¶ 375.)

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."

Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 376-378.) As explained above in Sections VII.I.1.a and VII.I.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.

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 POSITA 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"). (OTH-A, Exp. Decl., ¶ 378.)

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."

Dijkstra in combination with Boersma and Beasley discloses or suggests this feature. (OTH-A, Exp. Decl., ¶¶ 379-381.) As explained above in Sections VII.I.1.a and VII.I.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. (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].)

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 POSITA 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"). (OTH-A, Exp. Decl., ¶ 381.)

J. SNQ 10: Claims 1 and 11 are Obvious in view of Dijkstra, Boersma, and Van Doorn

As discussed in Section VII.A.2.b, 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. This SNQ 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, Requesters respectfully submit 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

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 VII.H, as referenced in the table provided below.

Claim 1	Reference
1[pre]	*See supra* Section VII.H.2.a.
1[c]	*See supra* Section VII.H.2.c.
1[d]	*See supra* Section VII.H.2.d.
1[e]	*See supra* Section VII.H.2.e.
1[f]	*See supra* Section VII.H.2.f.
1[g]	*See supra* Section VII.H.2.g.
1[h]	*See supra* Section VII.H.2.h.
1[i]	*See supra* Section VII.H.2.i.
1[j]	*See supra* Section VII.H.2.j.
1[k]	*See supra* Section VII.H.2.k.

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 VII.H.2.b.

Additionally, for the reasons discussed in Section VII.C.1 a POSITA would have understood Van Doorn's microphone having a sound port 56 to disclose a "microphone port." (OTH-A, Exp. Decl., ¶ 385.)

A POSITA 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. (*Id.*, ¶ 386.) Indeed, a POSITA 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 (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]; OTH-A, Exp. Decl., ¶ 386.) A POSITA 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. (*Id.*, ¶ 387 (citing PA-D, ¶¶ [0002], [0020]).) A POSITA 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). (*Id.*, ¶ 388). Accordingly, a POSITA 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"). (*Id.*)

2. Claim 11

With respect to dependent claim 11, Dijkstra in combination with Boersma meets it in the same way discussed above in Section VII.H.3.

Accordingly, for at least the same reasons, this claim would have been obvious over Dijkstra, Boersma, and Van Doorn. (OTH-A, Exp. Decl., ¶ 390.)

K. SNQ 11: Claims 2 and 8-9 are Obvious in view of Dijkstra, Boersma, Van Doorn, and Beasley

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 VII.I, as referenced in the table provided below.

Claim	Reference
2	*See supra* Section VII.I.1.
8	*See supra* Section VII.I.2
9	*See supra* Section VII.I.3

Accordingly, for at least the same reasons, these claims would have been obvious over Dijkstra, Boersma, Van Doorn, and Beasley. (OTH-A, Exp. Decl., ¶ 392)

VIII. CONCLUSION

In view of the prior art presented above, claims 1-11 of the '666 patent are not patentable. The prior art discloses, teaches, suggests, or otherwise renders obvious the subject matter of the '666 patent in such a manner that raise substantial new questions of patentability for claims 1-11. Accordingly, Requesters requests that the Patent Office grant this Request and initiate reexamination with special dispatch.

Respectfully submitted under 37 C.F.R. § 1.34,

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IX. CERTIFICATE OF SERVICE

The undersigned hereby certifies that a copy of this Request for *Ex Parte* Reexamination of U.S. Patent No. 11,244,666, together with all exhibits and other documents filed therewith, has been served via First Class Mail, postage prepaid, on February 24, 2023, upon Patent Owner via counsel of record at the United States Patent & Trademark Office under 37 C.F.R. § 1.33, at the following address:

150139 - Akerman LLP
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Date: February 24, 2023

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Exhibit PA-N

United States Patent [19]

Baumhauer, Jr. et al.

[11] **Patent Number: 4,533,795**

[45] **Date of Patent: Aug. 6, 1985**

[54] **INTEGRATED ELECTROACOUSTIC TRANSDUCER**

[75] Inventors: **John C. Baumhauer, Jr.; Harold J. Hershey,** both of Indianapolis, Ind.; **Tommy** L. **Poteat,** Bridgewater, **N.J.**

[73] Assignees: **American Telephone and Telegraph,** New York, **N.Y.; AT&T Bell Laboratories,** Murray **Hill, N.J.**

[21] Appl. No.: **511,640**

[22] Filed: **Jul. 7, 1983**

[51] **Int. Cl.**[3] **H04R 7/18**

[52] **U.S. Cl.** ..... **179/111** E; 29/25.35

[58] **Field of Search** ..... 179/111 E, 111 R; 29/594, 592 E, 25.35

[56] **References Cited**

U.S. PATENT DOCUMENTS

3,118,022	1/1964	Sessler et al.	179/111
4,225,755	9/1980	Block	179/111 E
4,321,432	3/1982	Matsutani	179/111 E

FOREIGN PATENT DOCUMENTS

84608	10/1982	European Pat. Off.	179/111 R
2094098	9/1982	United Kingdom	179/111 R

OTHER PUBLICATIONS

"On the Acoustical Resistance Due to Viscous Losses in the Air Gap of Electrostatic Transducers", Z. Skvor, *Acustica,* vol. 19, pp. 295-299, (1967/1968).

"Optimization of a Ridge Backplate for Electret Microphones", H. S. Madsen, *Journal of the Acoustical Society of America,* vol. 53, No. 6, pp. 1616-1619, (1973).

Primary Examiner-Gene Z. Rubinson

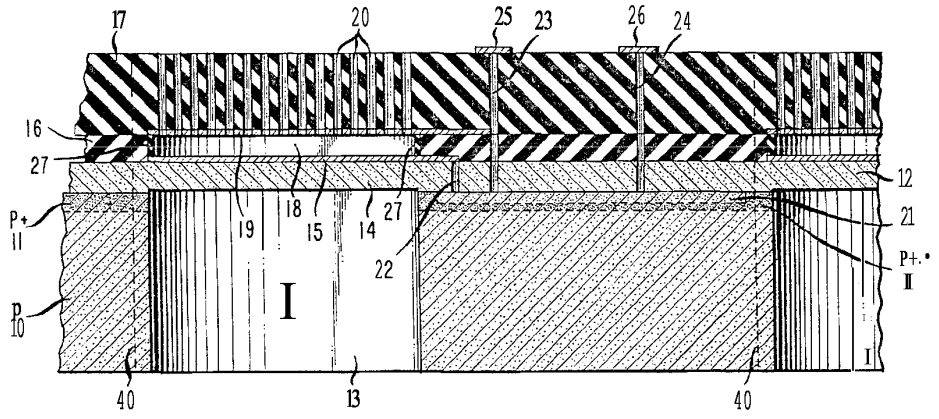
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[57] **ABSTRACT**

An electroacoustic transducer, primarily in the form of a capacitive microphone, for incorporation into a semiconductor substrate. The vibrating element comprises a largely nontensioned diaphragm, such as an epitaxial layer formed on the semiconductor substrate, so as to greatly reduce its mechanical stiffness. The substrate is etched away in the desired area to define the diaphragm and form an acoustic cavity. A continuous array of microscopic holes is formed in the backplate to cut down the lateral flow of air in the gap between capacitor electrodes. Narrow gaps made possible by the hole array allow low voltage diaphragm biasing. In at least one embodiment, the acoustic input can be provided through the air hole array. An acoustic port may be added to alter the frequency response of the device, and a back closure provided to act as a rear acoustic cavity and an EMI shield.

26 Claims, 6 Drawing Figures



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FIG. 1

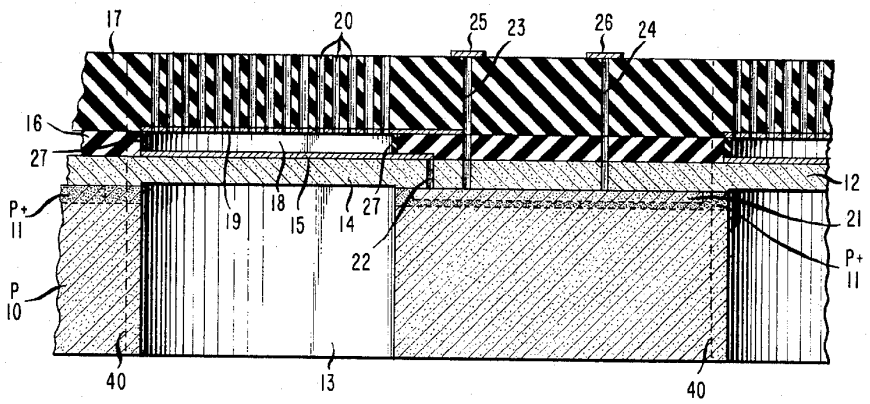


FIG. 2

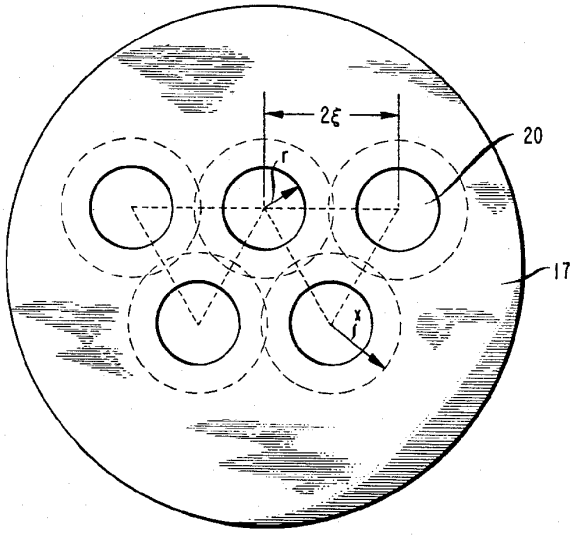


FIG. 3

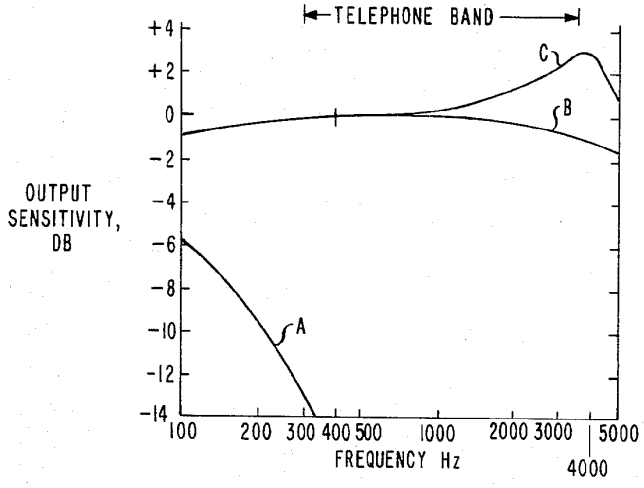

TELEPHONE BAND
+4
+2
0
-2
-4
-6
-8
-10
-12
-14
OUTPUT SENSITIVITY, DB
A
B
C
100
200
300
400
500
1000
2000
3000
5000
4000
FREQUENCY Hz

FIG. 4

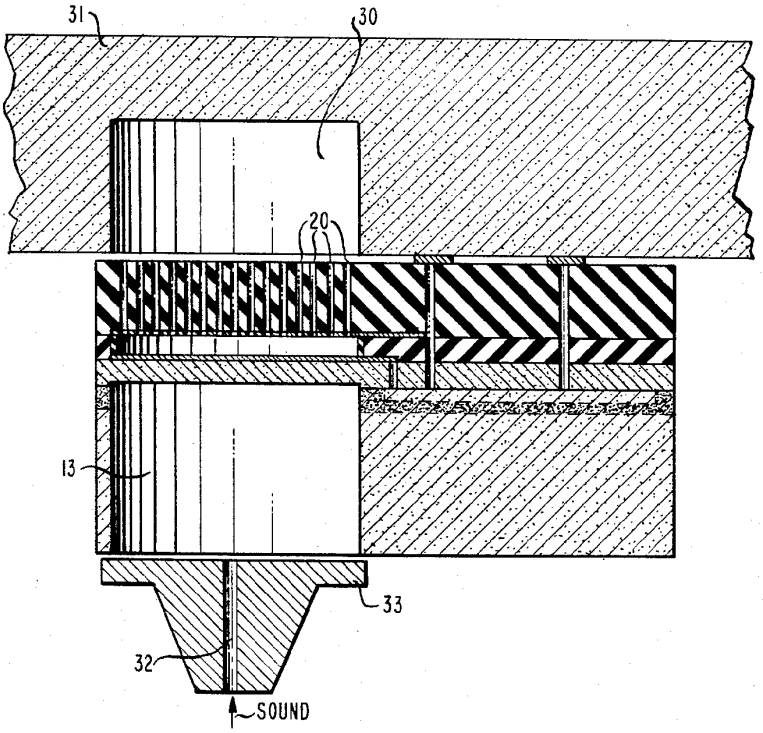

31
30
20
13
33
32
SOUND

FIG. 5

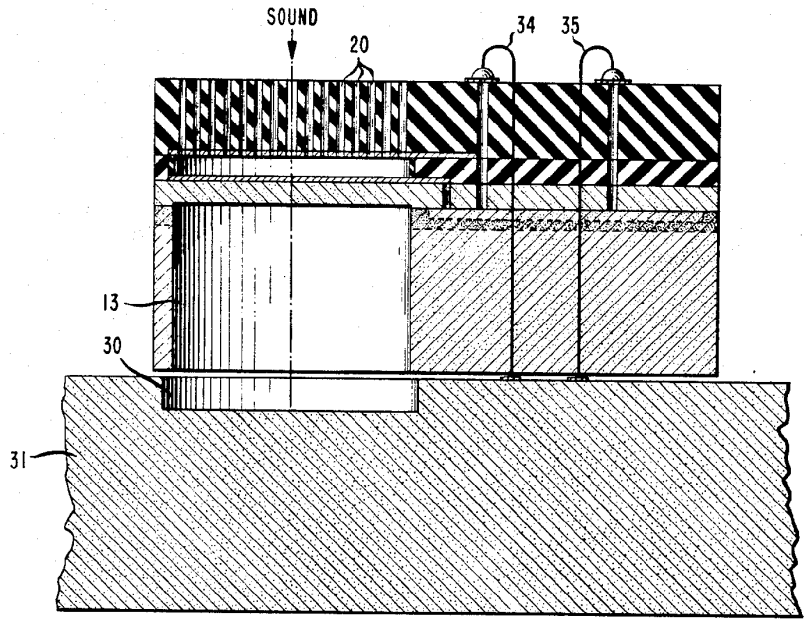
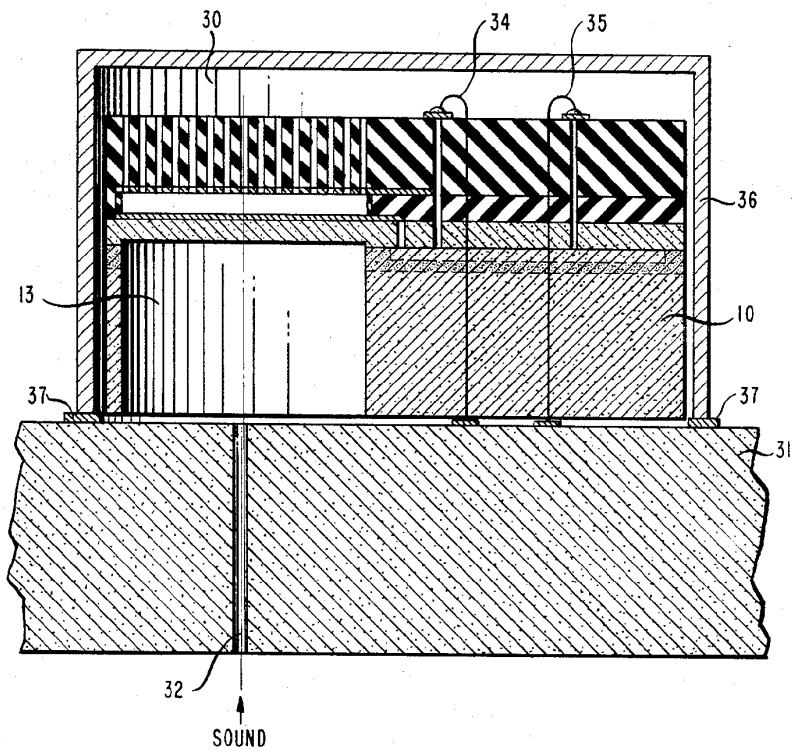


FIG. 6



INTEGRATED ELECTROACOUSTIC TRANSDUCER

BACKGROUND OF THE INVENTION

This invention relates to electroacoustic transducers, such as microphones, and in particular to a structure which is incorporated into a semiconductor substrate.

Demand is growing for electroacoustic transducers which may be formed as part of a semiconductor integrated circuit. These transducers may include, for example, microphones incorporated into the circuitry of telecommunications and audio recording equipment, hearing aid microphones and receivers, or miniature speakers. In the case of microphones, electrostatic device technology presently in widespread use generally takes the form of a metalized polymeric foil (which may be charged) supported over a metalized backplate or stationary structure so as to form a variable capacitor responsive to voiceband frequencies. While adequate, such devices are relatively large, discrete components which cannot be integrated into the semiconductor integrated circuitry with which they are used.

Recently, such an integrated microphone structure and a method of manufacture were proposed. (See U.S. Patent application of I. J. Busch-Vishniac et al., Ser. No. 469,410, filed Feb. 24, 1983 and assigned to Bell Telephone Laboratories, which is incorporated by reference herein.) Briefly, the microphone included a tensioned membrane formed from a thinned portion of a thicker semiconductor substrate. The membrane had an area and thickness such that it vibrated in response to incident sound waves. A pair of electrodes formed a capacitor, with one of the electrodes vibrating with the membrane to vary the capacitance when a biasing voltage was applied and produce an electrical equivalent to the acoustic signal. It has also been suggested that an integrated capacitive microphone can include an insulating layer with fixed charge for providing a built-in diaphragm bias for the device. (See U.S. Patent application of W. S. Lindenberger, T. L. Poteat, and J. E. West, Case 2-2-24, filed on an even date herewith and assigned to Bell Telephone Laboratories.)

While such structures offer considerable promise for the replacement of the distinct microphones now in use, several problems and considerations remain in the commercial realization of an integrated microphone. Foremost, it is desirable to make the area of the vibrating element as small as possible to reduce cost. However, a small area tends to cause a drop in sonic force on the diaphragm element, thereby lowering the sensitivity of the device. Further, smaller area diaphragms produce a smaller device capacitance which in turn tends to increase the noise associated with on-chip circuitry coupled to the device and also tends to further decrease the integrated microphone sensitivity through capacitance divider action. In order to alleviate such effects of reduced area (i.e., reduced signal-to-noise ratio), it is desirable to reduce the stiffness of the diaphragm.

The above-noted effects of reduced diaphragm area may also be compensated for by a reduction in the thickness of the air gap between capacitor electrodes. We have found, however, that for air gaps below approximately 1.5 μm with other dimensions optimized for certain telephone applications, the electrical output frequency response of a microphone with a tensioned diaphragm had a tendency to fall at an unacceptable rate with frequency when utilizing acoustic venting means, common in commercially available devices, comprising 4–20 holes around the periphery of the stationary electrode and backplate. That is, the devices would be overdamped, even at the critical 300–3,500 Hz portion of the audioband which is transmitted in telephone equipment. Specifically, with an air gap of 0.25 μm which yielded the optimum signal-to-noise ratio, the response fell more than 20 dB across the telephone frequency band indicating severe overdamping. Sensitivity levels were also inoperably low. It is, therefore, also desirable to provide some acoustic venting means which will permit reduced area and produce an acceptable output signal at telephone band frequencies or other frequency bands of interest.

A reduction in the air gap thickness will also have another beneficial effect, which is to reduce the external dc voltage level needed to bias the diaphragm. This would provide an alternative to the requirement of a built-in diaphragm bias as suggested in the application of Lindenberger, previously cited.

In addition, a silicon integrated microphone will generally have a nonrising output response as a function of frequency in the audio bandwidth. In some applications, it may be desirable to tailor the response to provide a peak at a certain frequency by means of an appropriately shaped acoustic port and coupling cavity. Further, the microphone chip may, under certain circumstances, be subject to high electromagnetic interference (EMI), and so some shielding means may be needed. In the design of an integrated microphone, therefore, it is desirable to provide these functions with a minimum number of piece parts.

It is therefore an object of the invention to provide an integrated electroacoustic transducer with a small diaphragm area which still provides an acceptable frequency response and signal-to-noise ratio. It is a further object of the invention to provide an electroacoustic transducer which can be operated at a low dc bias. It is a still further object of the invention to provide acoustical interconnection and tuning means, and EMI shielding means, for an integrated electroacoustic transducer.

SUMMARY OF THE INVENTION

These and other objects are achieved in accordance with the invention which is an electroacoustic transducer formed within a semiconductor substrate. The transducer comprises a diaphragm which vibrates in response to an input signal at audio and/or ultrasonic frequencies, and a pair of electrodes placed with respect to said diaphragm so that the electric field between the electrodes varies in relation to the vibrating diaphragm to permit conversion between electrical and acoustic signals. In accordance with one aspect of the invention, the diaphragm is essentially nontensioned and has a mechanical bending moment stiffness which allows it to vibrate at audioband and/or ultrasonic frequencies.

In accordance with a further aspect of the invention, the transducer comprises an air gap formed between the electrodes and acoustic venting means comprising a continuous array of small holes extending through one of the electrodes to the air gap, where the array covers essentially the entire area of the electrode co-extensive with the diaphragm.

In accordance with a further aspect of the invention, a front acoustic cavity is defined by a portion of a semiconductor substrate adjacent to the diaphragm, an air gap is formed between the electrodes, and a back acous-

tic cavity is formed adjacent to one of the electrodes and defined by a member which also shields the transducer from electromagnetic interference. Acoustic venting means is provided by an array of holes connecting the air gap to the back acoustic cavity. An acoustic port is defined by an element adjacent to the front acoustic cavity, with the size of the port being such as to produce a desired frequency response characteristic for the transducer.

BRIEF DESCRIPTION OF THE DRAWING

These and other features of the invention are delineated in detail in the following description. In the drawing:

FIG. **1** is a cross-sectional view of a microphone in accordance with one embodiment of the invention;

FIG. **2** is a top view of a portion of the device illustrated in FIG. **1**;

FIG. **3** is an illustration based on computer modeling of typical frequency response curves predicted for microphones fabricated in accordance with the invention as compared with predictions of a prior art type device where the reference level is the 400 Hz response level of curve B;

FIG. **4** is a cross-sectional view of a microphone in accordance with a further embodiment of the invention;

FIG. **5** is a cross-sectional view of a microphone in accordance with a still further embodiment of the invention; and

FIG. **6** is a cross-sectional view of a microphone in accordance with a still further embodiment of the invention.

It will be appreciated that for purposes of illustration, these figures are not necessarily drawn to scale.

DETAILED DESCRIPTION

Many of the basic features of the invention will be described with reference to the particular embodiment illustrated in FIG. **1**. It will be appreciated that the Figure shows only a small portion of a semiconductor substrate which in this example includes a great many other identical, integrated, electronic transducer devices which are separated along saw lines, **40**, following batch processing and testing.

The semiconductor substrate, **10**, is a standard p-type silicon wafer having a thickness of approximately 300–600 μm (12–24 mils) and a <100> orientation. It will be appreciated that n-type wafers and other crystal orientations may be employed. A surface layer, **11**, of p+ type is formed over at least portions of the substrate, for example, by an implantation of boron impurities to a depth of approximately 0.2 μm. The impurity concentration of this region is typically $5\times10^{+19}/cm^3$.

Formed over the semiconductor substrate is a monocrystalline silicon epitaxial layer, **12**. An appropriate area of the semiconductor substrate is etched to form a front acoustic cavity, **13**, so that the portion, **14**, of the epitaxial layer over the cavity forms the vibrating diaphragm element of the microphone. The characteristics of the diaphragm are discussed in more detail below.

Formed on the surface of the diaphragm is a metal layer, **15**, such as Ti-Au, which comprises one electrode of a capacitor. This layer is typically 0.1–0.5 μm thick. Formed over the epitaxial layer is a spacer layer, **16**, which is typically polycrystalline silicon and can be formed by chemical vapor deposition. The thickness of the layer in this example is 0.60 μm and will generally be in the range 0.1 μm–4.0 μm. A backplate layer, **17**, is formed on the spacer layer with a portion (hereinafter the backplate) over the area of the diaphragm to establish an air gap, **18**. The backplate may comprise a layer of BN or Si_3N_4 but could be any insulating layer or layers. The thickness of the layer is approximately 12 μm and will generally be in the range 6 μm–30 μm. A metal or other conducting layer, **19**, is formed on the surface of the backplate facing the air gap and comprises the second electrode of the capacitor. It can be the same material and thickness as the first electrode, **15**.

Formed through the backplate and electrode, **19**, to the air gap is a continuous array of holes, **20**, for acoustic venting. This feature will be discussed in more detail below.

The electronics for driving the device may be fabricated in the adjacent area of the semiconductor substrate designated **21**, and electrical contact to the electrodes may be provided through via holes, **22** and **23**. Contact to the outside may be provided through via holes, **23** and **24**, and contact pads, **25** and **26**, formed on the surface of the backplate layer. If desired, additional via holes (not shown) could be provided from the electronics to the outside to establish separate dc supply, signal and ground leads.

In operation, a dc bias is applied to electrodes, **15** and **19**, and an acoustic signal is made incident on diaphragm, **14**, through the front acoustic cavity. The signal causes the diaphragm to vibrate, thus varying the spacing between the electrodes and the capacitance of the capacitor. This change in capacitance can be detected as a change in voltage across some load element, integrated into area, **21**, such as a second capacitance and parallel resistance (not shown), and an electrical equivalent to the acoustic signal is produced. The array of holes, **20**, permits escape of air in the gap, **18**, so that air stiffness in the gap is not a significant factor in the diaphragm motion. Desirably, the amplitude of the output signal as a function of frequency will be as shown in curve B in FIG. **3**, where the signal (at constant sound pressure amplitude) is essentially flat or falls no more than 3 dB over the portion of the audioband transmitted in telephone applications (0.3–3.5 KHz). In accordance with a feature of the invention, the vibrating diaphragm is essentially nontensioned so that the stiffness of the diaphragm is dominated by the mechanical bending moment. This can be contrasted with previous designs for integrated microphones where the diaphragm was a thinned portion of the semiconductor substrate and had a stiffness which was dominated by tensile stress resulting from a heavy boron impurity concentration.

The advantage of using the nontensioned diaphragm results from the fact that the sensitivity (e) of the microphone depends upon the following parameters:

$$e \propto \left[\frac{VA}{k_m + k_a}\right] / h_e \qquad (1)$$

where V is the dc bias voltage across the electrodes, A is the area of the diaphragm, k_m is the effective mechanical lumped parameter (piston-like) stiffness of the diaphragm associated with the diaphragm's linearly averaged deflection, k_a is the stiffness of the rear acoustic cavity (shown, for example, in FIG. **4**), and h_e is the distance between capacitor electrodes, linearly aver-

aged over A, when the bias is applied to the electrodes. Thus, for the same sensitivity, the area, A, of the diaphragm may be reduced by decreasing the stiffness k_m of the diaphragm (k_a contributes less than $\frac{1}{3}$ of the total stiffness acting on the diaphragm). We have found that when the stiffness is due primarily to tensile stress, a practical lower limit exists for reducing the membrane stiffness. In the present invention, however, k_m is significantly lowered in order that the diaphragm area may be reduced by allowing the mechanical bending moment of the diaphragm to be the dominant mechanical stiffness component. Further, when the stiffness is due to the mechanical bending moment, the thickness of the diaphragm can be significantly increased over a tensioned membrane and still produce a significantly lower stiffness and diaphragm area.

It will be appreciated that effective stiffness k_m, as used in this application, is the stiffness of a piston-like model where the deflection of the piston is equal to the diaphragm deflection averaged over the area of the diaphragm. For a nontensioned diaphragm, this stiffness is given approximately by the relationship:

$$k_m \approx \frac{192\pi^2 D_p}{A} \tag{2}$$

where D_p is the flexural rigidity of the diaphragm.

In this example, the diaphragm thickness is 3.0 μm, the diaphragm is circular with a radius of 700 μm, k_m and k_a are approximately 625 and 175 N/m respectively. The biasing voltage across the electrodes is 1.3 volts. The resulting sensitivity is approximately −49 dB relative to 1 V/Pa at 400 Hz (excluding any signal voltage amplifiers that may be integrated on the chip microphone). For most nontensioned diaphragm applications, it is expected that the diaphragm thickness will range from 1 μm–5 μm, and area will range from 0.4 mm^2–5 mm^2 to achieve proper sensitivity. Stiffness, k_m, of the nontensioned diaphragm is desirably less than 5500 N/m.

While in this example the diaphragm was formed from a single crystal epitaxial layer, **12**, other types of nontensioned diaphragms might be employed. However, use of an epitaxial layer provides many advantages in terms of processing. For example, diaphragm thickness can be closely controlled by growing the layer over the boron-doped surface region, **11**, and then utilizing an etchant which removes the semiconductor substrate but stops at the region, **11**. The portion of region, **11**, under the diaphragm can then be removed by applying an appropriate etchant such as KOH and H_2O for a predetermined period of time. As shown in FIG. **1**, if desired, the etchant may be allowed to penetrate the epitaxial layer-substrate interface to achieve a desired diaphragm thickness less than the original epitaxial layer thickness. Close control of a nontensioned diaphragm's thickness, d, is important since $k_m \alpha d^{+3}$. This step may also be desirable to remove boron impurities which may have diffused into the epitaxial layer from the substrate and added tension to the layer. A further advantage is the fact that the epitaxial layer can be anisotropically etched to provide via holes (**22**, **23**, **24**) or other useful features.

It will be appreciated that, in this example, the epitaxial layer is essentially free of impurities so that there is essentially no tension component contributing to the stiffness of the diaphragm. However, it is contemplated that layers may be fabricated with some tension component, as alluded to above, and still produce advantageous results. The invention is therefore directed to "essentially nontensioned" diaphragms which are intended to include those having a bending stiffness contributing at least $\frac{2}{3}$ of the total stiffness of the diaphragm. It is further contemplated that impurities may be intentionally introduced into the epitaxial layer to satisfy certain needs. For example, the introduction of phosphorus impurities into the layer could provide compression in the layer to counteract any tension that might be produced by the presence of boron impurities.

In accordance with another aspect of the invention, the acoustic venting means comprises a continuous array of small holes, **20**, extending through the backplate and electrode, **19**, which array extends over essentially the entire area of the backplate and electrode portion co-extensive with the diaphragm. Such an array permits a narrow distance between electrodes, **15** and **19**, without overdamping, thereby permitting a reduction in the applied dc bias and in the area of the diaphragm.

FIG. **2** is a top view of some of the holes in the array, which are greatly enlarged for illustrative purposes. It will be appreciated that essentially the entire area of the backplate is covered by these holes and consequently the array is considered to be continuous. The dynamic magnification factor, D, is defined in this application as the amount, in dB, that the frequency response rises in traversing the band from 400 Hz to the upper bound frequency of interest. For telephone applications where the upper bound frequency is 3500 Hz, it is desirable that D be greater than 31 3.0 dB. FIG. **3** shows a generally desirable, calculated, output characteristic (curve B) for telephony, which is achieved in this example in accordance with the invention. Returning to FIG. **2**, such a characteristic can be achieved by a radius, r, of the holes and a center-to-center spacing of the holes, 2ξ, which satisfies the relationship:

$$D = -10 \log \left[\left[1 - \left[\frac{\omega}{\omega_n} \right]^2 \right]^2 + 4 \left[\frac{\omega}{\omega_n} \frac{R_a}{R_c} \right]^2 \right] \tag{3}$$

where D is the minimum desirable magnification factor, ω is the highest acoustic input frequency which will be transmitted expressed in radians (here, $2\pi \times 3.5$ KHz), ω_n is the natural radian frequency of vibration of the diaphragm ($\sqrt{K/M}$), K is the combined stiffness, $k_m + k_a$, of the diaphragm and back acoustic cavity, M is the effective lumped parameter mass of the diaphragm compatible with the prior definition of k_m (which, for a nontensioned diaphragm is approximately 9/5 times the actual mass), R_c is the critical damping level ($2M\omega_n$), and R_a is the actual acoustic damping level of the air film in the air gap between electrodes. The acoustic damping is determined from the following:

$$R_a = \frac{NX^4 B\, 12\pi\eta}{h_e^3} \tag{4}$$

where N is the number of holes, X is the local "radius-of-action" associated with each hole (i.e., the radius of the approximate circular area of air which will be vented through each hole (see FIG. **2**), η is the coeffici-

ent of dynamic viscosity of air, and h_e was defined following equation (1). B is given by the equation:

$$B = \frac{1}{4}\left[\ln\left(\frac{X}{r}\right)^2\right] - \frac{3}{8} + \frac{1}{2}\left(\frac{r}{X}\right)^2 - \frac{1}{8}\left(\frac{r}{X}\right)^4 \quad (5)$$

It will be further appreciated that the geometry of the array yields the following relationships:

$$X = \lambda\xi \quad (6)$$

$$N = \frac{A}{\pi X^2} \quad (7)$$

$$\nu = 1 - \frac{N\pi r^2}{A} \quad (8)$$

where λ is determined by the hole configuration, A is the area of the backplate and electrode portion coextensive with the diaphragm area, N is the total number of holes, and ν is the fraction of the backplate or electrode, **19**, which is not consumed by the holes. For the pattern in this example, where the center of each hole lies at the corner of an equilateral triangle, $\lambda = 1.05$. For other patterns of holes, the backplate area (A) can easily be expressed as a unique function of ξ times N, thus determining X and λ from equations 6 and 7 above. For a square array of holes, for example, λ is 1.13. Also, if noncircular holes are used, the above relationships may still be used for first order calculations if the radius, r, is equated to

$$\sqrt{\frac{A_h}{\pi}}$$

where A_h is the area of the hole.

Thus, given the desired magnification factor D, the area, A, and mass of the diaphragm, M, the average distance between the electrodes when the bias is supplied, h_e, and the combined stiffness of the diaphragm and back acoustic cavity, K, the above equations can be solved to give combinations of hole radius and center-to-center spacing (r, 2ξ) or hole radius and number (r, N) which can be utilized for acoustic venting in accordance with the invention. The preferred combination is that which consumes the minimum amount of electrode area. (For a detailed discussion of the derivation of relationships governing acoustic impedance of the air gap in electrostatic transducers due to holes in one of the electrodes, see Skvor, "On the Acoustical Resistance due to Viscous Losses in the Air Gap of Electrostatic Transducers," *Acustica,* Vol. 19, pp. 259–299 (1967–68), which is incorporated by reference herein.)

In this example, D is −0.8 dB, h_e is 0.56 μm, N is approximately 2000, r is 9.8 μm and 2ξ is 29.8 μm for a diaphragm with area of 1.54×10^{-6} m^2, and effective mass of 1.93×10^{-8} kgm. This leaves a total area not consumed by holes of 61% (ν) of the backplate or electrode area. The capacitance is still sufficient, however, to produce a sufficiently high output signal as specified by the sensitivity previously given.

Of course, the above parameters may be varied according to specific needs. It is recommended, however, that there be a minimum of at least 50 holes per square millimeter to avoid overdamping in the output signal and that each hole have a diameter of less than 100 μm to allow sufficient diaphragm capacitance (at least 1 pF) for operation of the transducer. For the sake of comparison, curve A of FIG. **3** shows the calculated frequency response for a hypothetical microphone having the same dimensions as described in the example shown by curve B, but not including the venting means of the invention. As compared with the microphone of curve B which includes 2000 holes (1300 holes/mm^2) each having a diameter of 19.6 μm, the microphone of curve A includes only 20 holes (13 holes/mm^2) each having a diameter of 196 μm. Both designs have the same electrode area not consumed by holes (61%) so that the capacitances are equal and at least the potential signal-to-noise ratio is the same for both. Nevertheless, curve A shows a severely overdamped frequency response.

It should be appreciated that the average spacing between electrodes (h_e) when a bias is supplied should not vary too much from the air gap (h) with no bias applied if the system is to remain stable. It is recommended, therefore, that h_e be 4–10% less than h. In this example, $h_e = 0.94$ h.

As noted previously, a reduced air gap, which is possible with the venting means of the invention, should also permit a reduction in the external dc bias needed for operation. For a gap, h, between electrodes of less than 3.0 μm, it is expected that the microphone can be operated at less than 5 volts supplied to the capacitor electrodes.

It should also be appreciated that, although the air hole array is described with use of a nontensioned diaphragm, the hole array as described heretofore may also be used with tensioned diaphragms such as that shown in application of Busch-Vishniac, cited previously. In such cases, the area of the diaphragm may be larger than that for the nontensioned diaphragm, but would still be, advantageously, less than 8 mm^2.

It should be further appreciated that while the above relations allow one to specify a uniform air hole array (that is, a constant hole size and pattern), a somewhat nonuniform pattern that might possibly be desired may be designed by applying equation (4) piecewise across the backplate and electrode, and summing over N (holes).

The air gap, **18**, electrode, **19**, backplate layer, **17**, and the air hole array, **20**, may be conveniently formed by known deposition and photolithography steps. For example, layer **16**, which may comprise polycrystalline silicon, can be deposited by chemical vapor deposition and the area which will comprise the air gap is then defined by selectively etching the layer. An etch-stop material, **27**, such as BN or Si_3N_4 can be formed around the walls of the hole. The hole is then filled with a material such as polycrystalline silicon or SiO_2 and planarized. The electrode, **19**, may then be formed by a selective deposition leaving the desired hole array therein. The backplate layer, **17**, which may comprise BN or Si_3N_4 or a combination of like materials, is then deposited and the corresponding hole array formed therein by standard photolithography. The filler material can then be removed from the air cavity by applying another etchant through the hole array. Of course, during these various etching operations, the via holes, **22**, **23** and **24**, needed for interconnection can also be formed.

It will be appreciated that while the above example employed a circular diaphragm and backplate, the principles of the invention may be applied to any shaped diaphragm and backplate.

In accordance with a further aspect of the invention, various acoustical interconnection means and EMI shielding means may be incorporated into the basic microphone structure previously described. For example, FIG. **4** illustrates the formation of a back acoustic cavity, **30**, adjacent to the air hole array, **20**. This cavity is formed within a carrier substrate, **31**, against which the silicon microphone structure is placed. This substrate can be a printed wiring board or other carrier substrate to which electronic components are usually attached.

Coupled to the front cavity, **13**, is an acoustic port, **32**, which is formed from an element, **33**, which is typically a plastic closure. The acoustic port adds a degree-of-freedom to the microphone system and adds a peak to the frequency response of the device to serve various needs. Thus, in the present example, the response shown as curve C in FIG. **3**, where the peak is placed near the upper end of the telephone band, is obtained from the microphone characterized by curve B simply by adding an acoustic port having a diameter of 150 μm and a length of 1600 μm. In this example, the cavity, **13**, has a volume of 0.94 mm^3. Long holes with narrow diameters yielding high acoustic mass are generally needed in this silicon microphone application due to the large stiffness of the small, front acoustic cavity. In general, diameters of 100–180 μm and lengths of 600–2000 μm are expected to be useful for producing peaks where desired. For telephone applications, it is desirable to form the peak within the frequency range 2.8–4.5 KHz. Although not illustrated in these figures, all members forming acoustic ports or cavities are acoustically sealed by standard means such as with adhesives or by clamping.

If shaping of the frequency response, as shown in curve C, is not needed, the embodiment shown in FIG. **5** might be utilized. Here, the cavity, **30**, formed in carrier substrate, **31**, acts as an extension of front cavity, **13**, and the acoustic signal is made incident on the diaphragm through the hole array, **20**. In this embodiment, in fact, the cavity extension, **30**, may be eliminated so that the acoustic cavity is formd entirely within the semiconductor substrate. In any event, no extra parts are needed to form the acoustic interconnections. A further advantage is that electrical contact can be made to the microphone by wire bonds, **34** and **35**, from the carrier substrate, **31** to the top of the backplate layer.

In the embodiment illustrated in FIG. **6**, sound is again incident on the diaphragm through an acoustic port, **32**, coupled to the front cavity, **13**. Here, however, the sound port is formed in the carrier substrate, **31**, so that the microphone is again mounted with the backplate side-up permitting wire bonding. Additionally, an enclosure member, **36**, is provided surrounding the entire semiconductor microphone. This member can be made of conductive or conductively plated plastic or metal so as to provide a shield for the device against electromagnetic interference. At the same time, the member forms a back acoustic cavity, **30**, for the microphone. The member can be grounded, for example, by bonding to grounded pad, **37**, formed on the carrier substrate. Thus, EMI shielding is provided with a minimum of piece-parts.

It should be appreciated that although the acoustical interconnection means and EMI shielding means are described with use of a nontensioned diaphragm, such may also be used with tensioned membranes as for example shown in application of Busch-Vishniac, cited previously.

It will be appreciated that the inventive features discussed herein could also apply to a pressure gradient type microphone where sound is allowed to strike both sides of the diaphragm, thus effecting a noise-canceling and directional response. To produce such a device in FIGS. **4** and **5**, a secondary sound port would simply be placed in the carrier substrate, **31**, while in the embodiment illustrated in FIG. **6** a small secondary port would be placed through enclosure member, **36**. In any case, the second side of the diaphragm is accessed.

It will also be appreciated that although the above discussion has focused on the microphone, the principles of the invention may also be applicable to other types of electroacoustic transducers utilizing a capacitor whose capacitance varies in accordance with a vibrating diaphragm, whether an acoustic signal is converted to an electrical signal or vice-versa. For example, a loudspeaker or hearing aid receiver might be fabricated by applying a varying electrical signal to the capacitor electrodes (**15** and **19**) which causes vibration of the diaphragm (**14**) due to the varying deflection of the electrode (**15**) attached thereto. An acoustic output signal would therefore be produced. Thus, whichever way the energy conversion is taking place, the electric field between the electrodes varies in relationship with the vibrating diaphragm to permit conversion between electrical and acoustic signals.

It will also be realized that the invention is not limited to telephone band frequencies (0.3–3.5 KHz) but can be used in the full audio bandwidth (0.02–20 KHz). In fact, this silicon transducer invention can find application in the ultrasonic band (20–1000 KHz).

Various additional modifications will become apparent to those skilled in the art. All such variations which basically rely on the teachings through which the invention has advanced the art are properly considered within the spirit and scope of the invention.

What is claimed is:

1. An electroacoustic transducer formed within a semiconductor substrate comprising

an essentially nontensioned diaphragm comprising a semiconductor material having a mechanical bending moment stiffness such that it vibrates in response to an input signal at audio and/or ultrasonic frequencies; and

a pair of electrodes placed with respect to said diaphragm so that the electric field between the electrodes varies in relationship with the vibrating diaphragm to permit conversion between electrical and acoustic signals.

2. The device according to claim **1** wherein the transducer is a microphone and one of the electrodes is formed to vibrate with the diaphragm such that the capacitance varies in response to an acoustic signal incident on said diaphragm.

3. The device according to claim **2** wherein the diaphragm is a single crystalline epitaxial layer formed over the surface of a semiconductor substrate.

4. The device according to claim **3** wherein the thickness of the epitaxial layer is in the range 1–5 μm.

5. The device according to claim **2** wherein the area of the diaphragm is less than 5 mm^2.

6. The device according to claim **1** wherein the effective mechanical lumped parameter stiffness of the diaphragm, associated with the diaphragm's linearly averaged deflection, is less than 5500 N/m.

7. The device according to claim 3 wherein the epitaxial layer is formed over a surface layer in the substrate which has a higher impurity concentration than the bulk of the substrate.

8. The device according to claim 1 wherein an air gap is formed between the two electrodes and the device further comprises a continuous array of small holes extending through one of the electrodes to the air gap, said array covering essentially the entire area of that portion of the electrode which is co-extensive with the diaphragm.

9. The device according to claim 1 wherein the transducer is adapted for operation with a dc bias across the electrodes of less than 5 volts.

10. The device according to claim 1 wherein an air gap is formed between the electrodes and the device further comprises:

a front acoustic cavity defined by a portion of the semiconductor substrate adjacent to the diaphragm;

a rear acoustic cavity adjacent to one of the electrodes and defined by a member which also shields the transducer from electromagnetic interference;

acoustic venting means comprising an array of holes connecting the air gap to the back acoustic cavity; and

an acoustic port defined by an element adjacent to the front acoustic cavity, the size of the port being such as to produce a desired frequency characteristic for the transducer.

11. An electroacoustic transducer formed within a semiconductor substrate comprising:

a diaphragm which vibrates in response to an input signal at audio and/or ultrasonic frequencies;

a pair of electrodes placed with respect to said diaphragm so that the electric field between the electrodes varies in relation to the vibrating diaphragm to permit conversion between electrical and acoustic signals, the electrodes defining an air gap therebetween; and

acoustic venting means comprising a continuous array of small holes extending through one of the electrodes to the air gap, said array covering essentially the entire area of that electrode which is co-extensive with the diaphragm, where the density of the holes in the array is at least 50 per mm^2 and the diameter of the holes is less than 100 μm.

12. The device according to claim 11 wherein the transducer is a microphone and one of the electrodes is formed to vibrate with the diaphragm such that the capacitance varies in response to an acoustic signal incident on said diaphragm.

13. The device according to claim 11 wherein a backplate layer is included adjacent to said one of the electrodes and said array of holes is included through the layer.

14. The device according to claim 11 wherein the spacing between electrodes is less than 3.0 μm when no dc bias is supplied to the electrodes.

15. The device according to claim 11 wherein the area of the diaphragm is less than 8 mm^2.

16. The device according to claim 1 wherein the spacing between the electrodes is uniform when no bias is supplied to the electrodes.

17. The device according to claim 11 wherein the transducer is adapted to operate with a dc bias across the electrodes of less than 5 volts.

18. The device according to claim 11 wherein the dynamic magnification factor of the output response of the transducer is greater than or equal to −3.0 dB.

19. The device according to claim 11 where the radius, r, center-to-center spacing, 2ξ, and number, N, of holes in the array satisfies the relationships:

$$D = -10 \log \left[\left(1 - \left[\frac{\omega}{\omega_n} \right]^2 \right)^2 + 4 \left(\frac{\omega}{\omega_n} \frac{R_a}{R_c} \right)^2 \right]$$

$$R_a = \frac{NX^4 B\, 12\pi\eta}{h_e^3}$$

$$B = \frac{1}{4} \left[\ln \left(\frac{X}{r} \right)^2 \right] - \frac{3}{8} + \frac{1}{2} \left(\frac{r}{X} \right)^2 - \frac{1}{8} \left(\frac{r}{X} \right)^4$$

$$X = \lambda \xi$$

and

$$N = \frac{A}{\pi X^2}$$

where D is the minimum desirable dynamic magnification factor, ω is the highest radian frequency of the input signal to be transmitted, ω_n is the natural radian frequency of vibration of the diaphragm, R_a is the damping level of the air in the gap, R_c is the critical damping level, X is the local radius-of-action associated with each hole, h_e is the average distance between the electrodes when the desired dc bias is supplied thereto, λ is a factor relating X and ξ for the geometry of the hole array, A is the area of the diaphragm, and η is the coefficient of dynamic viscosity of air.

20. The device according to claim 11 further comprising:

a front acoustic cavity defined by a portion of a semiconductor substrate adjacent to the diaphragm;

a back acoustic cavity adjacent to the air hole array and defined by a member which also shields the transducer from electromagnetic interference; and

an acoustic port defined by an element adjacent to the front acoustic cavity, the size of the port being such as to produce a desired frequency response characteristic for the transducer.

21. The device according to claim 12 wherein the microphone is adapted to receive the acoustic signal through the hole array.

22. An electroacoustic transducer formed within a semiconductor substrate comprising:

a diaphragm which vibrates in response to an input signal at audio and/or ultrasonic frequencies;

a front acoustic cavity defined by a portion of the semiconductor substrate adjacent to the diaphragm;

a pair of electrodes placed with respect to said diaphragm so that the electric field between the electrodes varies in relation with the vibrating diaphragm to permit conversion between electrical and acoustic signals and so that an air gap is formed between the electrodes;

a back acoustic cavity adjacent to one of the electrodes and defined by a member which also shields the transducer from electromagnetic interference;

acoustic venting means comprising an array of holes connecting the air gap to the back acoustic cavity; and

an acoustic port defined by an element adjacent to the front acoustic cavity, the size of the port being such as to produce a desired frequency response characteristic for the transducer.

23. The device according according to claim 22 wherein the transducer is a microphone and one of the electrodes is formed to vibrate with the diaphragm such that the capacitance varies in response to an acoustic signal incident on said diaphragm.

24. The device according to claim 22 wherein the acoustic port is defined in a carrier to which the semiconductor substrate is attached.

25. The device according to claim 22 wherein the back acoustic cavity is defined by a conductive member which covers the entire semiconductor substrate in which the integrated microphone is formed.

26. The device according to claim 22 wherein the diameter and length of the acoustic port is such as to produce a peak in the output response in the frequency range 2.8–4.5 KHz.

* * * * *

Exhibit PA-O



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

(10) **Patent No.: US 8,014,553 B2**
(45) **Date of Patent: Sep. 6, 2011**

(54) **EAR-MOUNTED TRANSDUCER AND EAR-DEVICE**

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(73) Assignee: **Nokia Corporation**, Espoo (FI)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 1338 days.

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H04R 25/00 (2006.01)

(52) **U.S. Cl.** **381/380**; 381/151; 381/326

(58) **Field of Classification Search** 381/151, 381/326, 380; 600/379; 607/136–137
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,025,734	A *	5/1977	Aloupis	381/151
5,692,059	A *	11/1997	Kruger	381/151
5,933,506	A *	8/1999	Aoki et al.	381/151
6,463,157	B1	10/2002	May	
7,433,484	B2 *	10/2008	Asseily et al.	381/355
2005/0027515	A1	2/2005	Huang et al.	
2005/0033571	A1	2/2005	Huang et al.	
2006/0159297	A1	7/2006	Wirola et al.	
2007/0086600	A1 *	4/2007	Boesen	381/79

FOREIGN PATENT DOCUMENTS

EP	1 320 282	A2	6/2003
JP	02002125298		4/2002
WO	WO/02/07477		1/2002
WO	WO/02052890		7/2002
WO	WO/2004/066669		8/2004

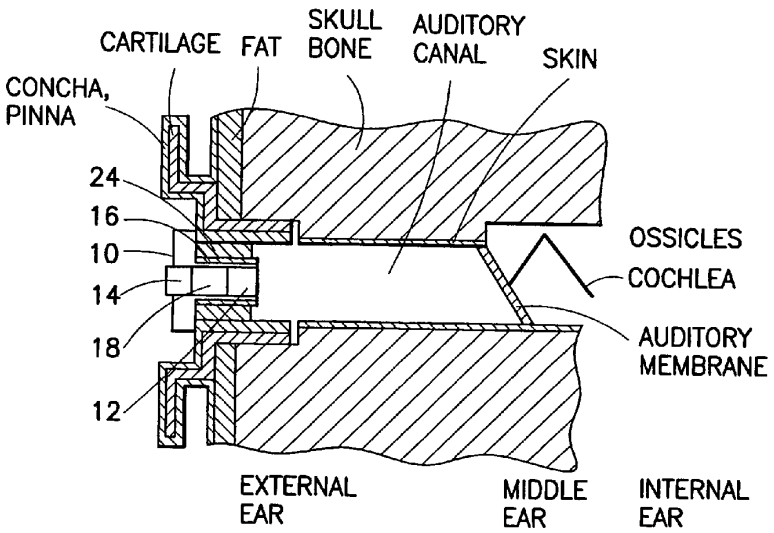
* cited by examiner

Primary Examiner — Davetta Goins
Assistant Examiner — Matthew Eason

(57) **ABSTRACT**

The specification and drawings present a new method, apparatus and software product for providing flexible audio communication solutions using ear-devices utilizing, e.g., electrode transducers with one or more sensors comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range for using, for example, in headsets and hearing aids. The ear-device can be configured for inserting it into a human ear for a handsfree operation and the sensors can be configured to detect human tissue vibrations using the surface resonator cavity. The acoustic communication solutions with these ear-devices may include: providing two-way communications in normal conditions as well as in noisy conditions, providing protection of hearing, recording the true sound field bin-aurally, providing a playback capability, providing volume enhancement and equalization for persons with hearing defects, etc.

44 Claims, 9 Drawing Sheets



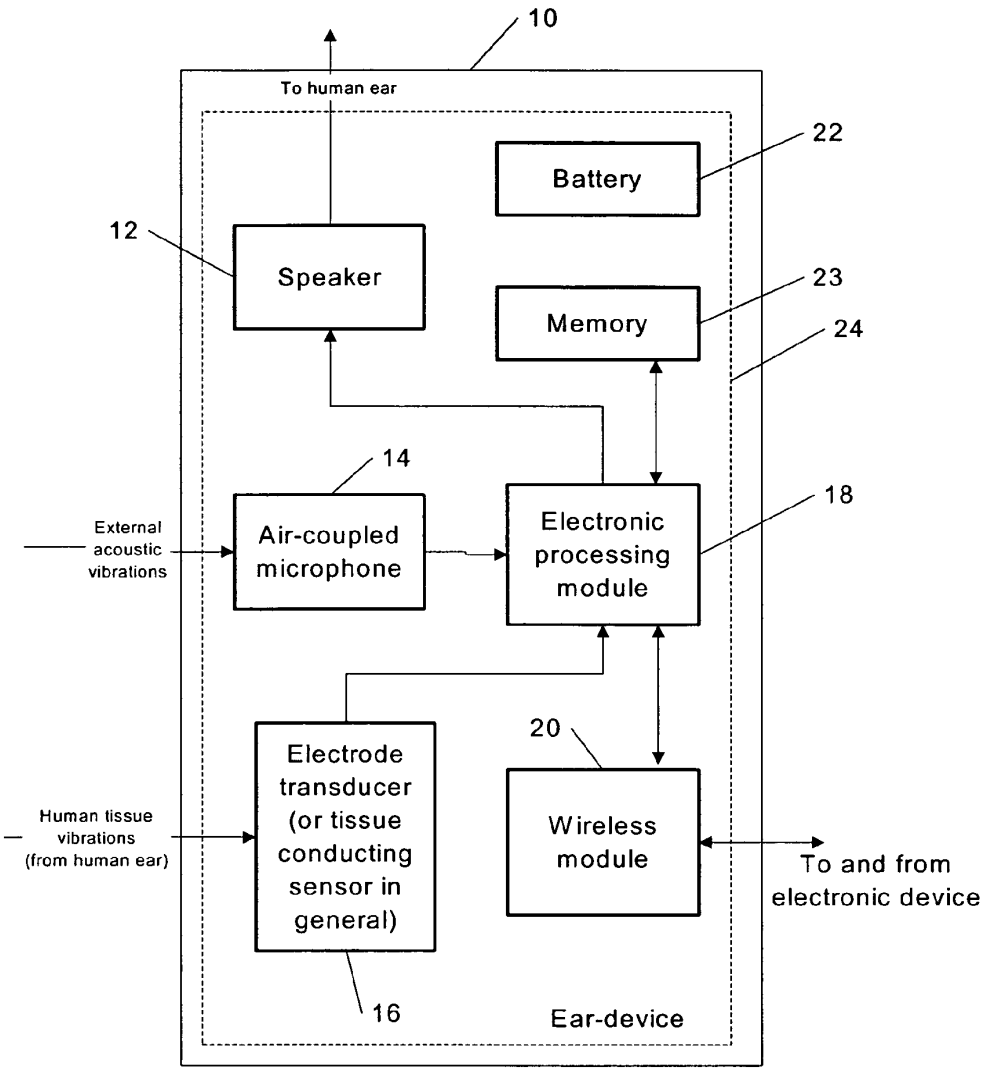

10
To human ear
22
Battery
12
Speaker
23
Memory
24
14
18
External acoustic vibrations
Air-coupled microphone
Electronic processing module
20
Electrode transducer (or tissue conducting sensor in general)
Wireless module
Human tissue vibrations (from human ear)
To and from electronic device
16
Ear-device

Figure 1

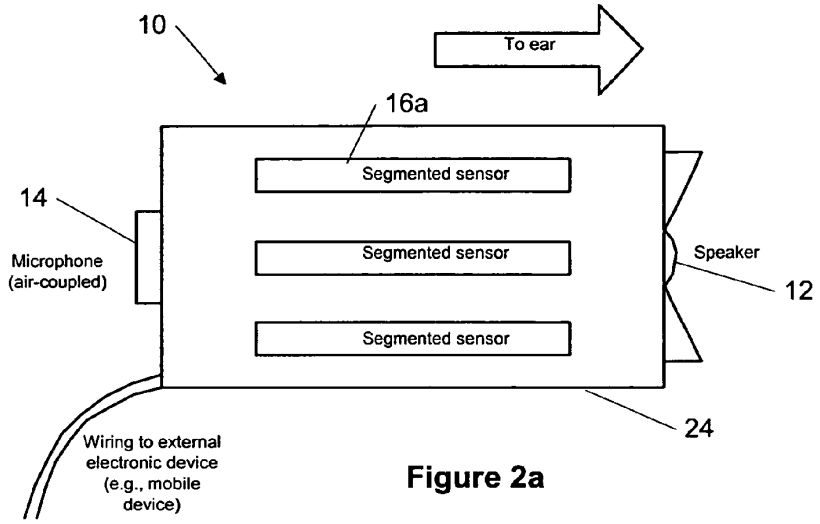


Figure 2a

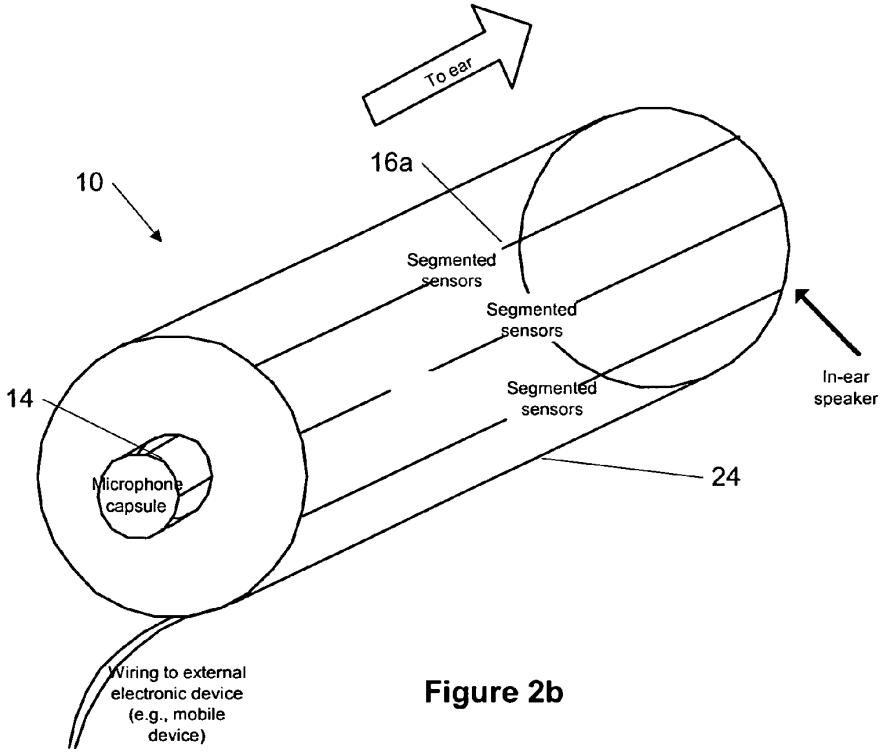


Figure 2b

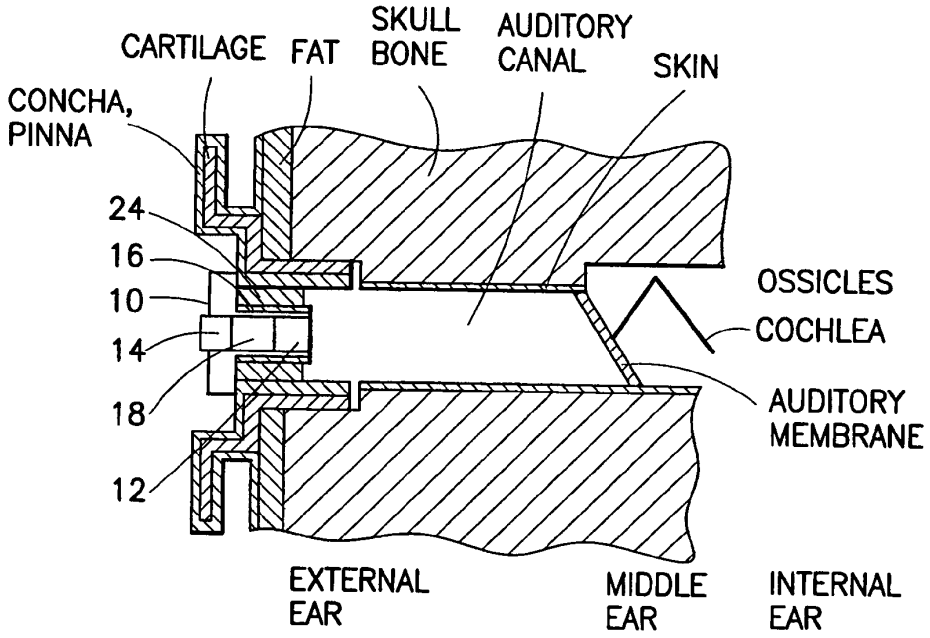

CARTILAGE
FAT
SKULL BONE
AUDITORY CANAL
SKIN
CONCHA, PINNA
24
16
10
14
18
12
OSSICLES
COCHLEA
AUDITORY MEMBRANE
EXTERNAL EAR
MIDDLE EAR
INTERNAL EAR

FIG.3

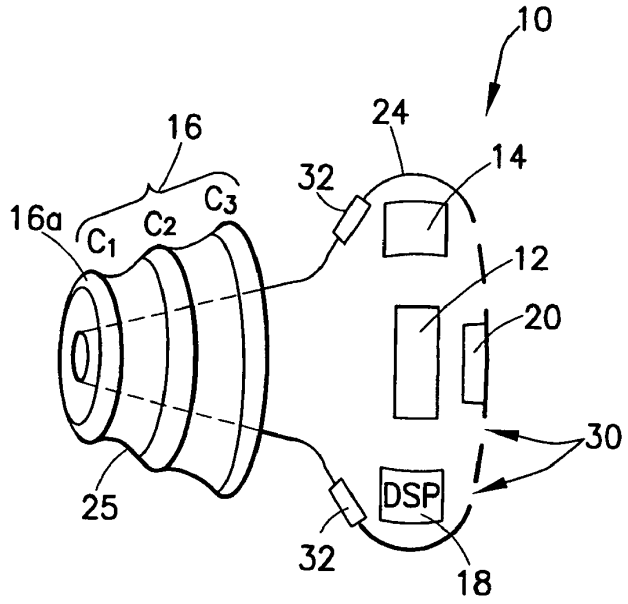

10
16
24
14
32
16a
C1
C2
C3
12
20
30
DSP
25
32
18

FIG.4a

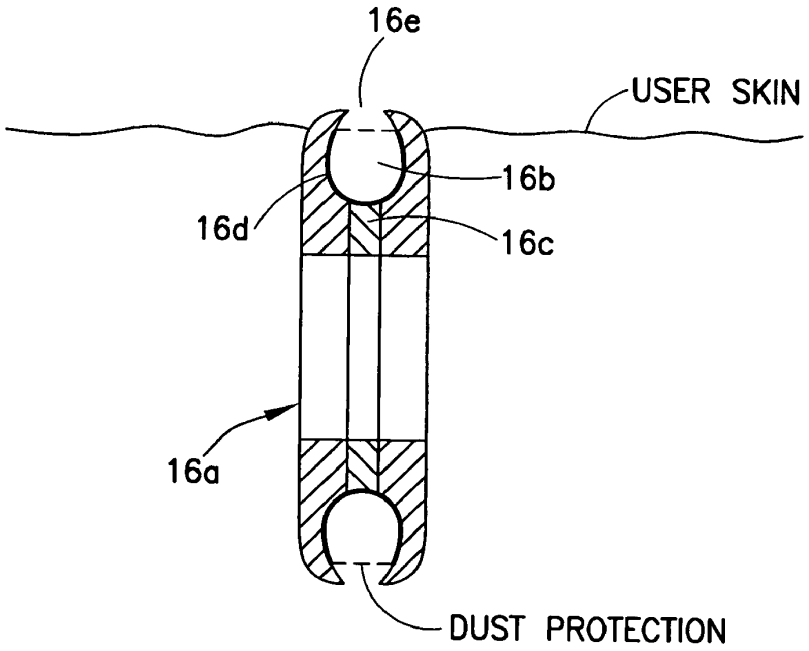

16e
USER SKIN
16b
16d
16c
16a
DUST PROTECTION

FIG.4b

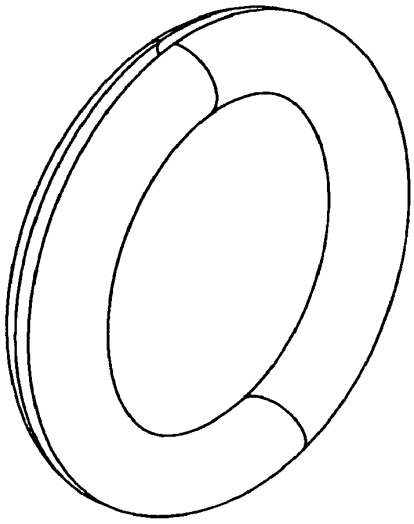


Figure 4c

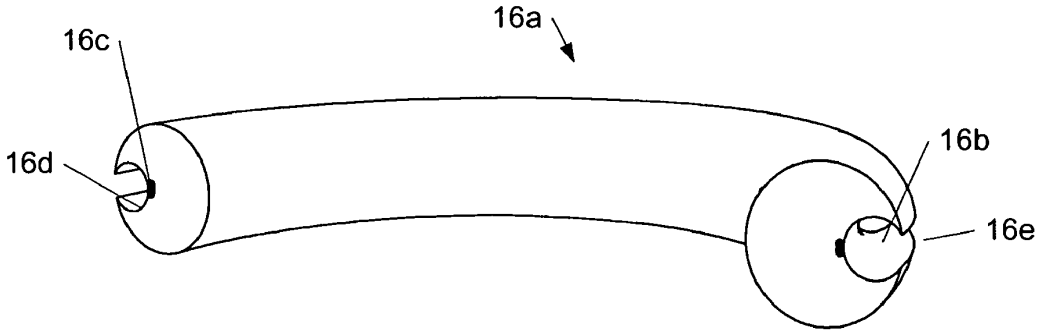


Figure 4d

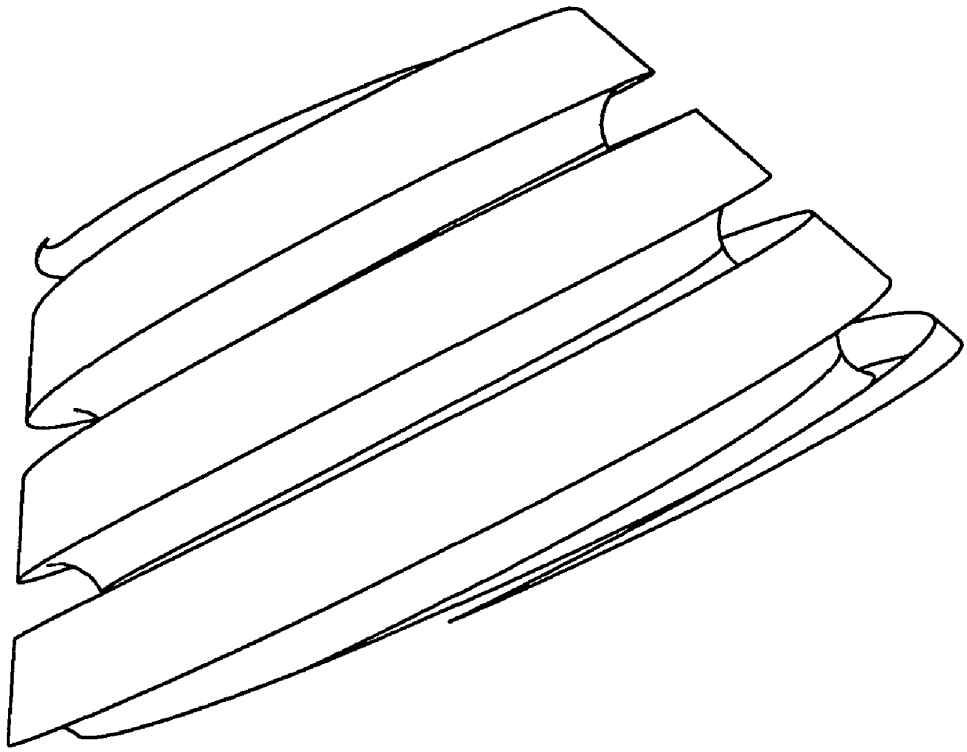


Figure 4e

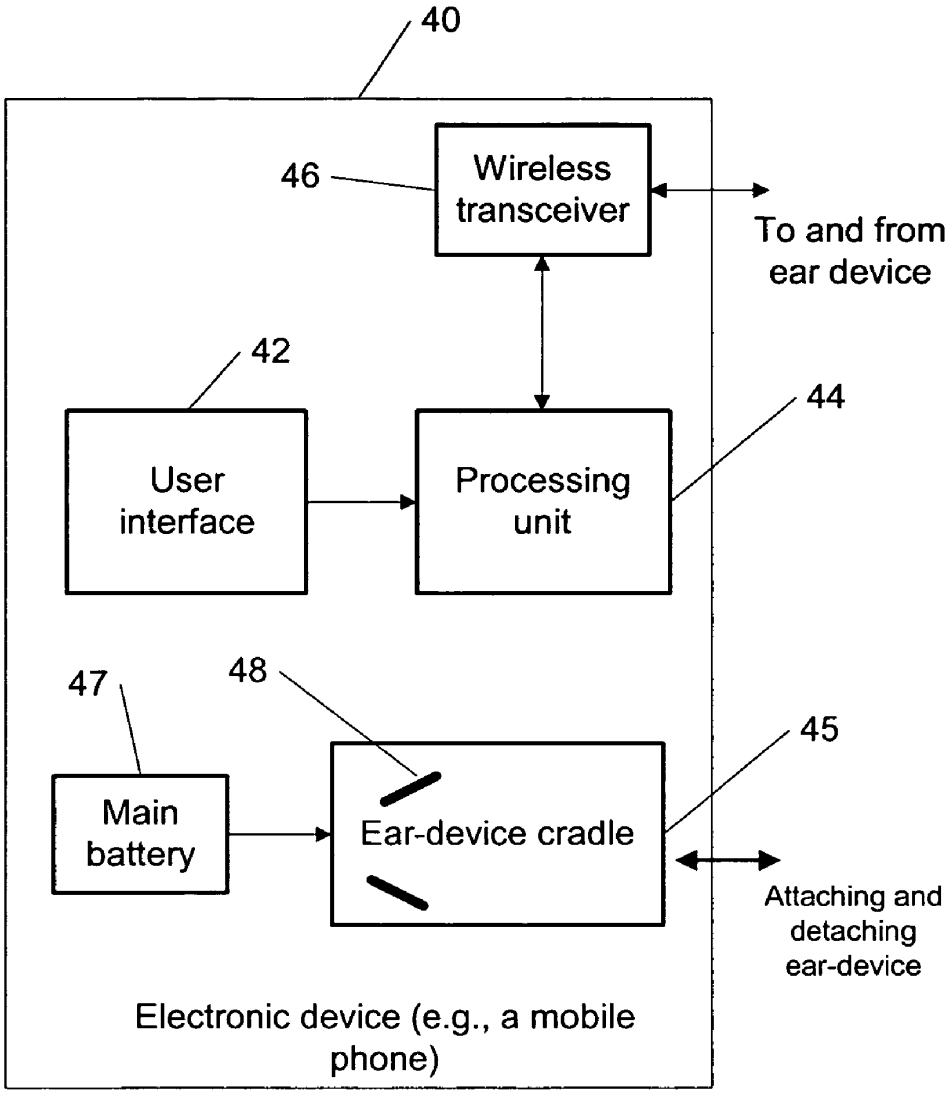

40
46
Wireless transceiver
To and from ear device
42
44
User interface
Processing unit
47
48
45
Main battery
Ear-device cradle
Attaching and detaching ear-device
Electronic device (e.g., a mobile phone)

Figure 5

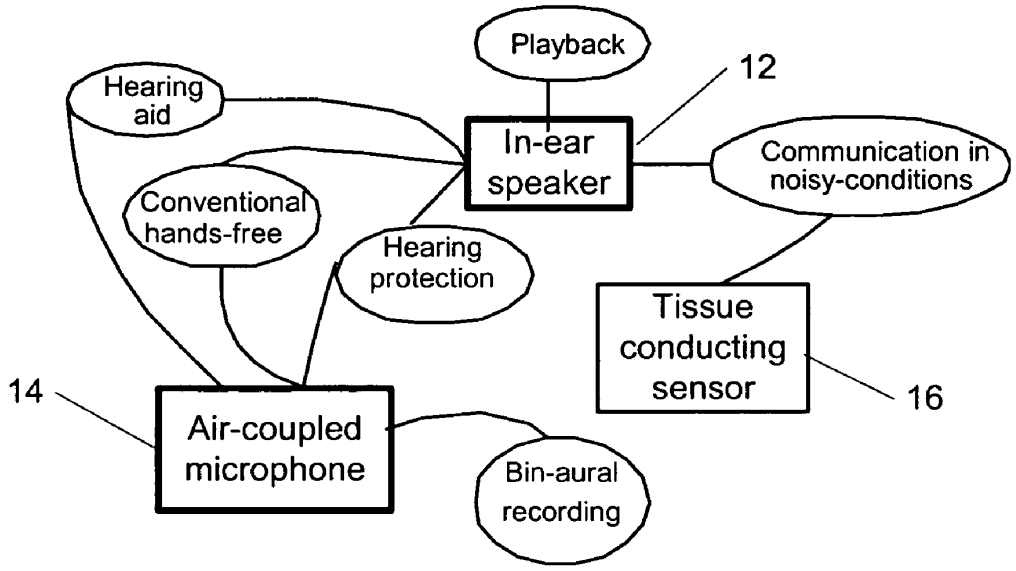

Playback
12
Hearing aid
In-ear speaker
Communication in noisy-conditions
Conventional hands-free
Hearing protection
Tissue conducting sensor
14
16
Air-coupled microphone
Bin-aural recording

Figure 6

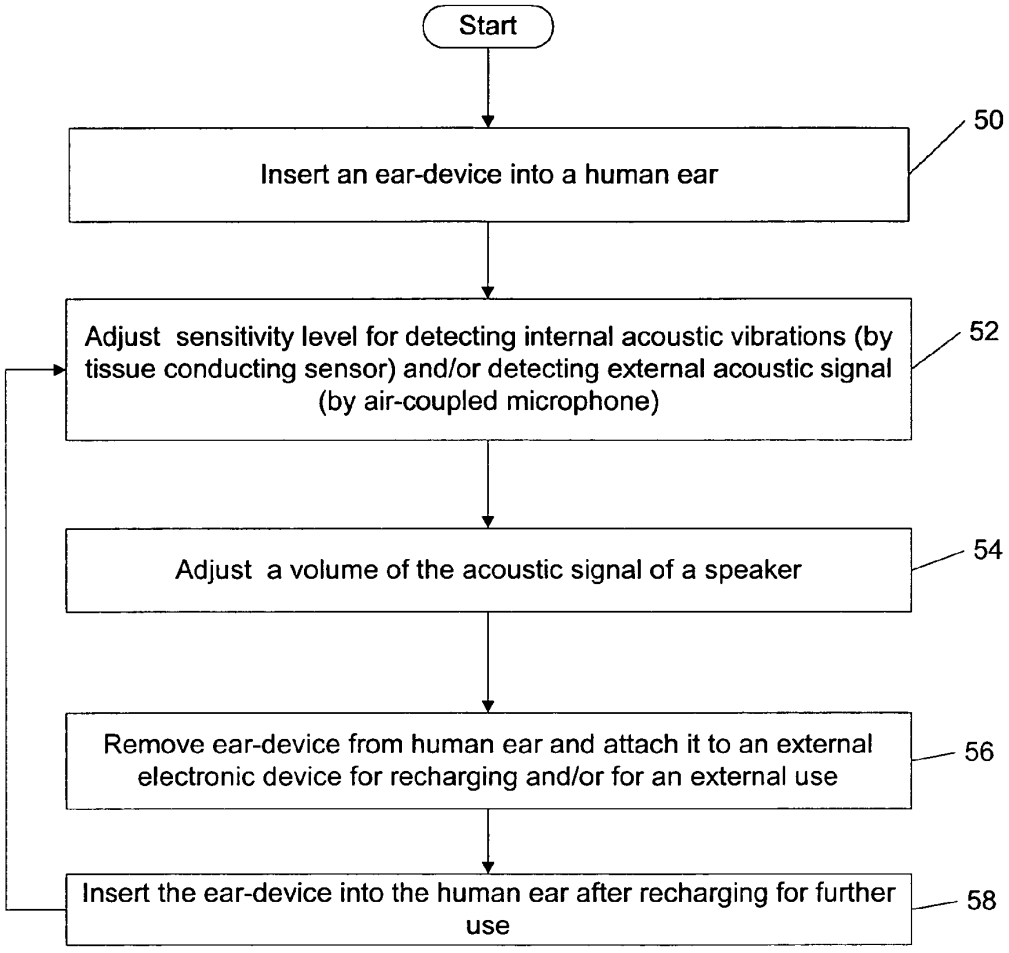

Start
Insert an ear-device into a human ear
50
Adjust sensitivity level for detecting internal acoustic vibrations (by tissue conducting sensor) and/or detecting external acoustic signal (by air-coupled microphone)
52
Adjust a volume of the acoustic signal of a speaker
54
Remove ear-device from human ear and attach it to an external electronic device for recharging and/or for an external use
56
Insert the ear-device into the human ear after recharging for further use
58

Figure 7

EAR-MOUNTED TRANSDUCER AND EAR-DEVICE

TECHNICAL FIELD

The present invention relates generally to communication devices and more specifically, to integrated ear-devices for providing acoustic communication solutions.

BACKGROUND ART

Novel multimedia devices are having multifunction applications. Phones and multimedia devices are used for connecting, finding, storing and spreading information via digital or audio channels (speech and listening). Such portable devices are used in different environments such as offices, silent cabinets, hospitals, metro, public but quiet places, etc., or in more noisy conditions as noisy streets, riding motorbike, etc. To satisfy sufficient quality of communications in all of these conditions the audio system has to provide a "silent input method" (do not disturb your environment while communicating) or it has to cancel noise from the surrounding environment (noise cancellation). The quality of communications is directly related to use conditions and limitations set by an audio-digital-audio signal conversion mechanism. Versatile usage conditions are difficult to cover by using present technical solutions for digital audio communications, which limits its quality of information exchange and user satisfaction by products and services.

To improve audio conversion mechanisms, different technologies have been used. An example of a relatively new and direct conversion mechanism which avoids the air as sound wave propagation medium is a Silent Violin where vibrations from the wires are propagated through solid materials towards an ADC (analog-to-digital converter) for converting to a digital signal, which is then amplified and finally released into the air providing a beautiful sound quality.

Hardware (HW) miniaturization trends put size limits on both microphones and speakers challenging the quality of audio communications. Usually an audio signal propagates from the user's mouth to the air, to a microphone "listening" the air and then digitally converted and electrically amplified and finally spread into the air (e.g., by a speaker). The air is not the best medium for propagation of sound waves. Furthermore, the air is usually full of different sounds (noises) which are not useful while communicating by the portable electronic equipment. The air (gas mixture) is not the best medium, solid and liquid state materials are much more efficient in sound propagation. Moreover, the audio signal in the air has large dissipation requiring the user to speak relatively loudly to achieve a good quality of the audio communication. Therefore, a direct coupling of a microphone to a user body (avoiding the air) can present a very advanced solution for silent and not disturbing communications, such that the user does not disturb the environment while speaking (high sensitivity level) and the user is not disturbed by the environmental conditions (noise free communication).

Ordinary 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).

In the available hands-free solutions, the background noise is clearly hearable to the receiver of the call. Certain algorithms are utilized in the noise reduction, but they can generally reduce the noise level only by 10-15 dB. However, the mobile devices are also used in circumstances, in which the noise level is high compared to the speech level. This poses a problem for the current sensor solutions (traditional pressure microphones) and, also, for actuator solutions (traditional loudspeakers). Moreover, often the users wish to protect their hearing under such conditions. For such protection either circumaural or insert-type ear-defenders are used. In both cases, the protection solution further complicates or prohibits the communication.

DISCLOSURE OF THE INVENTION

According to a first aspect of the invention, an apparatus, comprises: an electrode transducer, comprising at least one sensor, the at least one sensor comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range, wherein the apparatus is configured for inserting it into a human ear for a handsfree operation and the at least one sensor is configured for detecting human tissue vibrations using the surface resonator cavity.

According further to the first aspect of the invention, the electrode transducer may comprise one or more sensors of the at least one sensor with one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters. Further, each of the plurality of sensors may be optimized for a different acoustic frequency range. Further still, the electrode transducer may comprise a soft material between the sensors for adapting to the human ear.

According further to the first aspect of the invention, the at least one sensor may have a shape of a ring, line or a spiral shape.

Still further according to the first aspect of the invention, the electrode transducer may be configured for a speech detection by detecting the human tissue vibrations.

According further to the first aspect of the invention, the apparatus may further comprise an impedance-matching layer covering the at least one sensor for efficient and gentle acoustic coupling of the segmented sensors to the human ear. Further, when the apparatus is attached to an electronic device, the at least one sensor may be disengaged from a contact with the impedance-matching layer.

According still further to the first aspect of the invention, the apparatus may further comprise: a microphone, for detecting acoustic vibrations, wherein the apparatus is configured to adjust a sensitivity level or a sensitivity ratio of:

a) detecting the human tissue vibrations by the electrode transducer, and

b) detecting the acoustic vibrations by the microphone.

According still further to the first aspect of the invention, the apparatus may further comprise: a speaker, for providing an acoustic signal. Further, the speaker may be configured to adjust a volume of the acoustic signal coupled to the human ear. Further still, the speaker may be configured to adjust spectral content of the acoustic signal coupled to the human ear. Further still, the microphone may be configured to provide at least one of: a) a two-way communication in normal or noisy conditions, b) bin-aural recording, c) a hearing protection for the human ear from external noises, d) volume enhancement and equalization as a hearing aid, and e) a playback capability in the normal or noisy conditions.

According yet further still to the first aspect of the invention, the apparatus may further comprise an electronic processing module for supporting functionalities of all or selected components of the apparatus.

Yet still further according to the first aspect of the invention, the processing module may be configured to perform a

decoding process such that the apparatus is further configured to provide media player capabilities.

Still yet further according to the first aspect of the invention, the apparatus may further comprise at least one of: a) a battery for supporting an operation of all components of the apparatus requiring an electric power, and b) a memory for storing recorded files.

Still further still according to the first aspect of the invention, the apparatus may be a part of an electronic device and may be configured for detaching from the electronic device for the inserting into the human ear and for attaching back to the electronic device. Further, the apparatus, when attached to the electronic device, may be configured to provide a further handsfree operation. Further still, the apparatus may comprise a battery and the electronic device may be configured to recharge the battery when the apparatus is attached to the electronic device.

According further still to the first aspect of the invention, the apparatus may be connected to an electronic device and the apparatus may further comprise a wireless module for providing a wireless communication of the apparatus with the electronic device, or the apparatus may be connected by a wire to the electronic device. Further, the electronic device may be a wireless device, a portable communication device, a personal digital assistant or a mobile phone.

According yet further still to the first aspect of the invention, the apparatus may be configured to operate without external assistance.

According to a second aspect of the invention, an electrode transducer, comprises: at least one sensor, which comprises a surface resonator cavity sensitive to a predetermined acoustic frequency range, wherein the at least one sensor, when inserted into a human ear for a handsfree operation, is configured to detect human tissue vibrations using the surface resonator cavity.

According further to the second aspect of the invention, the electrode transducer may comprise one or more sensors of the at least one sensor with one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters.

Further according to the second aspect of the invention, each of the plurality of sensors may be optimized for a different acoustic frequency range.

Still further according to the second aspect of the invention, the electrode transducer may comprise a soft material between the sensors for adapting to the human ear.

According further to the second aspect of the invention, the at least one sensor may have a shape of a ring, line or a spiral.

According to a third aspect of the invention, a method, comprises: inserting an ear-device into a human ear for a handsfree operation, wherein the ear-device comprises: an electrode transducer comprising at least one sensor, the at least one sensor comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range; and detecting by the at least one sensor human tissue vibrations using the surface resonator cavity.

Further according to the third aspect of the invention, the ear-device may further comprise a microphone, and the method may further comprise: detecting acoustic vibrations using the microphone, wherein the ear-device is configured to adjust a sensitivity level or a sensitivity ratio of: a) detecting the human tissue vibrations by the electrode transducer, and b) detecting the acoustic vibrations by the microphone.

Still further according to the third aspect of the invention, the ear-device may further comprise a speaker providing an acoustic signal, and the method may further comprise: adjusting at least one of: a) a volume of the acoustic signal coupled to the human ear, and b) spectral content of the acoustic signal coupled to the human ear.

According further to the third aspect of the invention, the method may further comprise: taking the ear-device out of the human ear and attaching the ear-device to an electronic device for a further handsfree operation or for recharging a battery of the ear-device.

According still further to the third aspect of the invention, the ear-device may further comprise a wireless module, and the method may further comprise: providing a wireless communication of the ear-device with the electronic device.

According yet further still to the third aspect of the invention, the microphone may be configured to provide at least one of: a) two-way communications in normal or noisy conditions, b) bin-aural recording, c) a hearing protection for the human ear from external noises, d) volume enhancement and equalization as a hearing aid, and d) a playback capability in the normal or noisy conditions.

According further still to the third aspect of the invention, the electrode transducer may comprise one or more sensors of the at least one sensor with one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters. Further, each of the plurality of sensors may be optimized for a different acoustic frequency range.

Yet still further according to the third aspect of the invention, during the detecting, the surface resonator cavity may be located substantially in a vicinity of a human tissue but without a direct physical contact with the human tissue.

According to a fourth aspect of the invention, a computer program product comprises: a computer readable storage structure embodying computer program code thereon for execution by a computer processor with the computer program code, wherein the computer program code comprises instructions for performing the third aspect of the invention, indicated as being performed by any component or a combination of components of the ear-device or an electronic device connected to the ear device using a wireless or non-wireless method.

According to a fifth aspect of the invention, a system, comprises: at least one ear-device, comprising:

an electrode transducer comprising at least one sensor, the at least one sensor comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range, wherein the apparatus is configured for inserting it into a human ear for a handsfree operation and the at least one sensor is configured for detecting human tissue vibrations using the surface resonator cavity; and

an electronic device, for providing communicating acoustically generated signals to and from the ear-device.

According further to the fifth aspect of the invention, the system may further comprise: a microphone, for detecting acoustic vibrations, wherein the apparatus is configured to adjust a sensitivity level or a sensitivity ratio of:

a) detecting the human tissue vibrations by the electrode transducer, and

b) detecting the acoustic vibrations by the microphone; and

a speaker, for providing an acoustic signal. Further, the at least one ear-device may comprise two ear-devices and the two ear-devices, when inserted into both human ears, may be configured for at least one of:

a) to provide bin-aural recording,

b) to provide a hearing protection for the human ears from external noises; and

c) to provide an adjustable hearing protection for the human ears from external noises.

Further according to the fifth aspect of the invention, the at least one ear-device may comprise a battery for supporting an operation of all components of the ear-device requiring an electric power, and the electronic device may be configured for recharging the battery.

Still further according to the fifth aspect of the invention, during the detecting, the surface resonator cavity may be located substantially in a vicinity of a human tissue but without a direct physical contact with the human tissue.

According to a sixth aspect of the invention, an apparatus, comprises: transducer means, comprising at least one sensor, the at least one sensor comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range, wherein the apparatus is configured for inserting it into a human ear for a handsfree operation and the at least one sensor is configured for detecting human tissue vibrations using the surface resonator cavity.

According further to the sixth aspect of the invention, the transducer means may be an electrode transducer.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the nature and objects of the present invention, reference is made to the following detailed description taken in conjunction with the following drawings, in which:

FIG. **1** is a block diagram of an ear-device, according to an embodiment of the present invention;

FIGS. **2***a* and **2***b* are schematic representations (a side view and a 3-dimensional view), respectively, of an ear-device, with one end that is located inside an ear canal having a miniature speaker, another end having a microphone and having the tissue conducting sensors on the outer surface, according to an embodiment of the present invention;

FIG. **3** is a schematic representation of an ear-device inserted into a human ear, according to an embodiment of the present invention;

FIG. **4***a* through **4***e* are schematic representations of a block diagram of an ear-device with a conical shape tissue conducting sensor (FIG. **4***a*) showing a structure of a segmented ring sensor (FIG. **4***b*) and a spiral construction (FIG. **4***c*), according to an embodiment of the present invention;

FIG. **5** is a block diagram of an external electronic device (e.g., a mobile phone) which can be a host (mother-device) for an ear-device, according to an embodiment of the present invention;

FIG. **6** is a diagram demonstrating different applications utilizing an ear-device used for different applications described herein, showing individual components (rectangles) of an ear device utilized for different functionalities (ellipses), according to embodiments of the present invention; and

FIG. **7** is a flow chart illustrating utilization of an ear-device, according to an embodiment of the present invention.

MODES FOR CARRYING OUT THE INVENTION

A new method, apparatus and software product for providing flexible audio communication solutions using ear-devices (for example, multifunctional and integrated ear-devices) utilizing, e.g., electrode transducers with at least one sensor (i.e., it could be one or more sensors) comprising a surface resonator cavity sensitive to a predetermined acoustic frequency range for using, for example, in headsets and hearing aids. The ear-device can be configured for inserting it into a human ear for a handsfree operation and the at least one sensor can be configured to detect human tissue vibrations using said surface resonator cavity (surface resonator cavity can be located, e.g., in a vicinity of a human tissue with some or without a direct physical contact with said human tissue).

According to embodiments of the present invention, the acoustic communication solutions utilizing multifunctional integrated ear-devices described herein, may include (but are not limited to): providing two-way communications in normal conditions as well as in noisy conditions, providing protection of hearing, recording the true sound field bin-aurally, providing a playback capability, providing volume enhancement and equalization for persons with hearing defects, etc. According to various embodiments, the ear-device can operate by itself or it can be attached to a portable communication device like a mobile phone.

According to an embodiment of the present invention, an ear-device can comprise all or a combination of the following components: a tissue conducting sensor such as the electrode transducer (e.g., using a single sensor or a plurality of segmented sensors) for detecting human tissue vibrations, a microphone (e.g., an air-coupled microphone) for detecting primarily external acoustic vibrations, a speaker for providing an acoustic signal, and a housing for holding the tissue conducting sensor, the air-coupled microphone and the speaker and for inserting into a human ear for a handsfree operation of the ear-device. According to further embodiments, a sensitivity level or a sensitivity ratio of: a) detecting the human tissue vibrations using the tissue conducting sensor such as the electrode transducer, and b) detecting the acoustic (external) vibrations using the microphone, can be adjusted. Furthermore, a volume and a frequency content of the acoustic signal coupled to said human ear by the speaker can be also adjusted, e.g., by using the plurality of sensors optimized for different frequency ranges. Also using multiple (segmented) sensors can cover more area of the human tissue and improve sensitivity of detection.

According to a further embodiment, the ear-device can comprise an electronic processing module (e.g., digital signal processor) for supporting functionalities of the tissue conducting sensor, the microphone and the speaker. Alternatively a memory module can be included in the ear-device if data storage is required. Moreover, the ear-device can have a battery for supporting an operation of all components of the ear-device requiring an electric power (alternatively the ear-device can have a wiring connection to the external electronic device or another electric power source).

Typically, the ear-device can be a part of an external electronic device (e.g., a mobile phone) and can be configured for detaching from the electronic device for inserting into the human ear and for attaching back (e.g., using a magnetic structure, snaps, etc.) to said electronic device, e.g., for recharging the battery. In one embodiment, when attached to the electronic device, the ear-device can be configured to provide the handsfree operation with more details provided in regard to FIG. **4***a*. Moreover, the ear-device can comprise a wireless module (e.g., BLUETOOTH, radio transmitter/receiver, etc.), for providing a wireless communication with the electronic device. Alternatively, the ear-device and the electronic device can have a connection through a non-wireless (e.g., cable) connection.

FIGS. **1**, **2***a*, **2***b*, **3** and **4***a* show examples among others of schematic representations for an ear-device **10**, according to embodiments of the present invention.

FIG. **1** shows an example among others of a block diagram of the ear-device **10**, according to an embodiment of the present invention. The ear-device **10** can comprise a housing **24** holding other modules: an electrode transducer (or in general a tissue conducting sensor) **16**, a microphone **14**, a

speaker **12**, an electronic processing block (e.g., a digital signal processor), a battery **22** and a wireless module **20**. The electrode transducer **16**, according to embodiments of the present invention, is described in more detail in regard to FIGS. **4***a* through **4***c*.

The tissue conducting sensor **16** can comprise of a plurality of segmented sensors **16***a* (e.g., see FIGS. **2***a* and **2***b*, or having a ring shape), each of these segmented sensors **16***a* can be optimized for a different frequency range. The sensor **16** can be primarily used for picking up the user's own speech (and/or other sounds associated with human movement such as, for instance, tapping or knocking) and may be realized, besides electrode transducer implementation disclosed in regard to FIGS. **4***a*-**4***b*, by using e.g., piezoelectric benders or other technologies. These segmented sensors can respond to tissue vibrations. The benders may be located at the outer surface of the circularly shaped device (see FIG. **2***b*), e.g., on a outer surface of the housing **24**. The sensors (including electrode transducers described herein) can be coated with a soft and comfortable impedance-matching layer **25** (for efficient and gentle coupling to an interior user ear), such as presented in International Patent application WO2004066669A2, "Anisotropic Acoustic Impedance Matching Material" by M. C. Bhardwaj. Also, the matching layer can be made of other suitable materials, e.g., a silicone rubber (soft elastic rubber). Piezoelectric components may also need an FET (field-effect transistor) at the electrodes of segmented sensors for electric impedance matching. Other methods which can be used for implementing the sensor **16** can include (but are not limited to) using: a capacitive method (e.g., used in electrode transducers of FIGS. **4***a*-**4***c*), miniature accelerator meters, etc.

The microphone **14** and the speaker **12** may be of a standard type and can be located at the ends (e.g., opposite ends) of the ear-device (e.g., see FIGS. **2***a* and **2***b*). FIGS. **2***a* and **2***b* show an example among many others of schematic representations (a side view and a 3-dimensional view), respectively, of the ear-device **10** of a round shape, with one end that is located inside the ear canal having a miniature speaker **12** and having the tissue conducting sensors **16***a* on the outer surface with a microphone **14** on the other end, according to an embodiment of the present invention. It should be noted that the FIGS. **2***a* and **2***b* are exemplary only and do not necessarily represent the final implementation. The speaker **12** and the microphone **14** may, for instance, be located at the same end. In such a case the audio signal can be fed to the human ear canal via a narrow tube through the device structure.

The microphone **14** can be used for picking up the external sounds. In addition, it may be used together with the tissue conducting sensor **16** for picking up the user's speech and/or to improve overall frequency response. It is known that the sensor **16** can work in a fairly narrow frequency range (e.g., in the range 2.5-3 kHz). A conventional microphone **14** may then be used to complement the frequency range up to the desired frequency, e.g., by simultaneously using the microphone **14** and the sensor **16**. Another alternative for detecting a wider frequency range would be to optimize each of these segmented sensors **16***a* for different acoustic frequencies complimentary to each other (this can be also in addition to using the microphone **14**).

The speaker **12** radiates an acoustic wave into the human ear canal. Due to a low acoustic leak in the system (the speaker **12** can be in close proximity to the ear canal as shown in FIGS. **2***a*, **2***b* and **3** and its acoustic patent can be directional), the base response can be made excellent. The current hands-free solutions do have problems with the low-frequency response, but the solution described herein can solve the frequency response problem as well.

The electronic processing module **18** (e.g., a digital signal processor) can be included in the ear-device **10** to perform some signal processing to support various functionalities, which are discussed in more detail in regard to FIG. **6**. The required processor may, for instance, be located in the center of the ear-device **10** (see FIG. **3**). Power for the processing module **18** can be drawn from the battery **22** or externally, e.g., the mobile device (then there is no own battery in the ear-device **10**). Alternatively, the ear-device **10** may solely rely on the processor located in the mobile device or on a combination of both processors (could be a more cost-effective solution). The module **18** can be responsive to commands from the user (e.g., through a user interface implemented in the mobile device or possibly in the ear-device **10**) to set and/or change functionalities and performance characteristics of the modules **12**, **14** and **16** for a specific application (see FIG. **6**). According to an embodiment of the present invention, the block **18** can be implemented as a software or hardware module or a combination thereof. Furthermore, the block **18** can be split into several blocks according to their functionality. The ear-device **10** can also have a memory **23** for storing, e.g., recorded information and/or music files.

The shape of the ear-device **10** can be circular, conical, U-shape, etc., or custom tailored to a particular user in order to properly fit in the human ear canal. The device may also include an easy-to-use method for inserting the device into the ear. The method of insertion is important from the usability point of view. An example showing the ear-device **10** inserted into the human ear is shown in FIG. **3**.

FIG. **4***a* shows a further example among others of a schematic representation of the ear-device **10** with a conical shape electrode transducer **16** with, e.g., a capacitive detection mechanism (other mechanisms, such as piezoelectric or utilizing miniature accelerator meters as described herein, can be also used), according to a further embodiment of the present invention. In this example the ear-device **10** can serve two operational modes: when attached to the external electronic device (phone) **40** (see FIG. **5**, as further discussed below), as ordinary mode (OM), and when detached and placed into the human ear in a handsfree tissue mode (HTM) as an autonomous miniaturized device. When in the OM, the microphone can "listen" the air using the air coupled microphone **14** and also possibly using the electrode transducer **16** (which is primarily for tissue conducting detection) as well. In the latter case, a plurality of sensors **16***a* which are parts of, e.g., the electrode transducer (capacitive sensor) **16** can be decoupled from the soft layer **25** (e.g., a soft elastic rubber) to provide good sensitivity to the air. In FIG. **4***a*, the segments (capacitive electrodes) **16***a* have a shape of a ring and can be made of a magnetic material as shown in more detail in FIG. **4***b* with a sensor surface area **16***d* comprised of a resonator cavity **16***b* with an opening **16***e* towards the skin. The segmented sensors **16***a* (shown as C**1**, C**2** and C**3** in FIG. **4***a*) can be optimized to work in different frequency ranges (e.g., 300 Hz<C3<500 Hz, 500 Hz<C2<1 kHz, 2 kHz<C1<7 KHz, etc.) for providing wider overall frequency range. The surface resonator cavity **16***b* can be sensitive to a predetermined acoustic frequency range and couple its vibrations through a capacitive change to an electrode (e.g., metallic) **16***c* of a ring shape. The frequency response of the sensor **16***a* can depend (among other factors) on sensor dimensions, location, geometry, mechanics (e.g., diameter of the ring, size/shape of the resonator cavity, ratio of the opening **16***e* and the sensor surface area **16***d*, etc.). The surface resonator cavity **16***b* can be located, e.g., in a vicinity of a human tissue without a direct

physical contact with said human tissue or with a partial physical contact (e.g., through the soft layer **25**). The resonator transducer **16** can perform as an acoustic mass-spring system having desired filtering characteristics. FIGS. **4***c* and **4***d* show 3-dimensional views of a ring (segmented sensor) **16***a*. FIG. **4***e* demonstrates a spiral implementation of the sensor **16***a*. Other implementations of the sensor **16***a* (e.g., a simple line) can be used.

When the ear-device **10** is in the HTM detached from the mother-phone **40** and placed into the user ear, the capacitive sensors **16***a* are coupled to the soft embodiment and further to the interior user ear to provide direct coupling to the human tissue. For example, the change of status of the sensor **16***a* can be made by having small magnets **48** in a mother-phone **40** (see FIG. **5**) which will lift up the capacitive electrode **16***a* (made of the magnetic material) from the soft layer **25** when the ear-device **10** is at the mother-phone **40** by making a connection of the detaching/attaching contacts **32** with the magnets **48**. The magnets **48** may be integrated into a mother-phone holding part and when the ear-device **10** is placed at the phone **40**, the electrodes **16***a* are decoupled and the sensor **16** can "listen" the air. The sensors/electrodes of the capacitive electrode transducer **16** can have a cylindrical shape to provide passing through capability of the audio signals as well.

The modules **12**, **14**, **18** and **20** shown in FIG. **4***a* are described in detail in regard to FIGS. **1**, **2***a*, **2***b* and **3**. However, the ear-device **10** in FIG. **4***a* includes a miniaturized speaker **12** placed at the side of the device **10** furthest from the human ear. Since the electrode transducer **16** has, e.g., the conical shape among other options, the sound generated by the speaker **12** in FIG. **4***a* can pass through to the human ear. Such combination of miniaturized mechanisms and sensors can provide a portable device Phone-Hosted Detachable Tissue conducting Microphone-Speaker Handsfree module (PHMS). Such PHMS device module may be hosted at the phone (portable device), detached and placed into the human ear when the user needs, e.g., "silent communications" or other applications described in regard to FIG. **6**, according to various embodiments of the present invention.

While hosted at the mother phone **40**, the PHMS can serve as a phone speaker and also it can recharge its own battery **22** from the phone's main battery **47** (see FIG. **5**).

According to another embodiment, a usage scenario can be that while operating as an HTM independent module, the PHMS can be easily and shortly attached to the phone to pick-up the energy (i.e., recharge the battery) and then placed again in the ear and continue, e.g., the "silent communication" mode. This operation is rather easy for the user and can be made even more frequently without drastic disturbances of the continuous communication (for example, once in 15-20 minutes).

Furthermore, openings **30** in the housing **24** (the housing **24** can be made of a hard polymeric material) to provide audio properties of the ear-device **10** in the OM. These openings **30** can also be utilized in the HTM mode for facilitating various applications (e.g., bin-aural recording) described in FIG. **6**. Further, according to another embodiment of the present invention, a size of the openings **30** can be adjusted according to a need (i.e., to vary the acoustic isolation), e.g., by interposing a further cap (not shown) with a predetermined pattern of further openings with the openings **30** on a surface of the housing **24** comprising the openings **30**, and continuously varying the size of the openings **30** by moving (rotating) this further cap.

FIG. **5** shows an example among others of a block diagram of an external electronic device **40** (e.g., a mobile phone) which can be a host (mother-device) for the ear-device **10**, according to an embodiment of the present invention. The device **40** can comprise a user interface module **42**, so the user can provide an appropriate command to facilitate an appropriate application (see FIG. **6**) of the ear-device **10** (some commands can be also communicated through the user interface on the ear-device **10**, if available). These commands are then forwarded to a processing unit **44** (which can be, e.g., a part of a central processing unit, CPU) and then to a wireless transceiver **46** for communicating with the ear-device **10**. According to an embodiment of the present invention, the block **44** can be implemented as a software or a hardware block or a combination thereof. Furthermore, the block **44** can be implemented as a separate block or can be combined with any other standard block of the electronic device **40** or it can be split into several blocks according to their functionality.

The device **40** can also comprise an ear-device cradle **45** with the magnets **48** (described herein) for attaching and detaching the ear-device **10**, and possibly a main battery **47** for recharging the battery **22** of the ear-device **10**.

FIG. **6** shows an example among others of a diagram demonstrating different applications utilizing the ear-device with main modules **12**, **14** and **16** (rectangles) used for different application/functionalities (ellipses) described herein, according to embodiments of the present invention.

There are a growing number of people willing to protect their hearing during a loud event (e.g., a rock concert). For this purpose, insert-ear-defenders are commonly used. According to embodiments of the present invention, this earplug can be made as a smart ear-device. The ear-device can be used for an active control of a music volume and a frequency content. The user may be able to adjust the volume so that the music is at a comfortable level. On the other hand, the tissue sensor can detect the user speech or other sounds "internally" through the human tissue vibrations, as described herein, such the user can communicate with the outside world in a noisy environment. Moreover, since there is a microphone in the device, it may also be used for recording the concert as heard by the listener bin-aurally to a mobile device in order to create a personal content. Moreover, one could possibly provide the concert to another user via a wireless link. Also, the ear-device, according to an embodiment of the present invention, can provide a decoding process (e.g., using the processing module **18**) and media player capabilities. It is noted that if the hearing protecting or bin-aural recording is used, the user should have the ear-device **10** in both ears.

Moreover, sometimes it is virtually impossible to make a phone call in a noisy environment. According to embodiments of the present invention, the ear-device utilizes tissue-conduction as a means to pick-up speech with a highly reduced background noise. The in-ear speaker with ear-defender functionality can provide a clear call reproduction and high intelligibility even in such an audio-hostile environment as a concert. Other applications/use cases may include teleconferencing to enable speaker localization to both directions. In addition to advanced features, the embodiments of the present invention can also support the basic functions, such as playback in noisy conditions. It can further support other ways of communications, not only by audio but by tapping/knocking by user finger or jaw or teeth (in mouth movement), as well as a hearing aid concept, etc.

According to embodiments described herein, the user does not need to use a loud voice while communicating using a very low threshold for a signal generation which provides a silent communication capability, such that only the user's voice is transferred to the other side (surrounding noise cancellation).

FIG. **7** is an example of a flow chart illustrating utilization of the ear-device **10**, according to an embodiment of the present invention.

The flow chart of FIG. **7** only represents one possible scenario among many others. The order of steps shown in FIG. **7** is not absolutely required, so generally, the various steps can be performed out of order. In a method according to an embodiment of the present invention, in a first step **50**, the ear-device **10** is inserted into a human ear.

In a next step **52**, a sensitivity level for detecting internal acoustic vibrations (by the electrode transducer/tissue conducting sensor **16**) and/or detecting external acoustic vibrations (by the microphone **14**) are adjusted. In a next step **54**, a volume of the acoustic signal of the speaker **12** is adjusted. In a next step **56**, the ear-device **10** is removed from the human ear and attached to an external electronic device **40** for recharging the battery and/or for an external use. In a next step **58**, the ear-device **10** is inserted into a human ear after recharging for a further use, and the process goes back to step **52**.

As explained above, the invention provides both a method and corresponding equipment consisting of various modules providing the functionality for performing the steps of the method. The modules may be implemented as hardware, or may be implemented as software or firmware for execution by a computer processor. In particular, in the case of firmware or software, the invention can be provided as a computer program product including a computer readable storage structure embodying computer program code (i.e., the software or firmware) thereon for execution by the computer processor.

Also, it is noted that various embodiments of the present invention recited herein can be used separately, combined or selectively combined for specific applications.

It is to be understood that the above-described arrangements are only illustrative of the application of the principles of the present invention. Numerous modifications and alternative arrangements may be devised by those skilled in the art without departing from the scope of the present invention, and the appended claims are intended to cover such modifications and arrangements.

What is claimed is:

1. An apparatus, comprising:

an electrode transducer, comprising at least one sensor, said at least one sensor comprising a surface resonator cavity with an opening sensitive to a predetermined acoustic frequency range, wherein said apparatus is configured for inserting it into a human ear for a handsfree operation and said at least one sensor is configured for detecting human tissue vibrations with said surface resonator cavity opening towards the human tissue.

2. The apparatus of claim **1**, wherein said electrode transducer comprises one or more sensors of said at least one sensor with one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters.

3. The apparatus of claim **2** wherein each of said plurality of sensors is optimized for a different acoustic frequency range.

4. The apparatus of claim **2**, wherein said electrode transducer comprises a soft material between said sensors for adapting to said human ear.

5. The apparatus of claim **1**, wherein said at least one sensor has a shape of a ring, line or a spiral shape.

6. The apparatus of claim **1**, wherein said electrode transducer is configured for a speech detection by detecting said human tissue vibrations.

7. The apparatus of claim **1**, further comprising an impedance-matching layer covering said at least one sensor for efficient and gentle acoustic coupling of said segmented sensors to said human ear.

8. The apparatus of claim **7**, wherein, when said apparatus is attached to an electronic device, said at least one sensor is disengaged from a contact with said impedance-matching layer.

9. The apparatus of claim **1**, further comprising:

a microphone, for detecting acoustic vibrations, wherein said apparatus is configured to adjust a sensitivity level or a sensitivity ratio of:

a) detecting said human tissue vibrations by said electrode transducer, and

b) detecting said acoustic vibrations by said microphone.

10. The apparatus of claim **1**, further comprising:

a speaker, for providing an acoustic signal.

11. The apparatus of claim **10**, wherein said speaker is configured to adjust a volume of said acoustic signal coupled to said human ear.

12. The apparatus of claim **10**, wherein said speaker is configured to adjust spectral content of said acoustic signal coupled to said human ear.

13. The apparatus of claim **9**, wherein said microphone is configured to provide at least one of: a) a two-way communication in normal or noisy conditions, b) bin-aural recording, c) a hearing protection for said human ear from external noises, d) volume enhancement and equalization as a hearing aid, and e) a playback capability in said normal or noisy conditions.

14. The apparatus of claim **1**, further comprises an electronic processing module for supporting functionalities of all or selected components of said apparatus.

15. The apparatus of claim **14**, wherein said processing module is configured to perform a decoding process such that said apparatus is further configured to provide media player capabilities.

16. The apparatus of claim **1**, further comprising at least one of: a) a battery for supporting an operation of all components of said apparatus requiring an electric power, and b) a memory for storing recorded files.

17. The apparatus of claim **1**, wherein said apparatus is a part of an electronic device and is configured for detaching from said electronic device for said inserting into the human ear and for attaching back to said electronic device.

18. The apparatus of claim **17**, wherein said apparatus, when attached to said electronic device, is configured to provide a further handsfree operation.

19. The apparatus of claim **17**, wherein said apparatus comprises a battery and said electronic device is configured to recharge said battery when said apparatus is attached to said electronic device.

20. The apparatus of claim **1**, wherein said apparatus is connected to an electronic device and said apparatus further comprising a wireless module for providing a wireless communication of said apparatus with said electronic device, or said apparatus is connected by a wire to said electronic device.

21. The apparatus of claim **20**, wherein said electronic device is a wireless device, a portable communication device, a personal digital assistant or a mobile phone.

22. The apparatus of claim **1**, wherein said at least one sensor comprises a plurality of ring shaped sensor segments of different diameters formed on a soft, conically shaped material.

23. An electrode transducer, comprising:

at least one sensor, which comprises a surface resonator cavity with an opening sensitive to a predetermined acoustic frequency range, wherein said at least one sensor, when inserted into a human ear for a handsfree operation, is configured to detect human tissue vibrations with said surface resonator cavity opening towards the human tissue.

24. The electrode transducer of claim 23, wherein said at least one sensor comprises one or more sensors with each comprising any one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters.

25. The electrode transducer of claim 24 wherein each of said one or more sensors is optimized for a different acoustic frequency range.

26. The electrode transducer of claim 23, wherein said electrode transducer comprises a soft material between said sensors for adapting to said human ear and wherein said at least one sensor comprises a plurality of ring shaped sensor segments.

27. The electrode transducer of claim 23, wherein said at least one sensor has a shape of a ring, line or a spiral.

28. A method, comprising:

inserting an ear-device into a human ear for a handsfree operation, wherein said ear-device comprises: an electrode transducer comprising at least one sensor, said at least one sensor comprising a surface resonator cavity with an opening sensitive to a predetermined acoustic frequency range; and

detecting by said at least one sensor human tissue vibrations with said surface resonator cavity opening towards the human tissue.

29. The method of claim 28, wherein said ear-device further comprises a microphone, and the method further comprises:

detecting acoustic vibrations using said microphone, wherein said ear-device is configured to adjust a sensitivity level or a sensitivity ratio of:

a) detecting said human tissue vibrations by said electrode transducer, and

b) detecting said acoustic vibrations by said microphone.

30. The method of claim 28, wherein said ear-device further comprises a speaker providing an acoustic signal, and the method further comprises:

adjusting at least one of:

a) a volume of said acoustic signal coupled to said human ear, and

b) spectral content of said acoustic signal coupled to said human ear.

31. The method of claim 28, further comprising:

taking said ear-device out of said human ear and attaching said ear-device to an electronic device for a further handsfree operation or for recharging a battery of said ear-device.

32. The method of claim 31, wherein said ear-device further comprises a wireless module, and the method further comprises:

providing a wireless communication of said ear-device with said electronic device.

33. The method of claim 29, wherein said microphone is configured to provide at least one of: a) two-way communications in normal or noisy conditions, b) bin-aural recording, c) a hearing protection for said human ear from external noises, d) volume enhancement and equalization as a hearing aid, and d) a playback capability in said normal or noisy conditions.

34. The method of claim 28, wherein said electrode transducer comprises one or more sensors of said at least one sensor with one of: a) a capacitive detection mechanism, b) a piezoelectric detection mechanism, and c) a detection mechanism utilizing miniature accelerator meters.

35. The method of claim 28, wherein said at least one sensor comprises a plurality of sensors and each of said plurality of sensors is optimized for a different acoustic frequency range.

36. The method of claim 35, wherein said plurality of sensors comprises a plurality of ring shaped sensor segments formed on a soft, conically shaped material.

37. A computer program product comprising: a non-transitory computer readable storage structure embodying computer program code thereon for execution by a computer processor with said computer program code, wherein said computer program code comprises instructions for causing an apparatus to perform the method of claim 28.

38. A system, comprising:

at least one ear-device, comprising:

an electrode transducer comprising at least one sensor, said at least one sensor comprising a surface resonator cavity with an opening sensitive to a predetermined acoustic frequency range, wherein said apparatus is configured for inserting it into a human ear for a handsfree operation and said at least one sensor is configured for detecting human tissue vibrations with said surface resonator cavity opening towards the human tissue; and

an electronic device, for providing communicating acoustically generated signals to and from said ear-device.

39. The system of claim 38, further comprising:

a microphone, for detecting acoustic vibrations, wherein said apparatus is configured to adjust a sensitivity level or a sensitivity ratio of:

a) detecting said human tissue vibrations by said electrode transducer, and

b) detecting said acoustic vibrations by said microphone; and a speaker, for providing an acoustic signal.

40. The system of claim 39, wherein said at least one ear-device comprises two ear-devices and the two ear-devices, when inserted into both human ears, are configured for at least one of:

a) to provide bin-aural recording,

b) to provide a hearing protection for said human ears from external noises; and

c) to provide an adjustable hearing protection for said human ears from external noises.

41. The system of claim 38, wherein said at least one ear-device comprises a battery for supporting an operation of all components of said ear-device requiring an electric power, and said electronic device is configured for recharging said battery.

42. The system of claim 38, wherein said at least one sensor comprises a plurality of ring shaped sensor segments formed on a soft, conically shaped material.

43. An apparatus, comprising:

transducer means, comprising at least one sensor, said at least one sensor comprising a surface resonator cavity with an opening sensitive to a predetermined acoustic

frequency range, wherein said apparatus is configured for inserting it into a human ear for a handsfree operation and said at least one sensor is configured for detecting human tissue vibrations with said surface resonator cavity opening towards the human tissue.

44. The apparatus of claim **43**, wherein said at least one sensor comprises a plurality of ring shaped sensor segments formed on a soft, conically shaped material.

* * * * *

Exhibit OTH-B

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President, Founder

SuperAlert, Inc. **NEW YORK, NY**

SuperAlert has developed a range of wireless sensors used for monitoring building systems in large New York City residential buildings. SuperAlert manufactures, deploys, and operates sensor suites and generates notifications and alerts for building personnel and owners.

1980-present

President

NEW YORK, NY

Daedalus Technology Group, Inc. successor to NPS ASSOCIATES

Co-founded computer consulting firm. Employs up to twenty people as needed.

clients include

Intellectual Property Consulting. Various anonymous clients. **NEW YORK, NY**

Consulting on patent valuation and strategy: Act as an advisor on technical and intellectual property matters to various clients. Developed intellectual property for various companies. 2001- present.

SpokenHub. **NEW YORK, NY**

Predictive dialer deployment and operation: During the 2012 election cycle assembled and ran a team of engineers to deploy, debug, and operate a large scale outbound calling server farm. This server farm supported hundreds of agents and millions of minutes of calling. 2012.

Risk Solutions International. **NEW YORK, NY**

Risk management: Developed technology platform to allow for complex question and answer sessions with airport managers. The results of the sessions then generated planning documents used in risk management. Results were published as part of a Transportation Research Board research program. 2011-2013.

Smart Systems/Specialty Acquirer. **NEW YORK, NY**

Advanced Transit fare collection: Founding member of board of directors. Advisor on technical and intellectual property matters. Developed intellectual property for the company. The company is engaged in the development and sale of advanced fare collection systems for mass transit using RFID technologies. 2004-present.

Skywi. **FORT WORTH, TX**

Wireless broadband: Founding board of directors member. Advisor on technical and intellectual property matters. The company provides a variety of wireless broadband services to semi-rural customers using mesh networks. 2004-2009

Placecorp. **NEW YORK, NY**

Wireless text messaging system: Designed and developed a very large scale, multi-platform text messaging system. Acted as the lead technologist for a startup. In this capacity attended many venture capital meetings and developed all technical sections to the business plans. The system integrated advanced Interactive Voice Response (IVR) with email, paging, and SMS messaging. Oversaw and planned initial development of very small paging device to me manufactured in very high volume. 2000-2001.

Marketboy. **NEW YORK, NY**

Distributed market making system: Provided a range of services in the areas of product definition, specification, and implementation strategy. Marketboy was a distributed system that allowed prospective purchasers and vendors to make bid and ask offers for proposed transactions. 2000.

Savos. **NEW YORK, NY**

Telephone to streaming audio bridge: Developed, designed and deployed an industrial strength IVR system that allows users to access streaming media via their cell phones. The system supports up to 96 streams on a single chassis. Developed web based content management system and interface for customizable user preferences. 2000.

deliverENow. **NEW YORK, NY**

Online delivery system: Developed a proof of concept prototype of online delivery system. This consisted of a stock clerk server written for Windows CE (Clio) with a CDPD network connection. It demonstrated how purchase and delivery information would be sent to an in-store stock person who prepares them for pickup by a member of the deliverENow courier network. Provided strategic technical advice. Served on board of advisors. Wrote technical specification of business and logistical components of the system. 2000.

Togglethis. **New York, NY**

Animation distribtion system: Developed critical components of initial version of email-delivered animation system. Components included MacroMedia Director Xtras as well as encryption functions. Developed tools to distribute and manage Togglethis's content, DTG built a custom UNIX email list-server. This server allowed Toggle administrators to create groups, upload episodes, and schedule the delivery of their content. Developed system architecture and language specification for the next version of the IC Engine, which drives the characters. 1998-1999.

JuniorNet. **BOSTON, MA**

Children's on-line service: Designed, developed and lead the implementation of a large scale network system for delivery of children's entertainment and education content. Worked with the president and other key players to fit a technology strategy to corporate goals. System is required to serve from 200,000 to 1,000,000 subscribers and be supportable over three to five years. All project goals met within a $1,000,000+ budget. 1997-2000.

One Click Charge. **NEW YORK, NY**

Provide a wide range of consulting services starting with technology evaluation for the principal investor. Provided services in the areas of product definition, team formation and implementation strategy. Developed significant components of a substantial Internet micropayment and authentication system. 1998-2000.

Swatch. **WEEHAWKEN, NJ**

Ticketing system: Implemented a radio frequency identification (RFID) based system for providing access to the 1998 Goodwill Games utilizing RFID tagged wristwatches. 1998-1999.

Instant Video Technologies, Inc./Burst.com **SAN FRANCISCO, CA**

Burstware --. Designed and developed a series of products to deliver digital video and audio materials over wide-area internet using consumer grade computers. Products were designed to fit within the client's existing patent portfolio. 1995-1997.

Technical evaluation --. Provided technology strategy and evaluation services and acted as Technology Director. 1995-1997.

Assurenet. **BETHESDA, MD**

Internet Insurance system --. Designed and developed a system for utilization of the Internet for the distribution and control of insurance agent support software. System can handle tens of thousands of simultaneous users. Member of Board of Directors. 1995-2002.

New Zealand Antarctic Project (NZAP) and US Antarctic Program (USAP). **ANTARCTICA**

Penguin Weighbridge - Designed, developed, and deployed system for tracking the comings and goings, weights, and Ids of Adelie penguins in their natural environment on three colonies on Ross Island, Antarctica. Systems run continuously for months in adverse conditions. System was upgraded and replaced in 2016. 1994-present.

Electronic Digital Documents, Inc. **NEW YORK, NY**

Check Image Compression system --. 1995-1999.

American Veterinary Identification Devices, Inc. **NORCO, CA**

AVID reader -- Designed and developed digital computer components of radio frequency identification system. Work involved imbedded microcontrollers and real-time digital signal processing in very noisy environment. Designed communications protocols including encryption and error checking schemes. Product in high volume (50,000 unit) production. 1988-1995.

Amprobe Instrument, Inc. LONG ISLAND, NY

REMCON tester -- Designed and implemented product life-cycle test fixture for a new and innovative solid state switch. Test fixture used by Underwriters Labs for approval. 1988.

Personal Computer Card Corporation NEW YORK, NY

Smart Cards -- Designed smart card reader/writer system. Performed general design and component selection and wrote software for smart card communications. System design implemented in commercial product. 1987.

The Dun & Bradstreet Corporation NEW YORK, NY

Voice editing systems -- Designed and built all parts of four generations of interactive voice editing systems. Editing systems produce output used in extensive, high quality speech synthesis system that is part of larger information delivery system. The systems are built on networks of computers in a variety of programming languages and systems. The system involved interactive user interfaces, screen windows, signal processing, and voice file systems. 1982-1991.

1997-2001 **Co-Founder, Director and CTO**

I-RECALL,INC. NEW YORK, NY

Co-founded a company that developed software to enable note-taking and linking of notes to streaming media. Used platforms such as PalmPilot and CrossPad for note-taking and provided time-stamped links to audio and video media. Company sold to a focus group company.

1996-1999 **Co-Founder, Director, CTO & Principal Product Designer**

SOLILOQUY,INC. NEW YORK, NY

Natural language interfaces to databases: Co-founded in 1996 a company to provide next generation interfaces to structured databases. Developed prototype system using speech recognition and speech synthesis to provide access to music and ecommerce databases. Raised $1.5 million in angel investments. Company has raised a total of $8 million through a variety of institutional and individual investors. Hired replacement CTO and exited company in 1999.

1991-1992 **President & Principal Product Designer**

SIMPLICITY COMPUTING,INC. NEW YORK, NY

Co-founded in 1991 a computer peripherals manufacturing company concentrating on portable products with international mass-market appeal. Main product: Simplicity Portable Drive, an IBMPC parallel port connected, portable disk storage system. Marketed domestically and in 25+countries. Products marketed direct via telemarketers, trade shows, sales reps, and national advertising. Products distributed through dealers, distributors and national catalog houses. Company employed 10 people.

- Wrote business plan and raised $350,000 startup funding
- Established marketing plans
- Directed marketing of products in trade shows, national advertisements and focus groups
- Managed employees
- Designed hardware and software for products
- Supervised hardware & software implementation teams
- First units shipped less than six months from start of project
- Supervised all aspects of manufacturing and testing process
- Contracted with four manufacturing facilities and many suppliers for production
- Supervised and established call-in customer support services

1989-1999 **President & CEO**

MEASUREMENT & CONTROL PRODUCTS, INC. NEW YORK, NY

Founded in August 1989 an electronic products development company. M&CP designs, develops, markets and sells its own products through retail, wholesale and catalog channels domestically and abroad. All manufacturing is contracted. Company was profitable in its first year of operation.

Flagship product:

BitView -- Hand held RS-232 data communications diagnostic monitor. Invented, developed and produced hardware and software. Developed and oversaw implementation of marketing strategies, budgets, and business plans. Product currently in third production run.

education

1980-1993 **COLUMBIA UNIVERSITY** NEW YORK, NY

Ph.D. in Computer Science, May 1993,
Thesis: *Mixed Distance Measures for the Optimization of Concatenative Vocabularies in Speech Synthesis.*
MPhil in Computer Science, December 1989.
MS in Computer Science, December 1987.
BA in Physics, Columbia College, May 1984.

teaching positions

1997 **COLUMBIA UNIVERSITY** NEW YORK, NY

Adjunct Professor, Computer Science.

Advanced undergraduate course: Artificial Intelligence.

1989 **COLUMBIA UNIVERSITY** NEW YORK, NY

Graduate Lecturer, Computer Science.

Advanced undergraduate course: Software Design. Taught 68 advanced undergraduates. Supervised two teaching assistants. Students implemented full spreadsheet application under UNIX.

1984-1985 **CITY UNIVERSITY OF NEW YORK** NEW YORK, NY

Adjunct Professor, Experimental Psychology.
Graduate-level course: Computer Methods in Experimental Psychology.

publications and patents

2010 Michael L. Beigel, Nathaniel Polish, Steven R. Frank, and Robert E. Malm. Electronic Identification System with Improved Sensitivity. United States Patent 7,737,821B2. June 15, 2010.

2009 Silbernagl, Martin and Polish, Nathaniel. Learning Fare Collection System for Mass Transit. United States Patent 7,568,617. August 4, 2009.

2009 Beigel, Michael L.; Polish, Nathaniel; Frank, Steven R.;Malm, Robert E. Electronic identification system with improved sensitivity. United States Patent 6,472,975. October 29, 2002.

2002 Nathaniel Polish. Bilateral speech system. Speech dialogs for database access. United States Patent 6,430,531. August 6, 2002.

1999 Nathaniel Polish. System and method for distributing and managing digital video information in a video distribution network. United States Patent 5,963,202. October 5, 1999.

1993 Michael L. Beigel, Nathaniel Polish and Robert E. Malm. Multi-Mode Identification System. Fundamental RFID technology. United States Patent 5,235,326. August 10, 1993.

1991 Nathaniel Polish. Mixed Distance Measures for Synthetic Speech Evaluation. In Proceedings of ICASSP-91, Toronto, Canada, 1991.

1988 Nathaniel Polish. A Distributed Signal Processing Facility for Speech Research. In Proceedings of AVIOS88, San Francisco, CA. October 4-6 1988.

1987 Nathaniel Polish. Mixed Distance Measures for Optimizing Concatenative Vocabularies for Speech Synthesis: A Thesis Proposal. Columbia University Department of Computer Science technical report number CUCS-310-87, 1987.

1983 M. Morris, N. Polish, B. Zuckerman, and N. Kaifu. The Temperature of Molecular Gas in the Galactic Center Region. The Astronomical Journal, 88(8):1228-1235, August 1983.

research interests Computer Speech, Distributed Systems, Interactive Environments. Software Engineering, Systems Engineering, Technology Policy. Knowledge-Based Approaches to Signal Processing. Computational Physics, Non-Linear Dynamics.

professional societies International Society for Electrical and Electronics Engineers (IEEE), Association of Computing Machinery (ACM), American Association for the Advancement of Science (AAAS).

other interests Foreign Policy, Political Science, Long Distance Bicycling, United States Space Program.

grants

- AT&T Special Projects Grant of $25,000 for synthetic speech, 1986.
- Financial support for Ph.D. work was provided by the New York State Science and Technology Foundation Center for Advanced Technology in Computers and Information Systems at Columbia University.
- Santa Fe Institute Complex Systems Summer School, 2003. Month-long workshop in agent-based models, non-linear systems, and complexity.

1982-present

Technology development projects

Some technology development projects have included:

Wildlife Tracking Systems -- Implemented a remote monitoring system to track and weigh Adelie penguins on Ross Island, Antarctica. System was debugged on-site and made to function in adverse environmental circumstances. 1994-1997, 2016.

Internet Systems -- Implemented a LINUX Internet server with FTP and gopher services. Installed and Internet client for IBM/PC platform. 1994-1995.

Image Compression Technology -- Developed proof-of-concept system for a system to compress images of checks to very small file size. 1994.

Biomedical Systems -- Implemented a large biomedical system used in patient diagnostics. Provided all necessary materials for Food and Drug Administration compliance. 1992-1993.

Synthetic Speech Systems -- Implemented and supported several generations of synthetic speech systems for commercial as well as research purposes. 1985-1999.

Graphics -- Developed high-speed drivers for several graphical devices and evaluated their applicability for interactive uses. Devices included: Vectrix, RAMTEK, Apple Lisa, and IBM Enhanced Graphic Adapter. 1983-1987.

Voice Boards -- Specified function requirements for several generations of high quality voice boards for Q-bus (PDP-11), Apple, MultiBus, and IBMPC bus. Developed high-speed drivers for each of the boards. 1982-1986.

Video Disk and Touch Screens -- Developed experimental interactive system utilizing computer controlled video disks and touch screens. 1982.

1994-present

Intellectual property projects

A separate document detailing intellectual property projects is available on request.

tools

Languages commonly used:
Pascal 1980-1988
C 1985-present.
Other languages used: C++, Lisp, Java and Python .
Assembler:
Intel 8048, 8051, 8088-pentium
Motorola 680xx
TI 320xx signal processors
Zilog Z8, Z80, PIC
Protocols include:
TCP/IP, UDP, RPC and NFS
Operating systems commonly used:
UNIX, MSDOS, Windows and p-system.
In circuit emulators.
Database tools.
ARM processors for embedded systems.
I^2C bus interfacing.

contact information

polish@dtgroup.com
Office: (212) 684-3890

Academic and business references available upon request.

Exhibit PAT-A



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

(10) **Patent No.: US 11,244,666 B2**
(45) **Date of Patent: Feb. 8, 2022**

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

(71) Applicant: **Staton Techiya LLC**, Delray Beach, FL (US)

(72) Inventors: **Steven Wayne Goldstein**, Delray Beach, FL (US); **John Usher**, Beer (GB); **Marc Andre Boillot**, Plantation, FL (US)

(73) Assignee: **Staton Techiya, LLC**, Delray Beach, FL (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.

(21) Appl. No.: **16/987,396**

(22) Filed: **Aug. 7, 2020**

(65) **Prior Publication Data**

US 2020/0365132 A1 Nov. 19, 2020

Related U.S. Application Data

(63) Continuation of application No. 16/669,490, filed on Oct. 30, 2019, which is a continuation of application No. 16/193,568, filed on Nov. 16, 2018, now Pat. No. 10,535,334, which is a continuation of application No. 14/574,589, filed on Dec. 18, 2014, now Pat. No.
(Continued)

(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 1/1016; H04R 1/1083; H04R 3/002; H04R 3/005; H04R 29/001; H04R 2225/41; H04R 2460/05; H04R 2499/11
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,208,867 A * 5/1993 Stites, III H04R 25/407 381/361
5,541,359 A 7/1996 Lee
(Continued)

FOREIGN PATENT DOCUMENTS

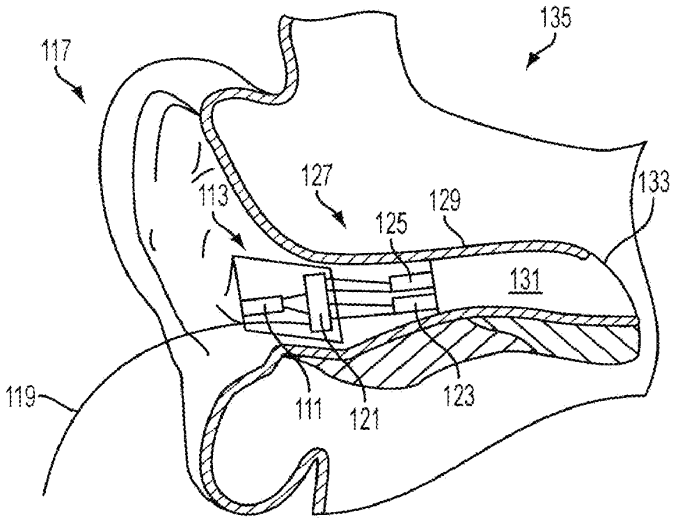
CN	101401399	4/2009
JP	5299030	9/2013

Primary Examiner — Andrew L Sniezek
(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti

(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.

11 Claims, 9 Drawing Sheets



Related U.S. Application Data

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.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,774,567	A	6/1998	Heyl	
6,023,517	A *	2/2000	Ishige	H04R 25/505 381/312
6,226,389	B1	5/2001	Lemelson et al.	
7,421,084	B2	9/2008	Jubien et al.	
7,903,826	B2	3/2011	Boersma	
7,986,802	B2	7/2011	Ziller	
8,199,942	B2	6/2012	Mao	
8,493,204	B2	7/2013	Wong et al.	
8,514,100	B2	8/2013	Yamashita	
8,577,052	B2	11/2013	Silber et al.	
8,638,239	B2	1/2014	Cases et al.	
8,792,648	B2 *	7/2014	Kim	H04R 3/005 381/57
9,041,545	B2	5/2015	Zelepugas	
9,648,436	B2	5/2017	Kraft et al.	
10,361,673	B1	7/2019	Matsukawa	
2002/0135485	A1	9/2002	Arakawa	
2003/0035551	A1	2/2003	Light et al.	
2003/0151678	A1	8/2003	Lee et al.	
2004/0179694	A1 *	9/2004	Alley	H03G 3/32 381/55
2005/0207597	A1	9/2005	Kageyama	
2006/0222185	A1	10/2006	Dyer et al.	
2006/0253282	A1	11/2006	Schmidt et al.	
2006/0262938	A1 *	11/2006	Gauger	G10L 21/02 381/56
2007/0147635	A1 *	6/2007	Dijkstra	H04R 25/40 381/94.1
2008/0165988	A1	7/2008	Terlizzi et al.	
2008/0240458	A1 *	10/2008	Goldstein	H04R 1/1083 381/72
2010/0136950	A1	6/2010	Backlund et al.	
2011/0069845	A1 *	3/2011	Cohen	G01H 3/14 381/73.1
2014/0185828	A1	7/2014	Helbling	
2016/0249128	A1 *	8/2016	Goldstein	H04R 29/00
2017/0124847	A1	5/2017	Schulz et al.	

* cited by examiner

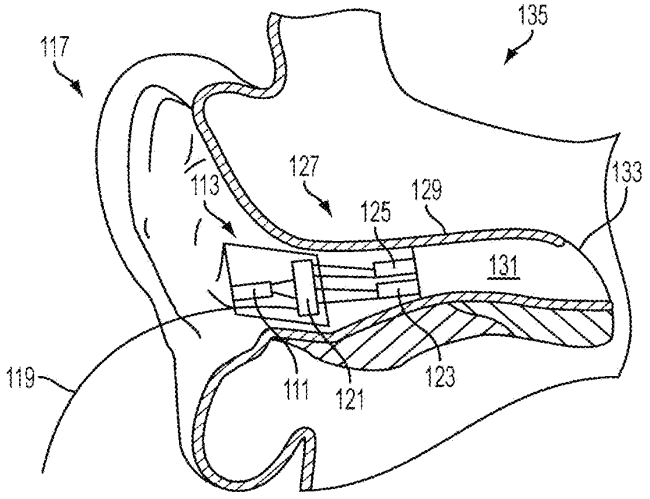

135
117
127
113
129
125
133
131
119
111
121
123

FIG. 1

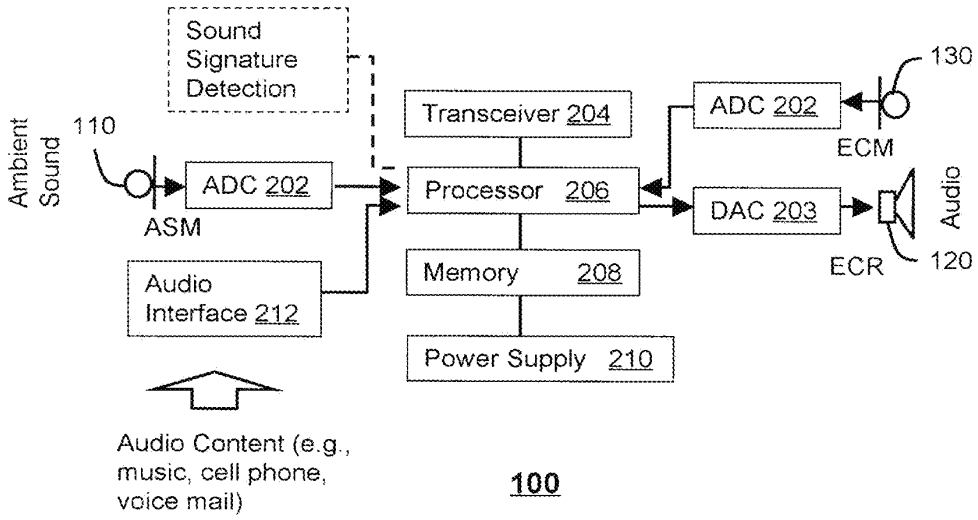
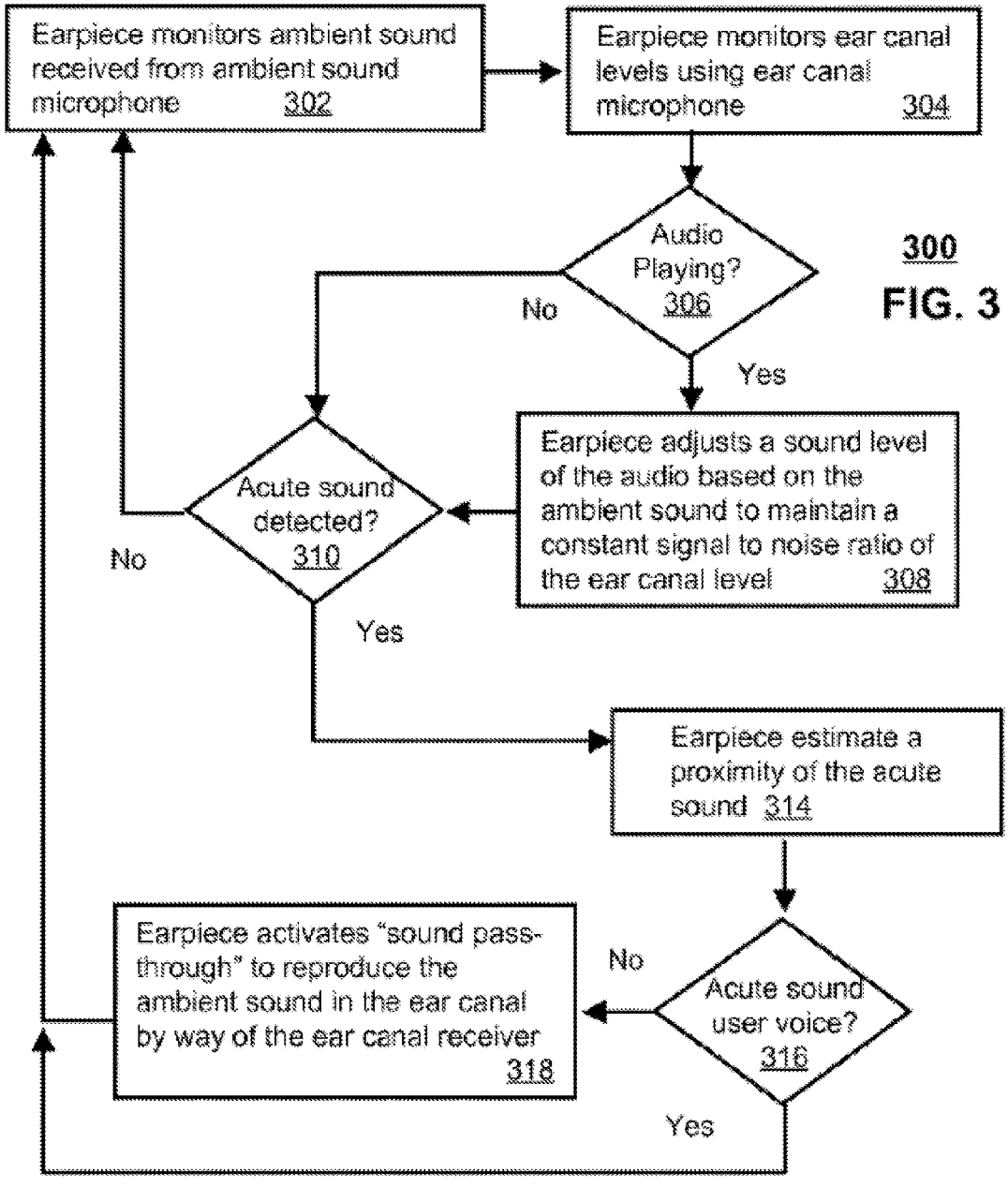

Sound Signature Detection
Transceiver 204
ADC 202
130
ECM
110
Ambient Sound
ADC 202
Processor 206
DAC 203
Audio
ASM
ECR
120
Audio Interface 212
Memory 208
Power Supply 210
Audio Content (e.g., music, cell phone, voice mail)
100

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

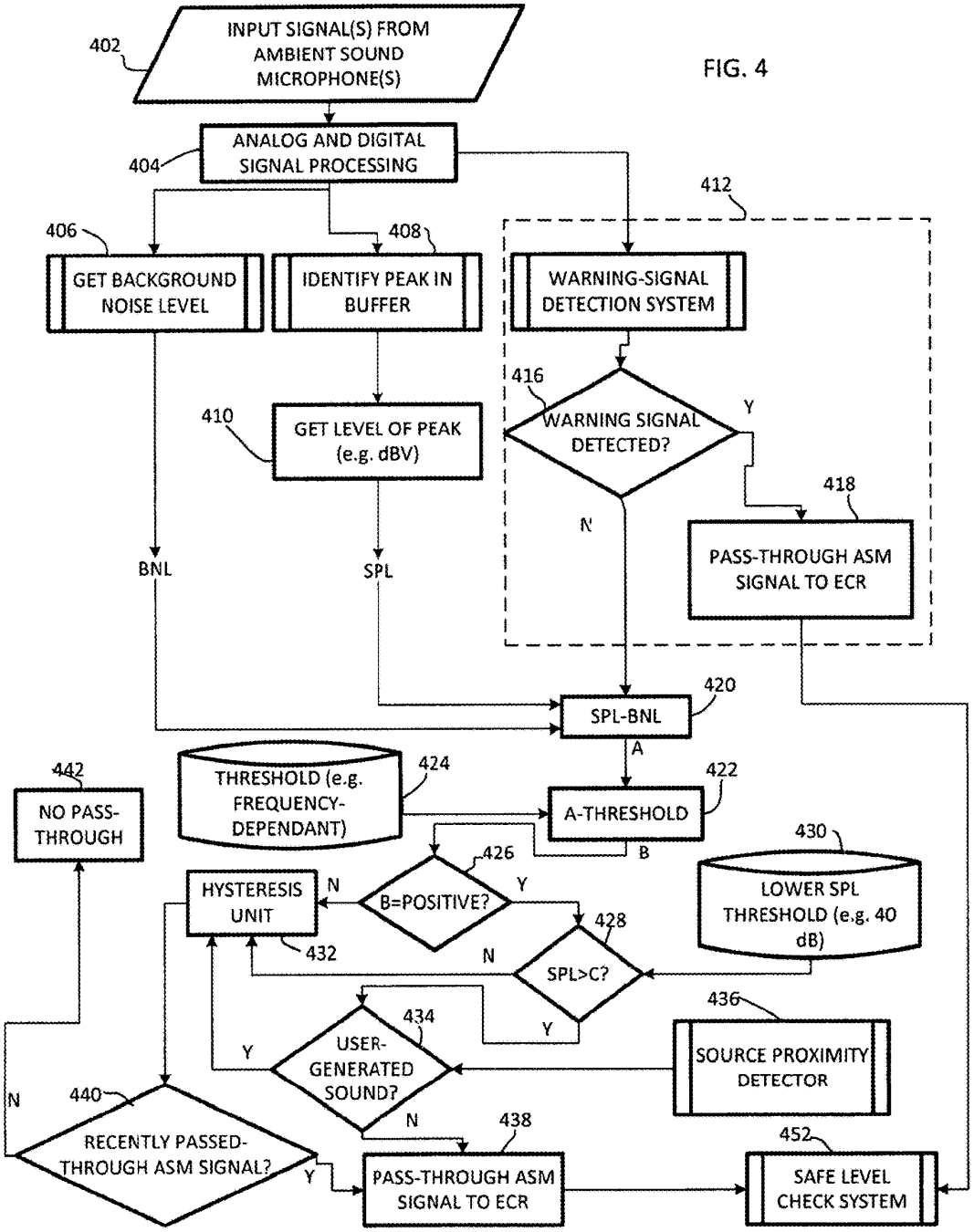
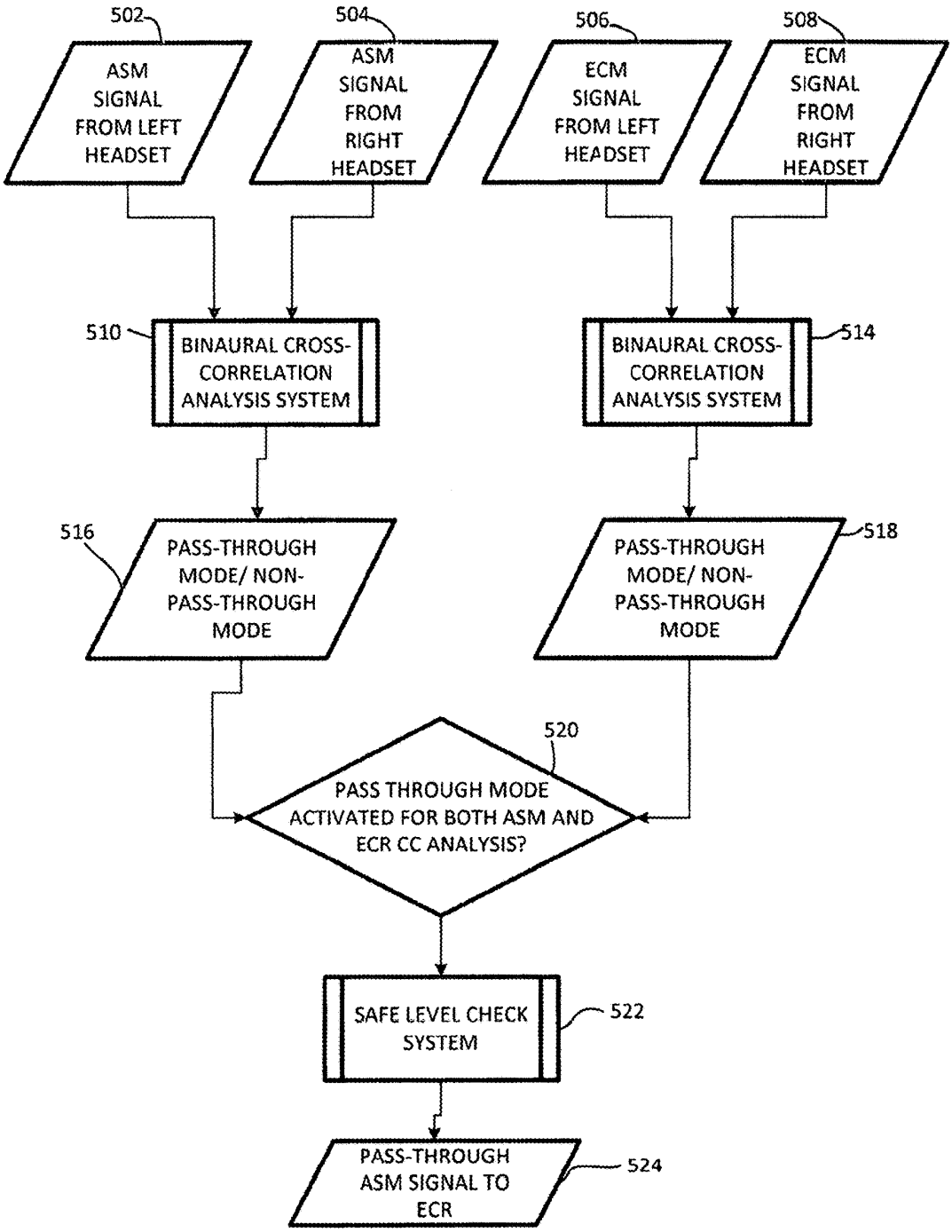

402
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONE(S)
404
ANALOG AND DIGITAL SIGNAL PROCESSING
406
GET BACKGROUND NOISE LEVEL
408
IDENTIFY PEAK IN BUFFER
410
GET LEVEL OF PEAK (e.g. dBV)
412
WARNING-SIGNAL DETECTION SYSTEM
416
WARNING SIGNAL DETECTED?
Y
N
418
PASS-THROUGH ASM SIGNAL TO ECR
BNL
SPL
420
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?
522
SAFE LEVEL CHECK SYSTEM
524
PASS-THROUGH ASM SIGNAL TO ECR

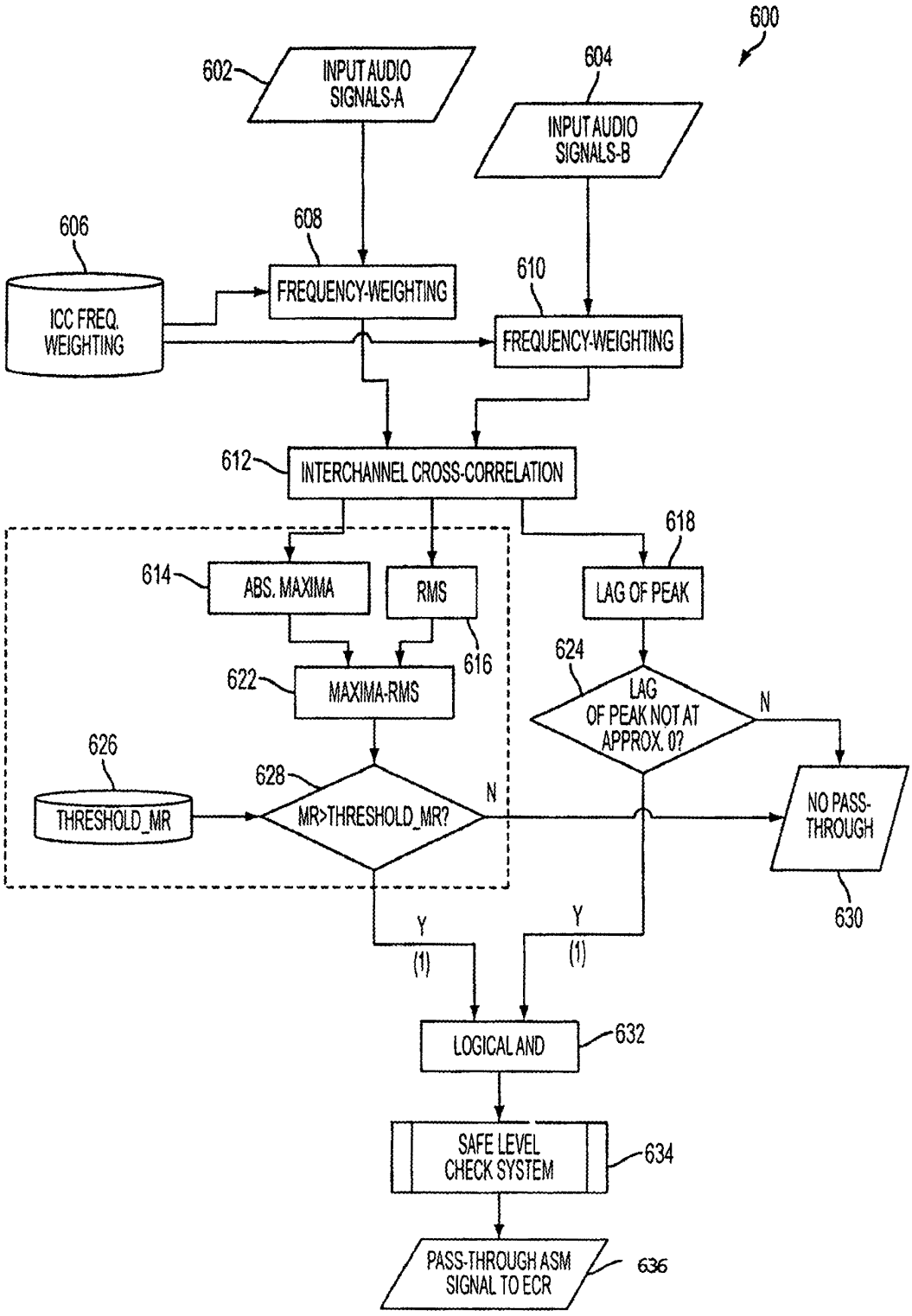
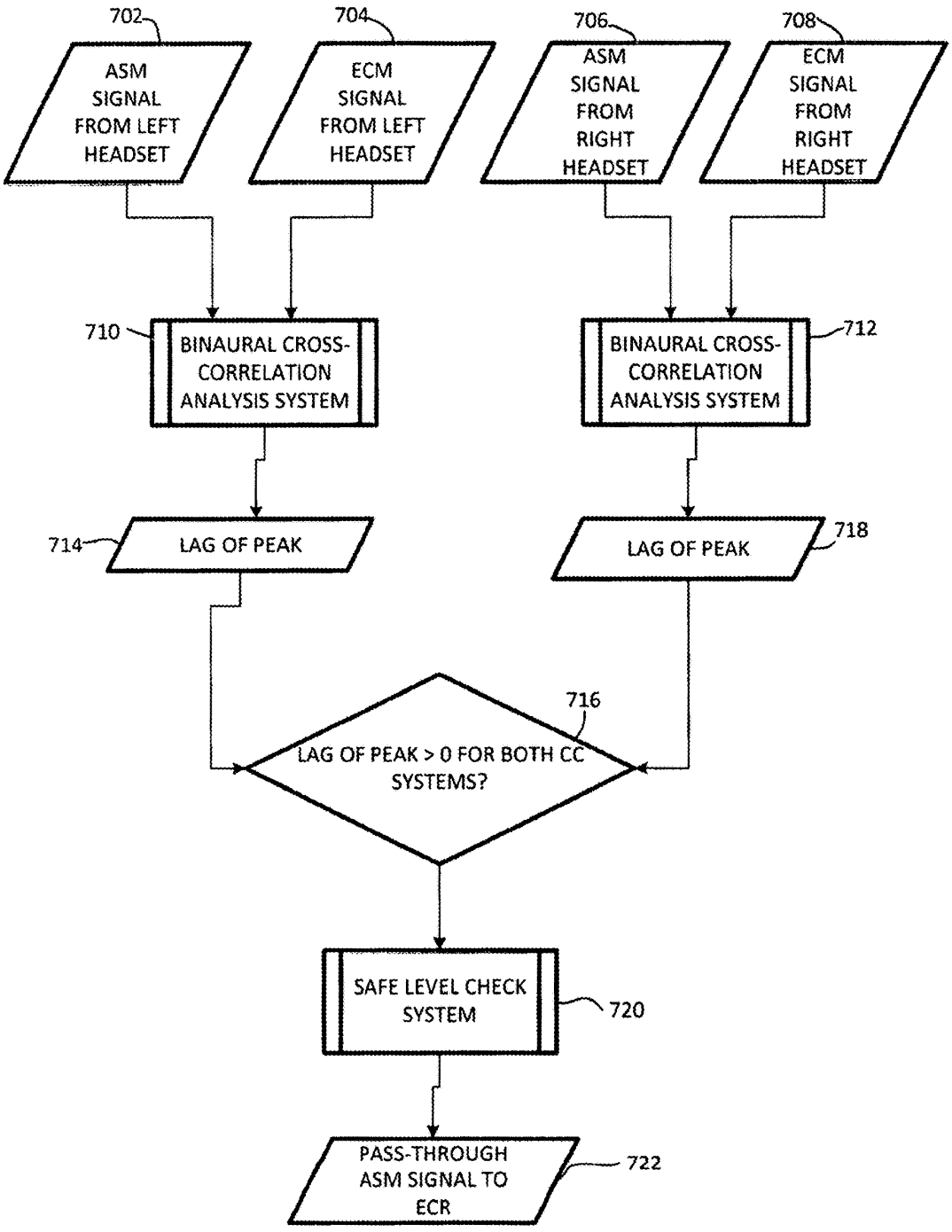

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH
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

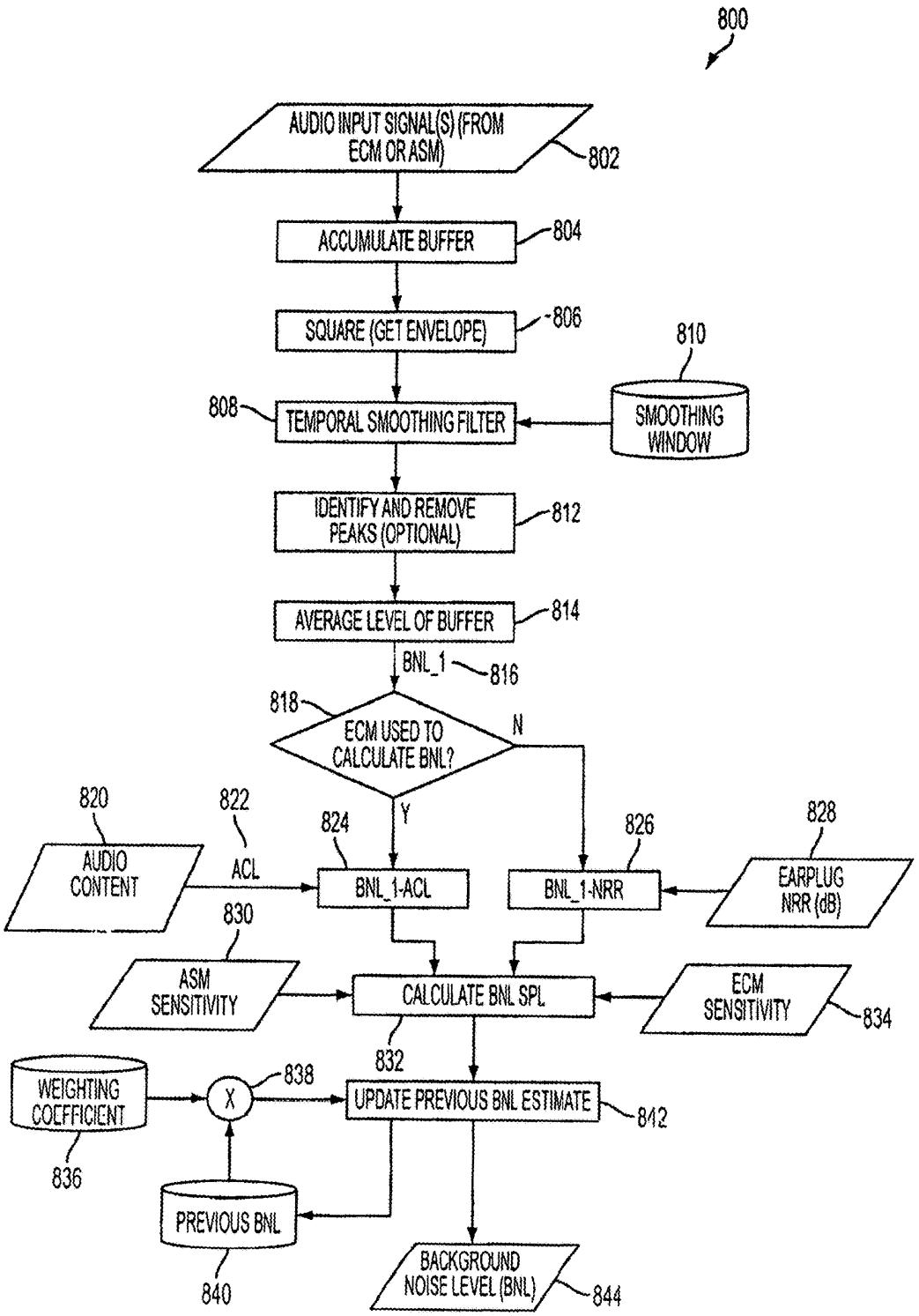

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT
822
ACL
824
BNL_1-ACL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

FIG. 8

FIG. 9

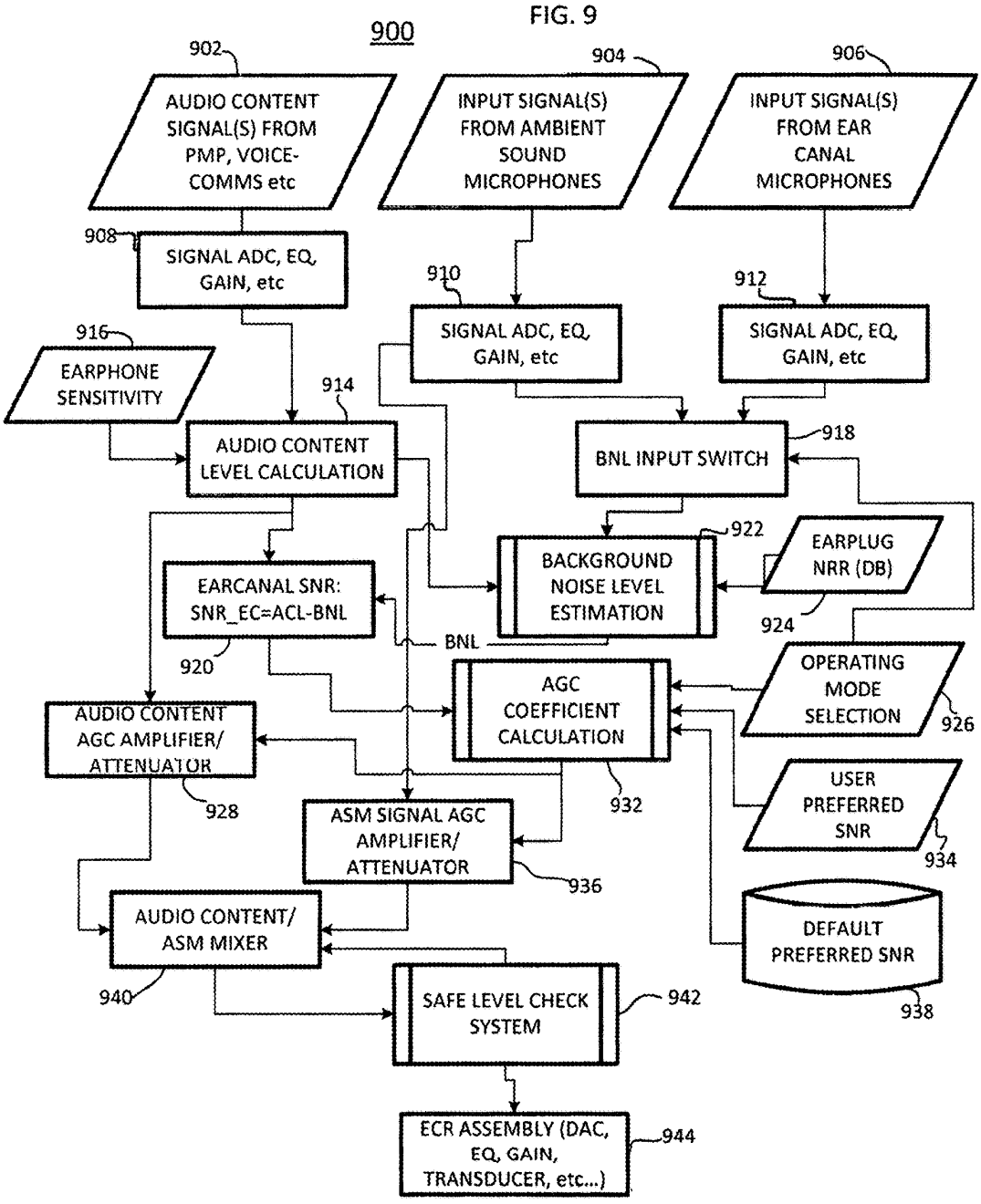
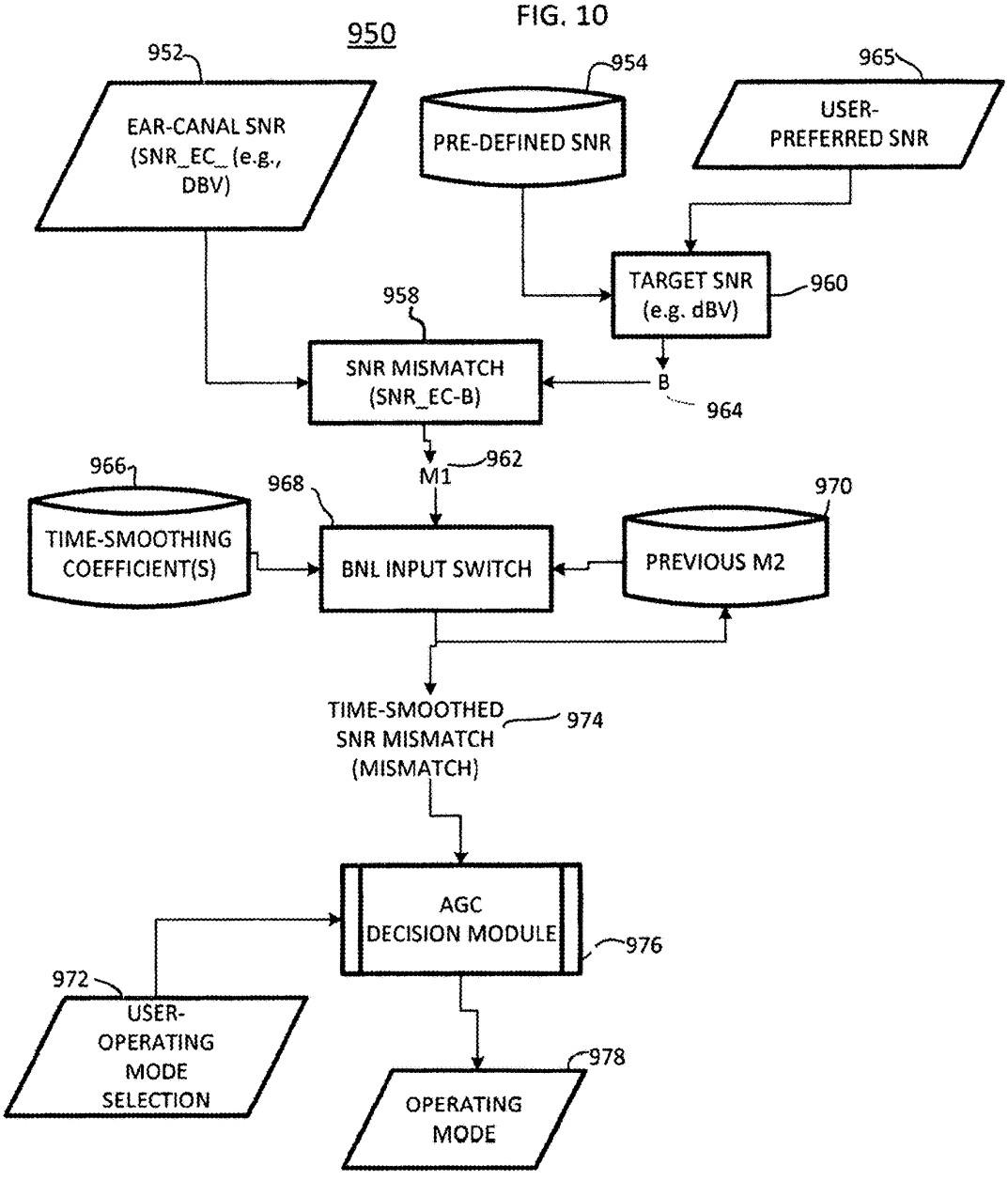

900
902
AUDIO CONTENT SIGNAL(S) FROM PMP, VOICE-COMMS etc
904
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES
906
INPUT SIGNAL(S) FROM EAR CANAL MICROPHONES
908
SIGNAL ADC, EQ, GAIN, etc
910
SIGNAL ADC, EQ, GAIN, etc
912
SIGNAL ADC, EQ, GAIN, etc
916
EARPHONE SENSITIVITY
914
AUDIO CONTENT LEVEL CALCULATION
BNL INPUT SWITCH
918
922
EARPLUG NRR (DB)
924
BACKGROUND NOISE LEVEL ESTIMATION
EARCANAL SNR: SNR_EC=ACL-BNL
920
BNL
OPERATING MODE SELECTION
926
AGC COEFFICIENT CALCULATION
AUDIO CONTENT AGC AMPLIFIER/ ATTENUATOR
928
932
USER PREFERRED SNR
934
ASM SIGNAL AGC AMPLIFIER/ ATTENUATOR
936
AUDIO CONTENT/ ASM MIXER
DEFAULT PREFERRED SNR
938
940
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 is a continuation of and claims priority to U.S. patent application Ser. No. 16/669,490, filed 30 Oct. 2019, which is a continuation of and claims priority to U.S. patent application Ser. No. 16/193,568, filed 16 Nov. 2018, now U.S. Pat. No. 10,535,334, which is a continuation of and claims priority to U.S. patent application Ser. No. 14/574,589, filed on Dec. 18, 2014, now U.S. Pat. No. 10,134,377, 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

Exhibit PAT-A

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 m accordance with an exemplary embodiment;

FIG. **4** is a more detailed approach to the method of FIG. **3** m 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) Ill 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** can 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:

$$i\text{ASL}=x\text{ASL}-\text{NRR} \quad \text{(EQ 1)}$$

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

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 hysteresis 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-dependent 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-dependent 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 dependent 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.

What is claimed is:

1. An earphone comprising:

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;

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;

a speaker configured to play an audio 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 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.

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

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.

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.

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.

5. The earphone according to claim **4**, where the operations further comprise:

sending a notification signal to the speaker.

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.

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.

8. The earphone according to claim **2**, where the operations further comprise:

adjusting the mixed audio signal when the user's voice is detected.

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.

10. The earphone according to claim **9**, where the mixed audio signal includes a noise reduction signal derived from the second microphone signal.

11. The earphone according to claim **1**, wherein the earphone includes a sealing section.

* * * * *

Exhibit PAT-H



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

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(45) **Date of Patent: Nov. 20, 2018**

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

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(22) Filed: **Dec. 18, 2014**

(65) **Prior Publication Data**

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Related U.S. Application Data

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(Continued)

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G10K 11/00 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***G10K 11/002*** (2013.01); ***H04R 1/1016*** (2013.01); ***H04R 1/1083*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC H04R 1/1083; H04R 2225/55; H04R 25/407; H04R 25/554; H04R 25/43; H04R 25/55; H04R 25/02; H04R 25/70; H04R 25/405; H04R 25/606; H04R 2225/39; H04R 2225/41; H04R 2225/51; H04R 2225/61; H04R 2225/67; H04R 2225/021; H04R 2225/023; H04R 2225/025; H04R 2225/59; H04R 1/1016; H04R 3/005; H04R 2410/03; H04R 2460/05; H04R 2460/15; H04R 2499/11; G10L 2015/223; G10L 2015/225; G10L 2015/226; G10L 2015/227; G10L 2015/228; G10L 25/78; G10L 25/783; G10L 25/81; G10L 25/84; G10L 25/87; G10L 25/786; G10L 25/93; G10L 15/265; H04M 1/72591;
(Continued)

(56) **References Cited**

U.S. PATENT DOCUMENTS

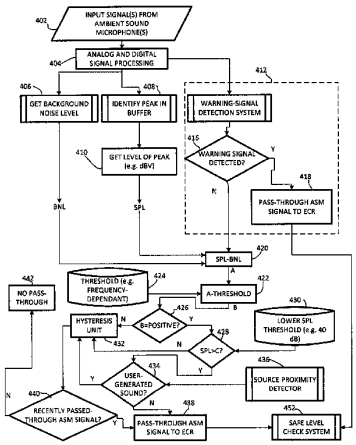
5,721,783 A 2/1998 Anderson
5,867,581 A * 2/1999 Obara H04R 25/505 381/312
(Continued)

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(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 it's 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



Related U.S. Application Data

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H04R 3/00 (2006.01)
H04R 1/10 (2006.01)

(52) **U.S. Cl.**
CPC ***H04R 3/002*** (2013.01); ***H04R 3/005*** (2013.01); *H04R 2225/41* (2013.01); *H04R 2460/05* (2013.01); *H04R 2499/11* (2013.01)

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CPC H04M 1/72597; H04M 1/7253; H04M 1/72533; H04M 1/72547; H04M 1/72569; H04M 1/05
USPC 381/56, 57, 58, 312, 317, 318, 328, 380, 381/71.6, 72, 74, 94.1
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

6,415,034	B1 *	7/2002	Hietanen	G10K 11/1788 381/151
6,754,359	B1 *	6/2004	Svean	H04R 1/1083 381/313
7,006,650	B1 *	2/2006	Wild	H04R 1/1016 128/848
7,929,713	B2 *	4/2011	Victorian	H04R 25/554 381/110
8,077,892	B2 *	12/2011	Dijkstra	H04R 25/30 381/312
8,098,832	B2 *	1/2012	Lin	G01S 3/8006 381/56
8,213,629	B2 *	7/2012	Goldstein	H04R 1/1041 381/107
8,229,127	B2 *	7/2012	Jorgensen	G10K 11/1788 381/318
8,774,433	B2 *	7/2014	Goldstein	H04R 1/1016 381/318
9,055,377	B2 *	6/2015	Kinsbergen	H04M 1/2475
9,137,611	B2 *	9/2015	Unno	H04R 3/005
9,232,322	B2 *	1/2016	Fang	H04R 25/50
9,357,314	B2 *	5/2016	Goorevich	H04R 25/554
9,613,622	B1 *	4/2017	Melanson	G10L 15/20
2001/0046304	A1 *	11/2001	Rast	H04R 1/1041 381/74
2005/0058313	A1 *	3/2005	Victorian	H04R 25/554 381/315
2006/0002574	A1 *	1/2006	Shennib	H04R 25/505 381/321
2006/0045282	A1	3/2006	Reber	
2006/0083388	A1	4/2006	Rothschild	
2006/0222194	A1 *	10/2006	Bramslow	H04R 25/305 381/314
2006/0262938	A1 *	11/2006	Gauger, Jr.	G10L 21/02 381/56
2007/0116310	A1 *	5/2007	Ludvigsen	H04R 25/356 381/316
2007/0189544	A1 *	8/2007	Rosenberg	G11B 27/105 381/57
2008/0159547	A1 *	7/2008	Schuler	G01H 3/14 381/56
2008/0181442	A1 *	7/2008	Goldstein	G01H 3/14 381/354
2008/0240458	A1	10/2008	Goldstein et al.	
2008/0260191	A1 *	10/2008	Victorian	H04R 25/554 381/315
2011/0195676	A1 *	8/2011	Victorian	H04R 25/554 455/90.3
2012/0020485	A1 *	1/2012	Visser	H04R 3/005 381/57
2016/0021469	A1 *	1/2016	Victorian	H04R 25/554 381/315

* cited by examiner

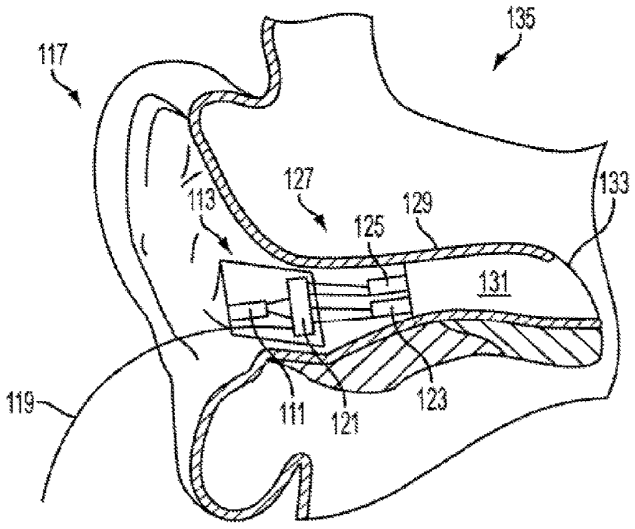


FIG. 1 100

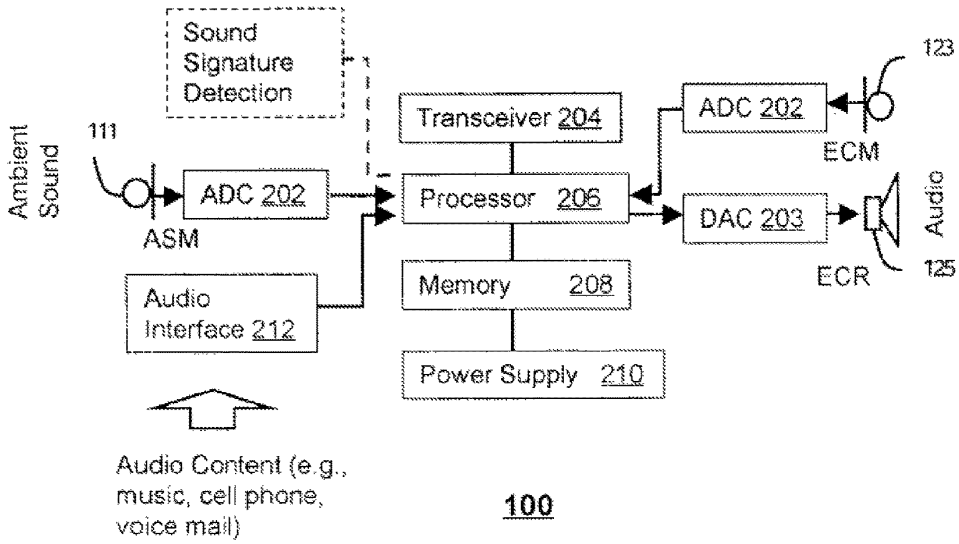
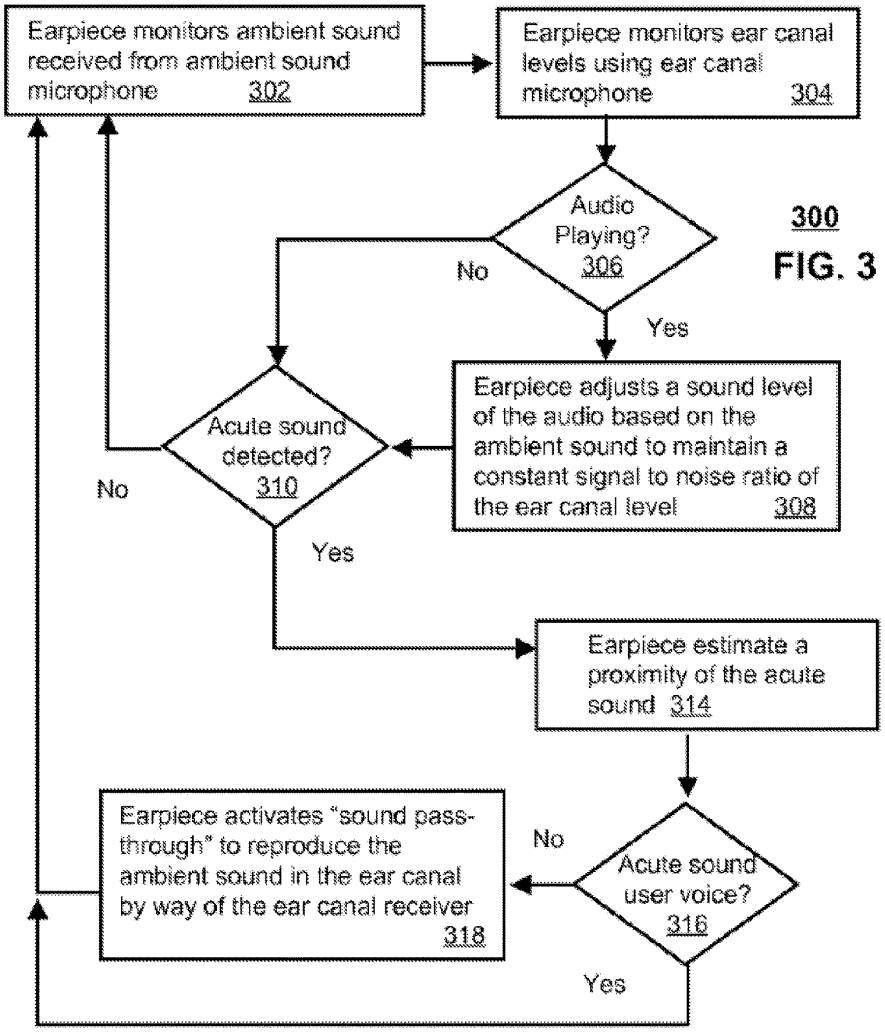


FIG. 2


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

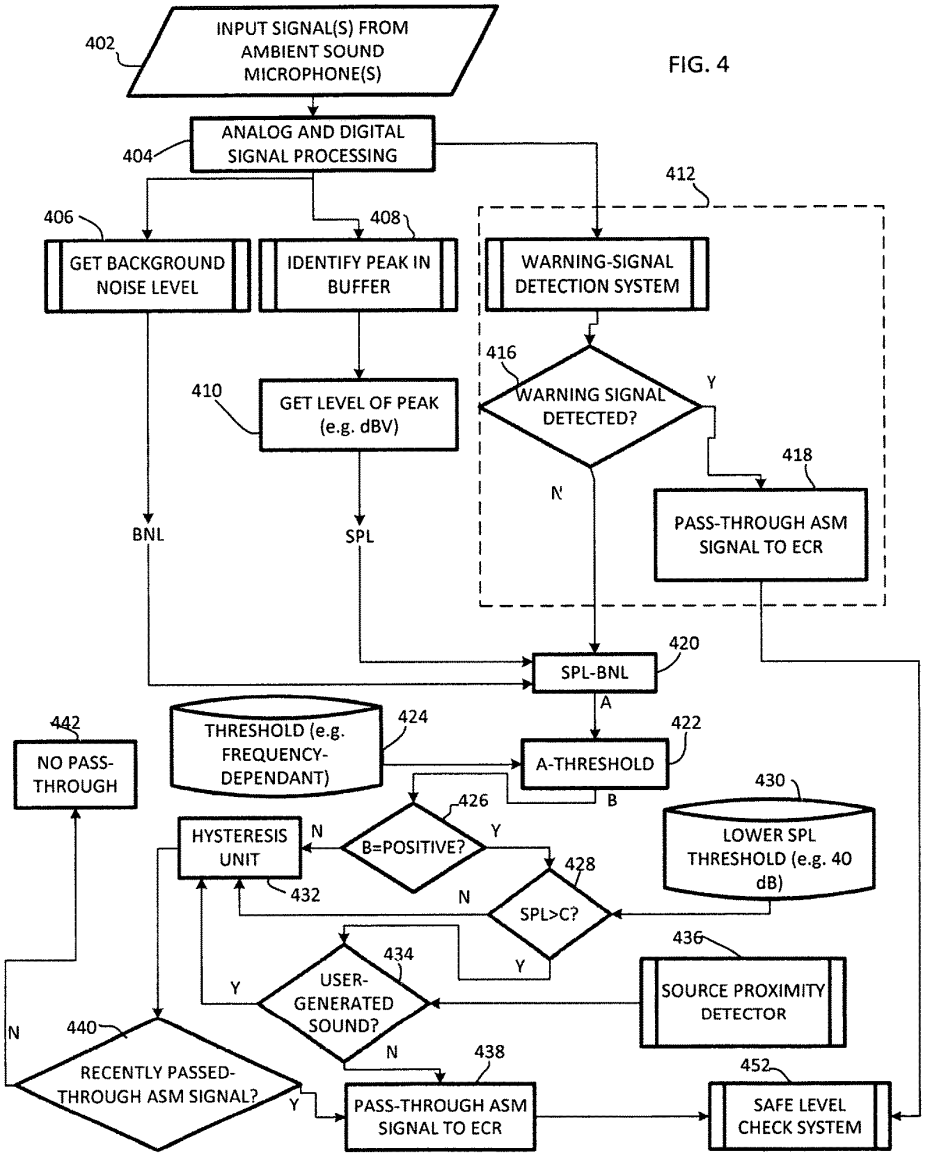
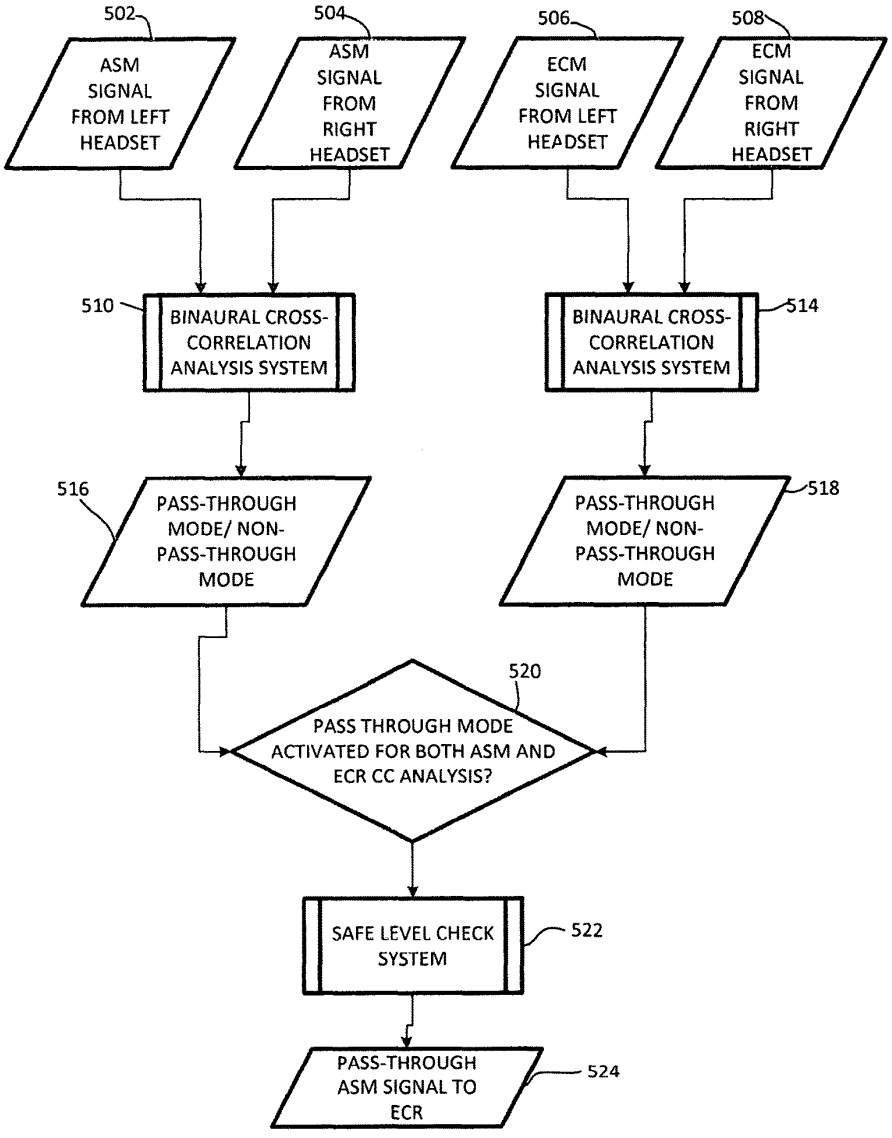

402
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONE(S)
404
ANALOG AND DIGITAL SIGNAL PROCESSING
406
GET BACKGROUND NOISE LEVEL
408
IDENTIFY PEAK IN BUFFER
410
GET LEVEL OF PEAK (e.g. dBV)
412
WARNING-SIGNAL DETECTION SYSTEM
416
WARNING SIGNAL DETECTED?
Y
N
418
PASS-THROUGH ASM SIGNAL TO ECR
BNL
SPL
420
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

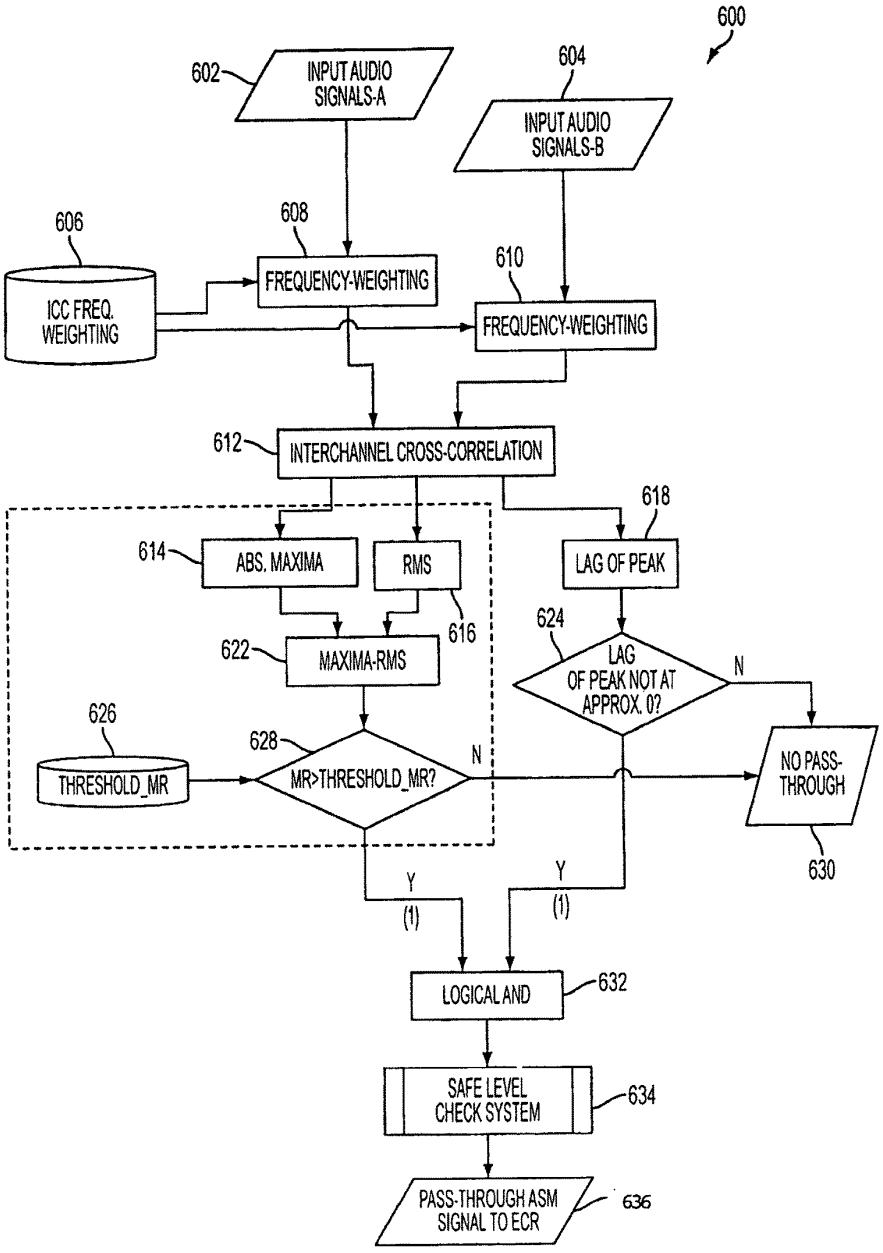
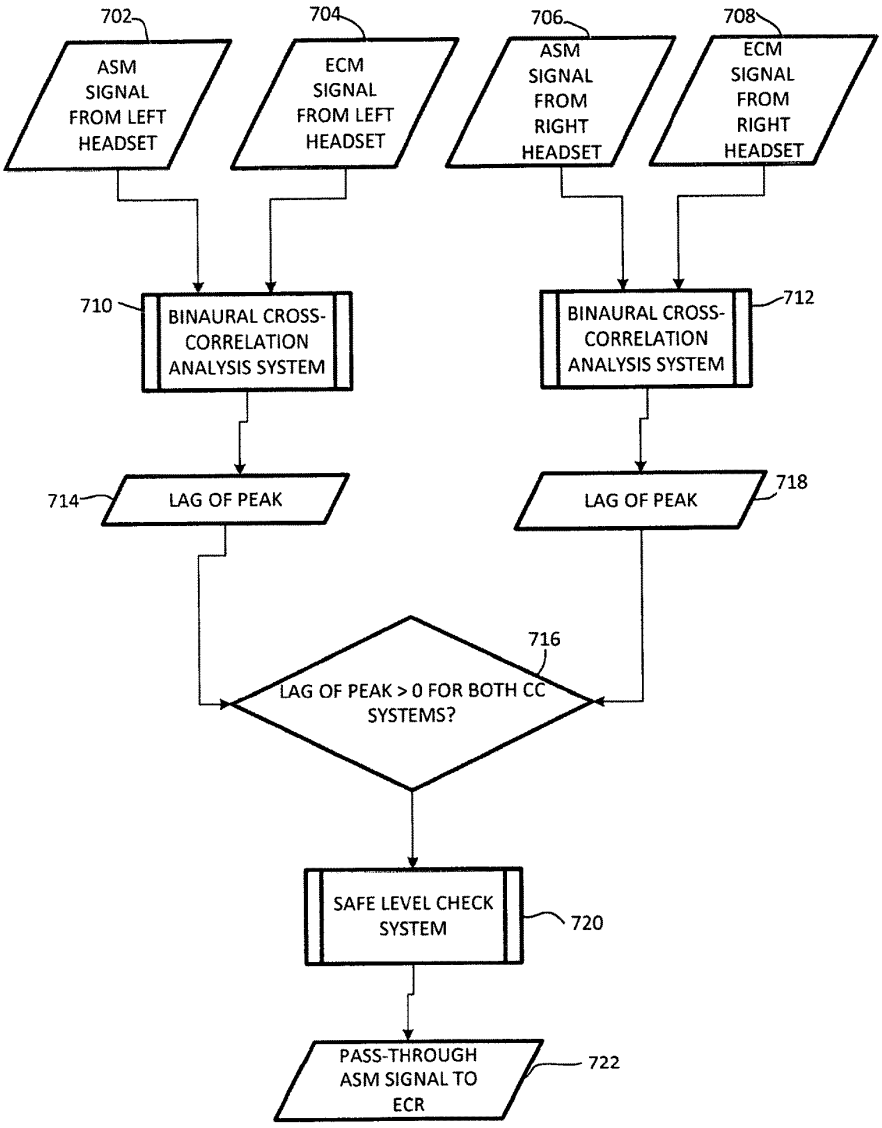

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH
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

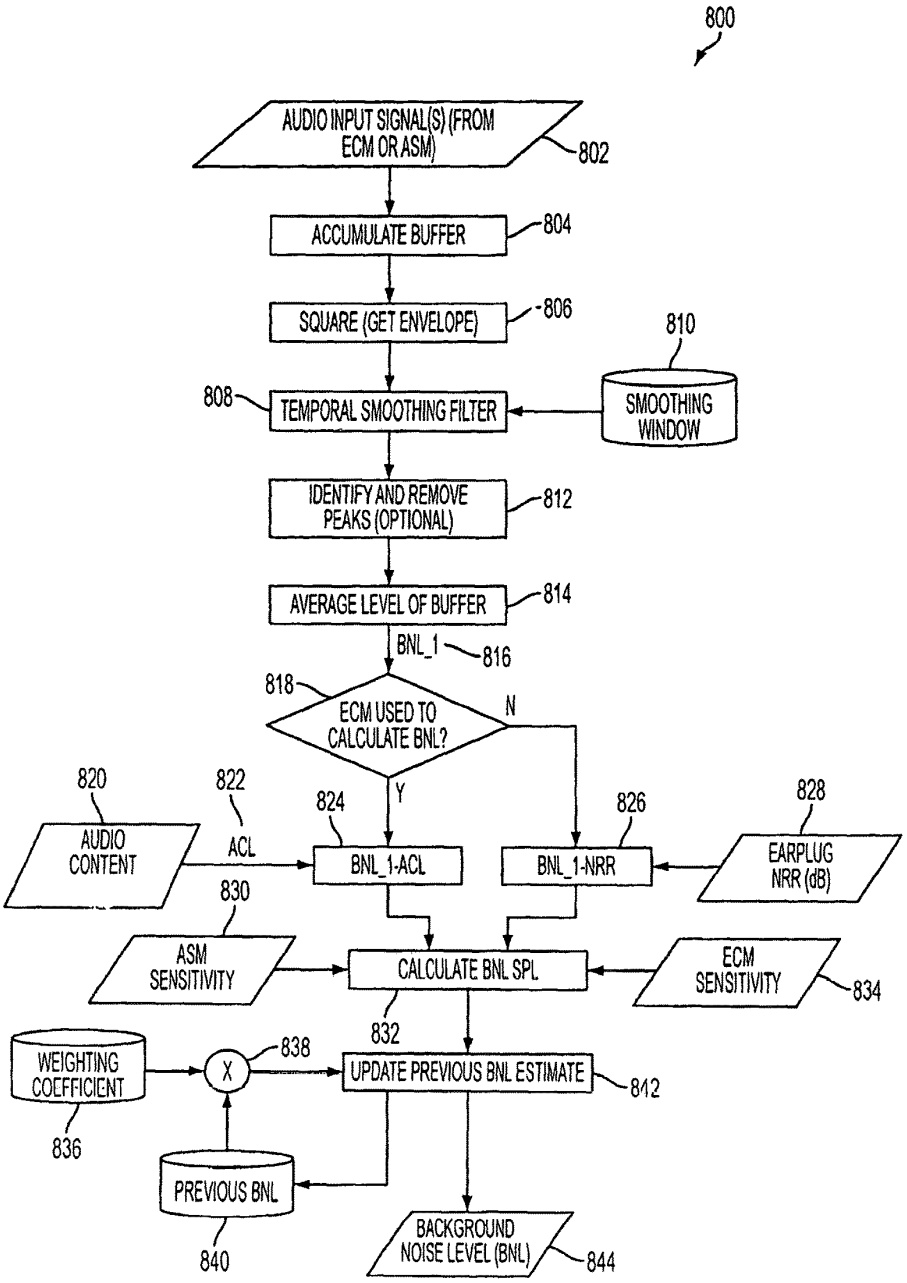

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT
822
ACL
824
BNL_1-ACL
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
812
836
PREVIOUS BNL
840
BACKGROUND NOISE LEVEL (BNL)
844

FIG. 8

FIG. 9

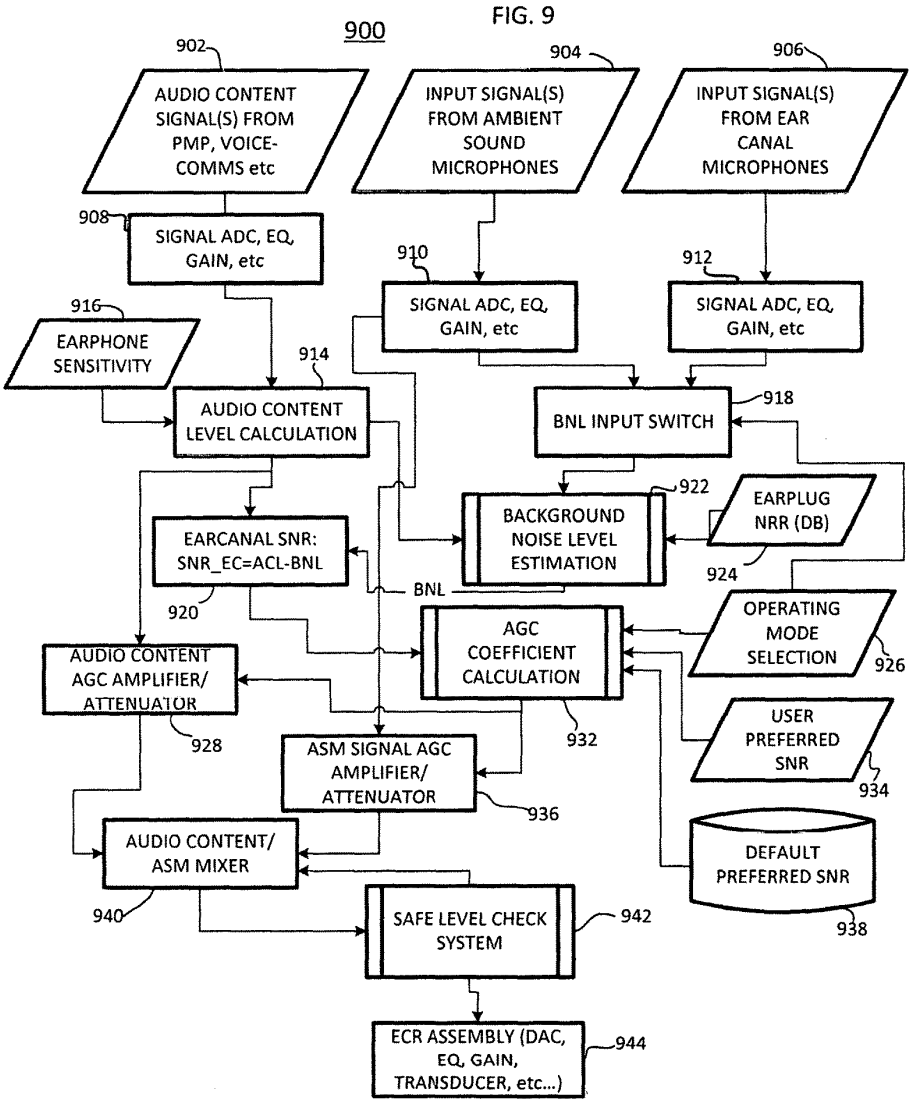
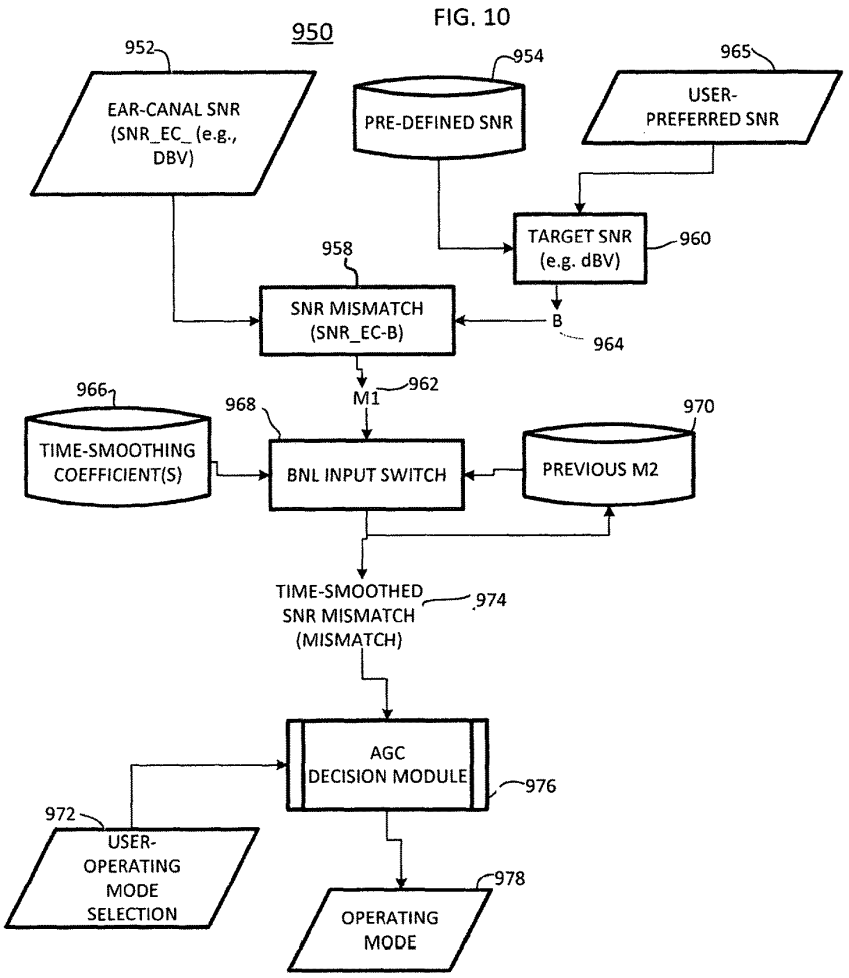

900
902
AUDIO CONTENT SIGNAL(S) FROM PMP, VOICE-COMMS etc
904
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES
906
INPUT SIGNAL(S) FROM EAR CANAL MICROPHONES
908
SIGNAL ADC, EQ, GAIN, etc
910
SIGNAL ADC, EQ, GAIN, etc
912
SIGNAL ADC, EQ, GAIN, etc
916
EARPHONE SENSITIVITY
914
AUDIO CONTENT LEVEL CALCULATION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

This application is Continuation Application of U.S. application Ser. No. 12/017,878 filed on Jan. 22, 2008 and which claims the priority benefit of Provisional Application No. 60/885,917 filed on Jan. 22, 2007, the entire disclosures of which are both incorporated herein by reference.

FIELD

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.

What is claimed is:

1. 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;

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 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.

2. The method of claim **1**, wherein the step of estimating a proximity further comprises determining a direction from the time lag.

3. The method of claim **1**, wherein the SPL-BNL difference is negative and wherein a hysteresis unit is configured to determine if the acute sound was previously detected.

4. The method of claim **3** wherein the acute sound is coupled to an ear canal receiver when the hysteresis unit determines if the processor had activated a pass-through mode.

5. The method of claim **3** wherein the acute sound is not coupled to the ear canal receiver when the hysteresis unit determines if the processor had not activated a pass-through mode.

6. The method of claim **3** further including a step of comparing the at least one peak level (SPL) to a second predetermined threshold when the SPL-BNL difference is greater than the first predetermined threshold wherein the acute sound is indicated when the SPL is greater than the second predetermined threshold.

7. The method of claim **6**, wherein the hysteresis unit is configured to determine if the acute sound was previously detected when the SPL is less than the second predetermined threshold.

8. A system configured for acute sound detection and reproduction, comprising:

a microphone used for measuring an external ambient sound level (xASL);

a memory containing computer instructions;

one or more processors coupled to the memory, the one or more processors configured when executing the computer instructions to perform operations comprising:

estimating a background noise level (BNL);

storing an external ambient sound in an audio buffer wherein the audio buffer accumulates the external ambient sound over a predetermined time period;

identifying and measuring at least one peak level (SPL) stored in the audio buffer:

subtracting the estimated background noise level (BNL) from the measured at least one peak level (SPL) identified in the audio buffer 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;

detecting a proximity of the acute sound to determine if the acute sound was user generated; and

passing the acute sound to an ear canal receiver if the acute sound was not user generated.

9. The system of claim **8**, further including the steps of:

estimating a proximity of the acute sound by performing a cross correlation analysis by at least two microphones;

identifying a peak in a cross correlation during the cross correlation analysis;

identifying a time lag in the acute sound between the at least two microphones; and

determining if the acute sound was user generated.

10. The system of claim 9 wherein the step of estimating the proximity further comprises determining a direction from the associated time lag.

11. The system of claim 8 wherein the SPL-BNL difference is negative and wherein a hysteresis unit is configured to determine if the acute sound was previously detected.

12. The system of claim 8, further comprising computer instructions which when executed by the processor performs the operation of measuring an internal ambient sound level (iASL) due to residual ambient sound within the ear canal at least partially occluded by the earpiece.

13. The system of claim 12 wherein monitoring the change in the xASL comprises monitoring a high frequency change between the xASL and the iASL.

14. The system of claim 8 further including a step of comparing the at least one peak level (SPL) to a second predetermined threshold when the SPL-BNL difference is greater than the first predetermined threshold wherein the acute sound is indicated when the SPL is greater than the second predetermined threshold.

15. A system configured for acute sound detection and reproduction, comprising:

at least two microphones including a microphone used for measuring an external ambient sound level (xASL);

a memory containing computer instructions;

one or more processors coupled to the memory, the one or more processors configured when executing the computer instructions to perform operations comprising:

estimating a background noise level (BNL) from the external ambient sound level (xASL);

storing an external ambient sound in an audio buffer wherein the audio buffer accumulates the the external ambient sound over a predetermined time period;

identifying and measuring at least one peak level (SPL) in the processed signal stored in the audio buffer;

comparing a SPL-BNL difference to a first predetermined threshold wherein an acute sound is indicated when the SPL-BNL difference is greater than the first predetermined threshold;

comparing the at least one peak level (SPL) to a second predetermined threshold wherein the acute sound is indicated when the SPL is greater than the second predetermined threshold;

estimating a proximity of the acute sound by performing a cross correlation analysis of signals captured by the at least two microphones;

identifying a peak in a cross correlation during the cross correlation analysis:

identifying a time lag in the acute sound between the at least two microphones to determine if the acute sound is user generated; and

passing the acute sound to an ear canal receiver if the acute sound was not user generated.

16. The system of claim 15, wherein the one or more processors are further configured to prevent the reproduction of the acute sound via the ear canal receiver when no time lag is identified in the acute sound between the at least two microphones.

17. The system of claim 15, wherein the earpiece is configured to measure the external ambient sound level of ambient sound outside of an ear canal when the ear canal is at least partially occluded by the earpiece.

18. The system of claim 15, further comprising computer instructions which when executed by the one or more processors performs the operation of measuring an internal ambient sound level (iASL) due to residual ambient sound when the ear canal is at least partially occluded by the earpiece.

19. The system of claim 18, wherein monitoring the change in the external ambient sound level (xASL) comprises monitoring frequency changes between the xASL and an internal ambient sound level (iASL).

20. The system of claim 19, wherein the at least one processor is configured to produce sound levels within the ear canal at a same sound pressure level as the acute sound measured at an entrance of the ear canal.

* * * * *

Exhibit OTH-E

Trials@uspto.gov
571-272-7822

Paper 10
Date: January 3, 2023

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SAMSUNG ELECTRONICS CO., LTD. and
SAMSUNG ELECTRONICS AMERICA, INC.,
Petitioner,

v.

STATON TECHIYA, LLC,
Patent Owner.

IPR2022-01099
Patent 11,244,666 B2

Before NATHAN A. ENGELS, MONICA S. ULLAGADDI, and SCOTT B. HOWARD, *Administrative Patent Judges.*

ULLAGADDI, *Administrative Patent Judge.*

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

I. INTRODUCTION

Samsung Electronics Co., Ltd. and Samsung Electronics America, Inc. ("Petitioner") filed a Petition (Paper 3, "Pet.") requesting *inter partes* review of claims 1, 2, and 8–11 ("the challenged claims") of U.S. Patent No. 11,244,666 B2 (Ex. 1001, "the '666 patent"), accompanied by the supporting Declaration of Dr. Nathaniel Polish (Ex. 1002). Staton Techiya, LLC ("Patent Owner") filed a Preliminary Response (Paper 7, "Prelim. Resp."), accompanied by the supporting Declaration of Dr. David Kleinschmidt (Ex. 2001). With our authorization, Petitioner filed a Reply (Paper 8, "Preliminary Reply" or "Prelim. Reply") and Patent Owner filed a Sur-reply (Paper 9, "Preliminary Sur-reply" or "Prelim. Sur-reply") addressing discretion under 35 U.S.C. § 314(a) and whether Platz qualifies as prior art.

The standard for institution is set forth in 35 U.S.C. § 314, which provides that an *inter partes* review may not be instituted unless the information presented in the Petition and the Preliminary Response shows that "there is a reasonable likelihood that the petitioner would prevail with respect to at least 1 of the claims challenged in the petition." 35 U.S.C. § 314 (2018); *see also* 37 C.F.R § 42.4(a) (2021) ("The Board institutes the trial on behalf of the Director.").

For the reasons discussed below, we do not institute an *inter partes* review of any of the challenged claims of the '666 patent.

II. BACKGROUND

A. Real-Parties-in-Interest

Petitioner identifies itself, Samsung Electronics Co., Ltd. and Samsung Electronics America, Inc., as the real parties-in-interest. Pet. 71.

Patent Owner identifies Staton Techiya, LLC and Synergy IP Corporation as the real parties-in-interest. Paper 5, 1.

B. Related Matters

The parties indicate that the '666 patent is involved in *Staton Techiya, LLC v. Samsung Electronics Co., Ltd.*, Case No. 2:22-cv-00053 (E.D. Tex.). Pet. 71; Paper 5, 1.

Petitioner identifies patents and patent applications that are in the same family as the '666 patent. Pet. 71; Ex. 1015. Pending patent applications that are continuations of the '666 patent include:

U.S. Appln. No. 17/321,892, filed on May 17, 2021 and published on September 2, 2021; and

U.S. Appln. No. 17/592,143, filed on February 3, 2022.

Petitioner further identifies the following proceedings before the Board in which Petitioner previously challenged other patents of Patent Owner:

IPR2022-00234 (challenging U.S. Patent No. 9,124,982, instituted);

IPR2022-00242 (challenging U.S. Patent No. 8,111,839, instituted);

IPR2022-00243 (challenging U.S. Patent No. 8,111,839, instituted);

IPR2022-00253 (challenging U.S. Patent No. 9,491,542, instituted);

IPR2022-00281 (challenging U.S. Patent No. 9,270,244, instituted);

IPR2022-00282 (challenging U.S. Patent No. 8,315,400, instituted);

IPR2022-00302 (challenging U.S. Patent No. 9,609,424, instituted);

IPR2022-00324 (challenging U.S. Patent No. 8,254,591, instituted);

IPR2022-00369 (challenging U.S. Patent No. 10,405,082, instituted);

IPR2022-00388 (challenging U.S. Patent No. 10,966,015, instituted); and

IPR2022-00410 (challenging U.S. Patent No. 10,979,836, institution denied).

Pet. 71–72.

Petitioner further identifies the following proceedings before the Board in which Petitioner concurrently challenges other patents of Patent Owner:

IPR2022-01078 (challenging U.S. Patent No. 11,057,701, instituted);

IPR2022-01098 (challenging U.S. Patent No. 11,217,237, institution denied); and

IPR2022-01106 (challenging U.S. Patent No. 11,039,259, pending institution decision).

Id. at 72.

C. The '666 Patent

The '666 patent is titled "Method and Device for Acute Sound Detection and Reproduction." Ex. 1001, code (54). The '666 patent concerns monitoring sound directed to an occluded ear and more particularly, concerns an earpiece that detects acute sounds and allows the acute sounds to be reproduced in an ear canal of the occluded ear. *Id.* at 1:23–28. Figure 1 of the '666 patent, reproduced below, shows an exemplary embodiment of an earpiece.

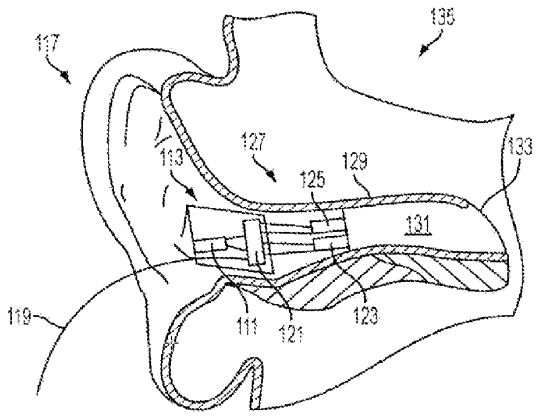


FIG. 1

Figure 1 shows an earpiece including electro-acoustical assembly 113 placed in ear canal 131 of user 135. *Id.* at 4:57–60. The earpiece includes ambient sound microphone (ASM) 111 for capturing ambient sound, ear canal receiver (ECR) (i.e., speaker) 125 for delivering audio to ear canal 131, and ear canal microphone (ECM) 123 for assessing a sound exposure level within ear canal 131. *Id.* at 4:66–5:3, 5:14. The earpiece further includes a processor operatively coupled to ASM 111, ECR 125, and ECM 123. *Id.* at 5:36–40. The earpiece can also include an audio interface operatively coupled to the processor to receive audio content, for example from a media player or cell phone, and direct sound to ear canal 131 via ECR 125. *Id.* at 5:56–59, 6:60–7:4. The processor monitors ambient sound captured by ASM 111 for acute sounds in the environment, such as an abrupt high energy sound corresponding to the onset 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.). *Id.* at 5:40–47. Using signals from ECM 123, the processor may determine whether a detected acute sound was generated by the earpiece user's voice or an external sound. *Id.* at 8:41–48; 10:8–25. The

processor can adjust the audio content and pass the acute sounds to ear canal 131 via ECR 125 responsive to detecting acute sounds. *Id.* at 7:25–44.

D. Illustrative Claim

Claim 1, reproduced below, is the sole independent claim.

[1pre] An earphone comprising:

[1a] 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;

[1b] 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;

[1c] a speaker configured to play an audio signal;

[1d] a memory that stores instructions; and

[1e] 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:

[1f] receiving a first microphone signal from the first microphone;

[1g] receiving a second microphone signal from the second microphone;

[1h] 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;

> [1i] applying an ambient sound gain to the ambient sound signal to generate a modified ambient sound signal;
>
> [1j] mixing the modified ambient sound signal with an audio content signal to generate a mixed audio signal; and
>
> [1k] sending the mixed audio signal to the speaker.

Ex. 1001, 13:44–14:17 (Petitioner's bracketed references added).

E. Asserted Grounds

Petitioner presents the following challenges as summarized in the chart below. Pet. 2. Petitioner supports its challenges with the Declaration of Dr. Nathaniel Polish. Ex. 1002.

Ground	Claim(s) Challenged	35 U.S.C. §	Reference(s)/Basis
1A	1, 11	103(a)[1]	Platz[2]
1B	2	103(a)	Platz, Kvaløy[3]
1C	8–10	103(a)	Platz, Kvaløy, Inanaga[4]

[1] The Leahy-Smith America Invents Act, Pub. L. No. 112–29, 125 Stat. 284 (2011) ("AIA"), included revisions to 35 U.S.C. § 103 that became effective after the effective filing date of the challenged claims. The earliest possible filing date is January 22, 2007. Ex. 1001, code (60). Therefore, we apply the pre-AIA version of 35 U.S.C. § 103(a).

[2] U.S. Patent Pub. No. 2010/0119077 A1, published May 13, 2010, to Platz et al. (Ex. 1005, "Platz").

[3] U.S. Patent No. 6,728,385 B2, issued April 27, 2004, to Kvaløy et al. (Ex. 1008, "Kvaløy").

[4] U.S. Patent No. 5,276,740, issued January 4, 1994, to Inanaga et al. (Ex. 1007, "Inanaga").

Ground	Claim(s) Challenged	35 U.S.C. §	Reference(s)/Basis
2A	1, 11	103(a)	Victorian[5], Inanaga
2B	2, 8–10	103(a)	Victorian, Inanaga, Kvaløy
3A	1, 11	103(a)	Inanaga
3B	2, 8–10	103(a)	Inanaga, Engle[6]

III. ANALYSIS

A. Legal Standards

A patent claim is unpatentable under 35 U.S.C. § 103(a) if the differences between the claimed subject matter 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. *KSR Int'l Co. v. Teleflex Inc.*, 550 U.S. 398, 406 (2007). The question of obviousness is resolved on the basis of underlying factual determinations including: (1) the scope and content of the prior art; (2) any differences between the claimed subject matter and the prior art; (3) the level of ordinary skill in the art; and (4) when in evidence, objective evidence of nonobviousness.[7] *Graham v. John Deere Co.*, 383 U.S. 1, 17–18 (1966).

B. Level of Ordinary Skill in the Art

In determining the level of ordinary skill in the art, various factors may be considered, including the "type of problems encountered in the art; prior art solutions to those problems; rapidity with which innovations are

[5] European Patent Pub. No. EP 1519625 A2, published March 30, 2005, to Victorian et al. (Ex. 1006, "Victorian").
[6] U.S. Patent No. 8,477,955 B2, issued July 2, 2013, to Engle et al. (Ex. 1009, "Engle").
[7] Patent Owner does not present objective evidence of nonobviousness at this stage of the proceeding.

made; sophistication of the technology; and educational level of active workers in the field." *In re GPAC Inc.*, 57 F.3d 1573, 1579 (Fed. Cir. 1995). According to Petitioner,

> [a] person of ordinary skill in the art at the time of the alleged invention of the '666 patent ("POSITA"), which for purposes of this proceeding is no earlier than January 22, 2007, would have had a bachelor's degree in electrical engineering, computer science, or a similar field and two years of experience in the design of digital audio systems and associated signal processing. The POSITA could have also obtained similar knowledge and experience through other means.

Pet. 11 (citing Ex. 1002 ¶ 20). Patent Owner "applies the level of ordinary skill in the art that is proposed in the Petition." Prelim. Resp. 11.

We adopt Petitioner's definition of the level of skill in the art, which is consistent with the disclosure of the '666 patent and asserted prior art of record.

C. Claim Construction

In *inter partes* review proceedings, we construe claims using the same claim construction standard that would be used in a civil action under 35 U.S.C. § 282(b), as articulated in *Phillips v. AWH Corp.*, 415 F.3d 1303 (Fed. Cir. 2005) (en banc), and its progeny. *See* 37 C.F.R. § 42.100(b).

Petitioner contends that "no express constructions of the claims are necessary to assess whether the prior art reads on the challenged claims" and that "the claims should be given their plain and ordinary meaning." Pet. 11. Patent Owner "proposes that all claim terms be afforded their plain and ordinary meaning as understood by a person of ordinary skill in the art at the time of the invention of the '666 patent." Prelim. Resp. 11.

We need not construe any claim term to resolve the institution question, and therefore, do not expressly define any claim term in our

Decision. *See Nidec Motor Corp. v. Zhongshan Broad Ocean Motor Co.*, 868 F.3d 1013, 1017 (Fed. Cir. 2017) (explaining that construction is needed only for terms that are in dispute, and only as necessary to resolve the controversy).

D. Discretion to Deny Under 35 U.S.C. § 314(a)

The parties present arguments regarding exercising our discretion to deny institution under 35 U.S.C. § 314(a). Pet. 66–69; Prelim. Resp. 19–27; Prelim. Reply 1–5; Prelim. Sur-reply 1–3.

For the reasons discussed in Sections III.E, III.F, and III.G, we deny institution on the merits of Petitioner's challenges and as such, do not address the parties' arguments regarding the issue of whether to exercise our discretion to deny institution under 35 U.S.C. § 314(a).

E. Challenges 1A, 1B, and 1C Based on Platz

Petitioner contends that claims 1 and 11 would have been obvious in view of Platz. Pet. 12–26. Petitioner contends that claim 2 would have been obvious over Platz in view of Kvaløy. *Id.* at 27–31. Petitioner further contends that claims 8–10 would have been obvious over the combination of Platz, Kvaløy, and Inanaga. *Id.* at 31–35. For the reasons that follow, we are not persuaded that Petitioner establishes a reasonable likelihood of prevailing with respect to any of its challenges applying Platz.

1. Overview of Platz

Platz is titled "Active Hearing Protection System." Ex. 1005, code (54). Platz relates to providing hearing protection to a user. *Id.* ¶ 12. Figure 1 of Platz, reproduced below, shows a hearing protection system to be worn by the user.

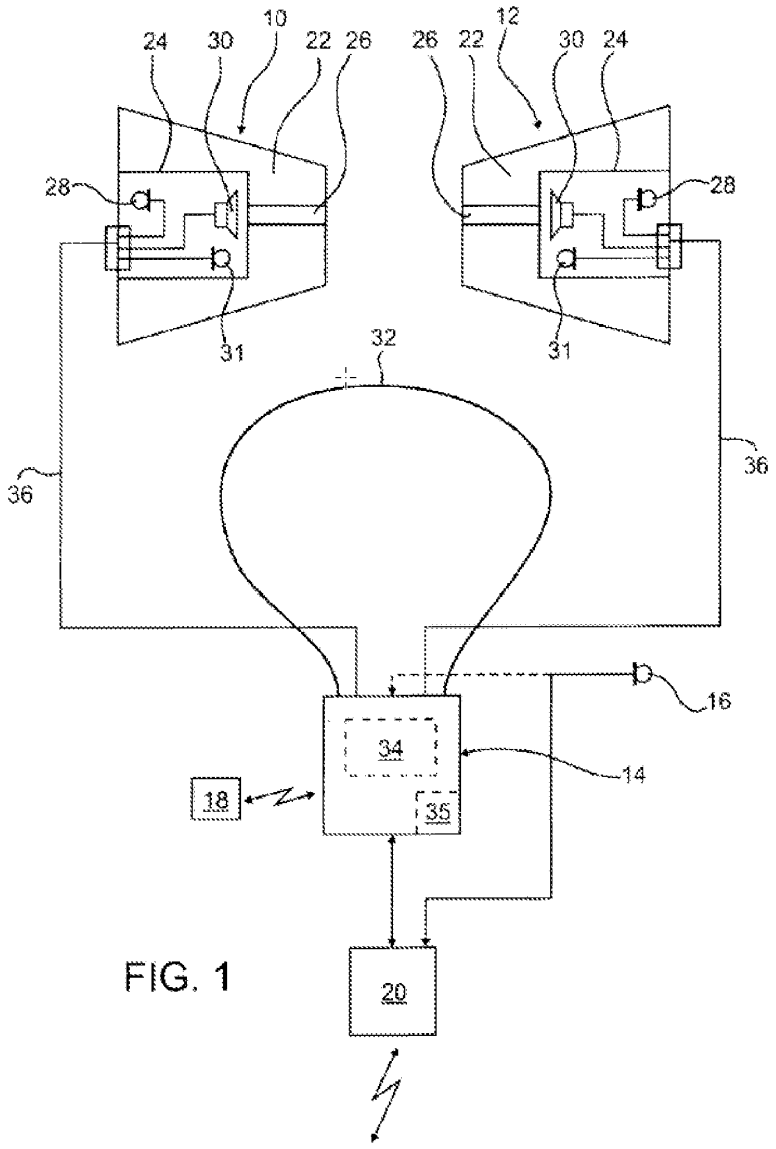


FIG. 1

As shown in Figure 1, the hearing protection system comprises left and right earplugs 10, 12, each of which provide for an acoustic attenuation. *Id.* ¶ 2. Active unit 24 comprises microphone 28 for capturing audio signals from ambient sound and loudspeaker 30 for providing audio signals into the user's ear canal via sound channel 26. *Id.* ¶ 22. The hearing protection system further comprises wearable central unit 14 that is worn on the user's body.

Id. ¶ 23. Central unit 14 comprises audio signal processing unit 34 for receiving and processing the audio signals captured by microphones 28 and for supplying loudspeakers 30 with audio signals to be reproduced in the user's ear. *Id.*

Central unit 14 is connected to communication device 20 such as a mobile phone or walkie-talkie. Ex. 1005 ¶¶ 20, 24. Once an audio signal from communication device 20 is detected, central unit 14 switches to a "communication mode" in which input from right ear microphone 28 is replaced by the audio signals from communication device 20. *Id.* ¶ 35. A mixture of audio signals representative of the ambient sound as captured by left ear microphone 28 and the audio signals provided by communication device 20 is then provided to both loudspeakers 30. *Id.* ¶¶ 35–36. For bidirectional communication, active unit 24 may include inner microphone 31 to capture primarily the user's voice, in which case central unit 14 performs blind source separation (BSS) on the audio signals provided by microphones 28, 31 in order to separate the user's voice from background noise. *Id.* ¶¶ 25, 42.

2. Independent Claim 1

[1pre] "An earphone comprising:"

Petitioner contends that, "[t]o the extent the preamble is limiting, *Platz* discloses or suggests this feature." Pet. 12 (citing Ex. 1002 ¶¶ 51–53). Specifically, Petitioner contends that "*Platz* discloses an active hearing protection system having a left earplug 10, a right earplug 12, and a central unit 14" and that "[a] POSITA would have understood each of *Platz*'s earplugs 10, 12 to comprise '[a]n earphone device' at least because they each include a speaker 30 configured to provide audio signals received from a communication device 20 to the user's ear." *Id.* (citing Ex. 1005 ¶¶ 20, 24,

25; Ex. 1002 ¶ 52). Petitioner also contends that "*Platz*'s earplugs 10, 12 are also of the same general construction as the 'earphone device' (or 'earpiece') described in the '666 patent." *Id.* (citing Ex. 1001, 4:52–65, 6:21–26).

> *[1a] "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;"*

> *[1b] "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;"*

Petitioner contends that "*Platz* discloses that active unit 24 of each earplug 10, 12 includes a microphone 28 configured to 'captur[e] audio signals from ambient sound.'" Pet. 12 (citing Ex. 1005 ¶¶ 22, 25) (alteration by Petitioner). Petitioner contends "*Platz* provides that 'microphone 28 is acoustically oriented towards ambience in order to capture primarily ambient sound.'" *Id.* (citing Ex. 1005 ¶ 42). Petitioner further contends that "*Platz* discloses that active unit 24 also includes a second, inner microphone 31 that 'is acoustically oriented towards the ear canal in order to capture primarily the user's voice.'" *Id.* Petitioner also contends that "*Platz*'s Figure 1 depicts inner microphone 31 as closer than microphone 28 to the outlet of sound channel 26, through which *Platz* indicates active unit 24 is acoustically connected to the ear canal." *Id.* (citing Ex. 1005 ¶¶ 22, 42, Fig. 1; Ex. 1002 ¶ 55).

With respect to the claimed "first microphone port that is configured to face away from a user when the earphone is inserted" and the claimed "second microphone port that is configured to face toward the user when the earphone is inserted," Petitioner contends

> *Platz* does not expressly state "the first microphone ***has a first microphone port***" or that "the second microphone ***has a second microphone port***." However, it would have been obvious to a POSITA to incorporate a port into the microphones 28 and 31 to receive sounds, because this was a common way of designing microphones incorporated into earpieces. Incorporating a port into microphones 28 and 31 would also have been obvious because the port would provide less acoustic impedance to sounds than if the microphone was completely covered by its packaging, which a POSITA would have recognized would have unnecessarily or inappropriately attenuated sounds before reaching the microphone. Thus, a POSITA would have been motivated to incorporate such a port into *Platz*'s microphones 28 and 31 to facilitate or improve their intended purpose of capturing/measuring ambient and ear canal sounds, respectively.
>
> It would also have been obvious to a POSITA to orient microphone 28's port to face away from the user and microphone 31's port to face towards the user, when the earphone is inserted. A POSITA would have recognized, as stated in *Platz,* that it was well known within the art to acoustically orient microphones towards desired sound sources. A POSITA would have been motivated to orient microphone 28's port away from the user to ensure that the microphone performed its intended purpose of capturing ambient sounds, and to orient microphone 31's port towards the user's ear canal to ensure that the microphone performed its intended purpose of capturing sound closer to the user's ear canal.

Id. at 15–16 (citing Ex. 1005 ¶ 42, Fig. 1; Ex. 1002 ¶¶ 59–64 (citing Ex. 1011, 5:40–45, Fig. 1) [8]).

Petitioner relies on Dr. Polish's testimony to support its contentions:

> the '666 patent does not recite a "port" in its specification, but the applicant argued during prosecution that this feature is supported by Figures 1 and 2. . . . As can readily be observed, neither figure depicts a microphone port — only Figure 1 notes

[8] U.S. Patent 6,754,359 B1, issued June 22, 2004, to Svean et al. (Ex. 1011, "Svean").

two microphones, an ASM 111 and an ECM 123. Applicant's argument supports the notion that microphone ports were so common and well understood that the mere schematic depiction of a microphone would have led a person of ordinary skill in the art to understand that a port was incorporated in said microphone.

Incorporating a port into microphones 28 and 31 would also have been obvious because the port would provide less acoustic impedance to sounds than if the microphone were completely covered by its packaging, which a person of ordinary skill in the art would have recognized would have unnecessarily or inappropriately attenuated sounds before reaching the microphones.

In my opinion, therefore, a person of ordinary skill in the art would have been motivated to incorporate such a port into *Platz*'s microphones 28 and 31 to ensure that the microphones served their intended purposes of capturing/measuring ambient and ear canal sounds, respectively.

It would also have been obvious to a person of ordinary skill in the art to orient the port of microphone 28 to face away from the user and to orient the port of microphone 31 to face towards the user, when the earphone is inserted. A person of ordinary skill in the art would have recognized, as stated in *Platz*, that it was well known within the art to acoustically orient microphones towards desired sources of sound:

> Preferably, each active unit 24 of the earplugs 10, 12 in this case may include **a second microphone 31 which is acoustically oriented towards the ear canal in order to capture primarily the user's voice, whereas the microphone 28 is acoustically oriented towards ambience in order to capture primarily ambient sound.**

Ex. 1002 ¶¶ 60–63 (quoting Ex. 1005 ¶ 42, Fig. 1) (citing Ex. 1004, 73–74).

Dr. Polish further testifies that

Svean shows microphones that face away from the user ("microphone M1) and toward the user ("microphone M2") to

detect, respectively, ambient external sounds and internal ear canal sounds:

> The **sound inlet of microphone M1 is connected to the outside of the earplug, picking up the external sounds**. The microphone M2 is connected to the inner portion of the meatus 3 by means of an acoustic transmission channel T1.

Id. ¶ 63 (quoting Ex. 1011, 5:40–45, Fig. 1).

Patent Owner's Argument

Patent Owner argues that

> Petition's Ground 1 theories fail because they all rely on Platz, which is not available as prior art. The Petition alleges that Platz is available as prior art under 35 U.S.C. § 102(e). For a patent application publication to be available as prior art under that section, the application must have been filed in the United States "before the invention by the applicant for patent." The inventions of the '666 patent claims were reduced to practice no later than November 18, 2006. This was shown by the applicant during prosecution. The PCT application that published as Platz was filed on December 18, 2006. Because Platz was filed after the inventions of the '666 patent, Platz is not available as prior art. Grounds 1A-1C.

Prelim. Resp. 12 (citing Pet. 3; 35 U.S.C. § 102(e); Prelim. Resp. § III; Ex. 1004, 44). We do not reach this argument because we resolve the institution question based on a failure to sufficiently show limitations 1[a] and 1[b] in Petitioner's challenges applying Platz, as discussed in detail below.

Analysis of Petitioner's Contentions for Challenges Applying Platz

As Petitioner acknowledges, Platz does not disclose either a "first microphone port that is configured to face away from a user when the earphone is inserted" or a "second microphone port that is configured to face toward the user when the earphone is inserted." The cited portion of Figure 1

of Svean, reproduced below, shows microphones M1 and M2, but Petitioner does not sufficiently show that it teaches a port, let alone the claimed "first microphone port" and "second microphone port" as recited in claim 1.

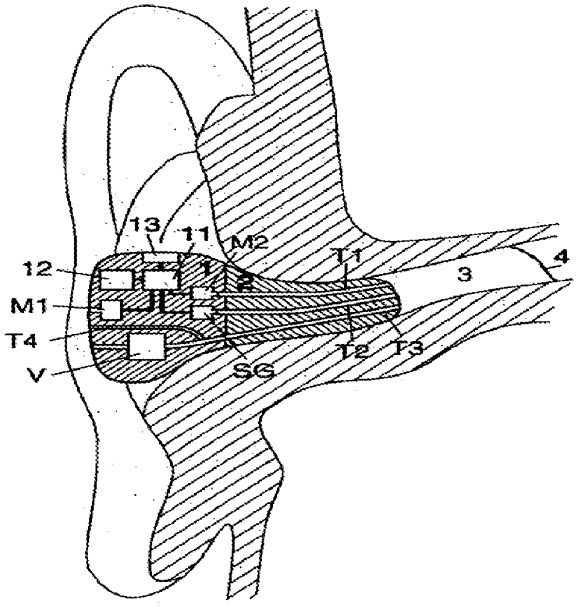


Fig. 1

Figure 1 of Svean illustrates microphones M1 and M2.

In grounds 1B and 1C, Petitioner does not cite Kvaløy or Inanaga to cure the deficiencies of Platz in ground 1A. Instead, Petitioner impermissibly relies on unsupported expert testimony to supply a limitation missing from Platz. Dr. Polish's testimony is not supported because none of the cited references or cited underlying evidence (i.e., Svean) teaches, suggests, or even mentions a port.

The Board's Trial Practice Guide expressly prohibits such an approach:

> Expert testimony, however, cannot take the place of a disclosure in a prior art reference when that disclosure is required as part of the unpatentability analysis . . . . [I]n an obviousness analysis,

> conclusory assertions from a third party about general knowledge in the art cannot, **without supporting evidence of record**, supply a limitation that is not evidently and indisputably within the common knowledge of those skilled in the art. *K/S Himpp v. Hear-Wear Techs., LLC*, 751 F.3d 1362, 1365 (Fed. Cir. 2014). Furthermore, because an inter partes review may only be requested "on the basis of prior art consisting of patents or printed publications," 35 U.S.C. § 311(b), expert testimony cannot take the place of disclosure from patents or printed publications. In other words, expert testimony may explain "patents and printed publications," **but is not a substitute for disclosure in a prior art reference itself**.

Consolidated Trial Practice Guide at 36 (November 2019) ("CTPG") (emphasis added).

In *K/S Himpp*, the Federal Circuit determined that a CRU[9] examiner did not err in deciding to reject a party's contentions that the structural features of certain claims were known prior art elements because the party "failed to point to any evidence on the record." *K/S Himpp*, 751 F.3d at 1366. The court also affirmed the Board's decision to refuse to adopt the party's obviousness contention because the party "failed to cite any evidence on the record to support its contention that" certain claims recited "known prior art elements." *Id.*

Petitioner does not provide a construction or otherwise explain what either the "first microphone port" or the "second microphone port" encompasses. *See* Pet. 11 (Petitioner arguing that "no express constructions of the claims are necessary to assess whether the prior art reads on the challenged claims" and that "the claims should be given their plain and ordinary meaning"). The record does not help us ascertain the plain and ordinary meaning of "microphone port." *See* Ex. 1002 ¶ 60 (contending "the

[9] Central Reexamination Unit

'666 patent does not recite a 'port' in the specification"). Notably, the limitations separately recite a first and second "microphone" and a first and second "microphone port." *See* Prelim. Resp. 3–4 (arguing the "microphone port" limitations were added during prosecution to overcome an examiner's rejection). Petitioner also did not point to a description, definition, figure, or mention in *any* reference, that explains what the claim terms mean, *for example*, in terms of their geometry and orientation.[10] As Petitioner does not explain and the record does not show what a "microphone port" is, Petitioner cannot sufficiently show that the prior art teaches or suggests a "first microphone port that is configured to face away from a user when the earphone is inserted" and a "second microphone port that is configured to face toward the user when the earphone is inserted" as recited in claim 1.

For the foregoing reasons, we determine that Petitioner has not met its burden and does not sufficiently show that Platz alone, or in combination with Kvaløy, or in combination with Kvaløy and Inanaga teaches or suggests limitations 1[a] and 1[b] of independent claim 1. As such, Petitioner fails to establish a reasonable likelihood of prevailing with respect to its obviousness challenge to independent claim 1 applying Platz. Petitioner also fails to establish a reasonable likelihood of prevailing with respect to its obviousness challenges to claims 2 and 8–11, which depend from claim 1.

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[10] This fact alone casts doubt on Dr. Polish's testimony that "microphone ports were so common and well understood that the mere schematic depiction of a microphone would have led a person of ordinary skill in the art to understand that a port was incorporated in said microphone." Ex. 1002 ¶ 60. Moreover, Dr. Polish's testimony regarding a port "in said microphone" only confuses the issue; while the claims separately recite a first and second "microphone" and a first and second "microphone port," it is unclear whether Petitioner a "microphone port" is part of the prior art microphones, part of the earphone housing, or something else.

Petitioner does not cite Kvaløy or Inanaga to cure the deficiencies of Platz, and these references do not cure the deficiencies of Platz with respect to independent claim 1.

F. Challenges 2A and 2B Based on Victorian

Petitioner contends that claims 1 and 11 would have been obvious in view of Victorian and Inanaga. Pet. 36–47. Petitioner contends that claims 2 and 8–10 would have been obvious over the combination of Victorian, Inanaga, and Kvaløy. *Id.* at 47–50. For the reasons that follow, we are not persuaded that Petitioner establishes a reasonable likelihood of prevailing with respect to any of its challenges applying Victorian.

1. Overview of Victorian

Victorian is titled "[e]xternal ear canal voice detection." Ex. 1006, code (54). Victorian discloses an ear-level, full-duplex audio communication system including one or two ear attachment devices that wirelessly communicate to a remote device (e.g., computer, personal digital assistant, cellular phone, walkie talkie, language translator). *Id.* at code (57). Figure 4A of Victorian, reproduced below, is a block diagram showing an embodiment of the circuit of an ear-level device worn by a user. *Id.* ¶ 13.

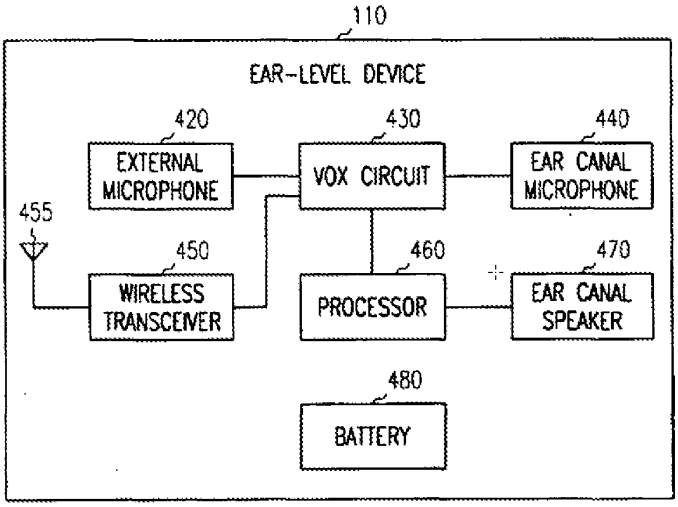


FIG. 4A

As shown in Figure 4A of Victorian, ear-level device 110 includes external microphone 420 which picks up ambient sound, ear canal microphone 440 which detects sound from the user's ear canal including the user's own speech, and ear canal speaker 470 which transmits sound received from a remote device and/or external microphone 420 to the user's ear canal. *Id.* ¶ 30. Ear-level device 110 further includes processor 460 which converts the sound picked up from the ear canal to an electrical signal to be transmitted to the remote device via wireless transceiver 450, and converts the signals received from the remote device via wireless transceiver 450 and/or from external microphone 420 into a sound audible to the user. *Id.* Ear-level device 110 also includes voice operated exchange (VOX) circuit 430 which 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. *Id.*

2. Independent Claim 1

[1pre] "An earphone comprising:"

Petitioner contends that "[t]o the extent the preamble is limiting, *Victorian* discloses or suggests this feature." Pet. 36 (citing Ex. 1002 ¶¶ 133–135). According to Petitioner, "*Victorian* discloses an ear-level communication system 100 comprising an ear-level device 110." *Id.* (citing Ex. 1006 ¶ 18). Petitioner further contends that "[e]ar-level device 110 is depicted in the user's ear in Figure 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.* (citing Ex. 1006 ¶¶ 24, 25). Petitioner also contends that "ear-level device 110 can be used as an ear piece." *Id.* (citing Ex. 1006 ¶ 8, Abstract).

[1a] "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;"

[1b] "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;"

Petitioner contends that "*Victorian* discloses ear-level device 110 including an '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.'" Pet. 36 (citing Ex. 1006 ¶ 30). According to Petitioner "it would have been obvious to incorporate a port into microphone 420, and to orient such a port away from the user to detect desired ambient sounds, for the same reasons discussed" with respect to the

challenges applying Platz. *Id.* (citing Ex. 1002 ¶ 137; Pet. § VIII.A.1.b); *see also id.* (citing Ex. 1002 ¶¶ 136–140).

Petitioner further contends that "*Victorian* discloses ear-level device 110 including an '[e]ar canal microphone 440 detects sound from the ear canal.'" Pet. 37 (citing Ex. 1006 ¶ 30) (alteration by Petitioner). According to Petitioner, "[t]he sound includes primarily speech of the person wearing ear-level device 110." *Id.* Petitioner also contends that "[i]t would have been obvious to incorporate a port into microphone 440, and to orient the port to face towards the user, for the same reasons discussed" with respect to the challenges applying Platz. *Id.* at 37–38 (citing Ex. 1002 ¶¶ 141, 142; Pet. § VIII.A.1.b).

Dr. Polish testifies that,

> [w]hile *Victorian* does not expressly state that microphone 420 has a port, it would have been obvious to incorporate a port into microphone 420, and to orient such a port away from the user to detect desired ambient sounds, for the same reasons . . . discussed in Section IX.B.1.b.
>
> . . . .
>
> *Victorian* does not expressly state that the ear canal microphone 440 has a port. However, it would have been obvious to incorporate a port into microphone 440, and to orient the port to face towards the user, for the same reasons . . . discussed in Section IX.B.1.b.

Ex. 1002 ¶¶ 137, 142.

Patent Owner's Argument

Patent Owner contends that "[t]he Petition's Ground 2 theories fail because it would not have been obvious to modify Victorian to be more complicated without any apparent benefit." Prelim. Resp. 12. According to Patent Owner, "[t]he Petition vaguely contends that it would have been

obvious to incorporate Inanaga's teachings into Victorian 'because doing so would have been consistent with and/or improved *Victorian*'s goal of allowing the wearer of the ear-level device to hear ambient sound in the wearer's vicinity while permitting the wearer to also listen to audio content from a remote device.'" *Id.* at 12–13 (citing Pet. 45). Patent Owner contends that this "argument is unpersuasive" and that "[t]he Petition's statement that the proposed modification would be 'consistent with' Victorian's goal is insufficient" because "Victorian already allows the wearer to hear both sounds in their environment and audio content from a remote device when a gating rule is applied (i.e., not in voice detect mode)." *Id.* at 13.

We do not address this argument because we resolve the institution question based on a failure to sufficiently show limitations 1[a] and 1[b] in Petitioner's challenges applying Victorian, as discussed in detail below.

Analysis of Petitioner's Contentions for Challenges Applying Victorian

Victorian does not describe or depict a "microphone port," and Petitioner, again, does not provide a construction or otherwise explain what "a first microphone port" or "a second microphone port" encompasses. Petitioner again fails to point to a description, definition, figure, or mention in *any* reference, that explains what these claim terms mean.

For the reasons discussed above in Section III.E.2 with respect to the challenges applying Platz, we determine that Petitioner has not met its burden and does not sufficiently show that Victorian alone, or in combination with Inanaga, or in combination with Inanaga and Kvaløy teaches or suggests limitations 1[a] and 1[b] of independent claim 1. As such, Petitioner fails to establish a reasonable likelihood of prevailing with respect to its obviousness challenge to independent claim 1 applying Victorian. Petitioner also fails to establish a reasonable likelihood of

prevailing with respect to its obviousness challenges to claims 2 and 8–11, which depend from claim 1. Petitioner does not cite Inanaga or Kvaløy to cure the deficiencies of Victorian, and these references do not cure the deficiencies of Victorian with respect to independent claim 1.

G. Challenges 3A and 3B Based on Inanaga

Petitioner contends that claims 1 and 11 would have been obvious in view of Inanaga. Pet. 53–62. Petitioner further contends that claims 2 and 8–10 would have been obvious over Inanaga in view of Engle. *Id.* at 62–66. For the reasons that follow, we are not persuaded that Petitioner establishes a reasonable likelihood of prevailing with respect to any of its challenges applying Inanaga.

1. Overview of Inanaga

Inanaga is titled "Earphone Device." Ex. 1007, code (54). Inanaga relates to an earphone device that is capable of reducing external noises. *Id.* at 1:8–10. Figure 4 of Inanaga, reproduced below, shows a block diagram of an earphone device.

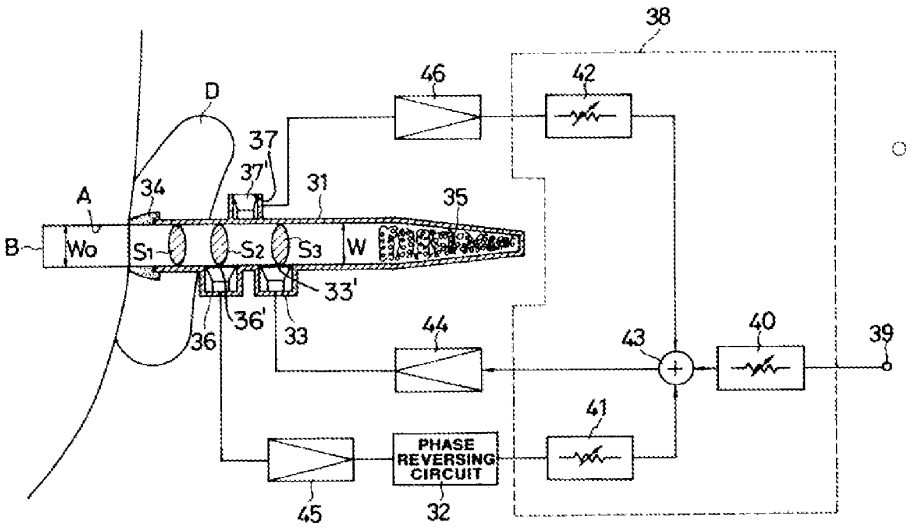


FIG. 4

As shown in Figure 4 of Inanaga, the earphone device comprises acoustic tube 31 which forms an earphone casing and has inner diameter W which is substantially identical with inner diameter W_0 of a user's external auditory canal A. *Id.* at 5:34–37, 5:66. The earphone device further comprises internal microphone unit 36, external microphone unit 37 which picks up external noises, and earphone unit 33 which provides sound to the user's eardrum B via external auditory canal A. *Id.* at 5:34–56, 6:25–27. The earphone device also comprises mixing circuit 38 which is capable of mixing an inputted audio signal (e.g., a music signal from a portable device), a signal obtained by negatively feeding back the signal obtained from internal microphone unit 36, and the external noise signal obtained from external microphone unit 37, at a desired mixing ratio. *Id.* at 5:34–56. Mixing circuit 38 comprises: variable gain control circuits 40, 41, 42 for variably changing the gain of the inputted audio signal, the negatively fed back signal, and the external noise signal; and adder 43 for adding the signals output from variable gain control circuits 40–42 and supplying the output to earphone unit 33. *Id.* at 5:57–65, 6:44–47.

2. Independent Claim 1

[1pre] "An earphone comprising:"

Petitioner contends that "[t]o the extent the preamble is limiting, *Inanaga* discloses or suggests this feature" because "*Inanaga* discloses an 'earphone device.'" Pet. 53 (citing Ex. 1002 ¶ 187; Ex. 1007, 2:29–50, Abstract, Fig. 4).

[1a] "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;"*
>
> *[1b] "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;"*

Petitioner contends that "*Inanaga* discloses an external microphone unit 37 disposed externally [to] the acoustic tube 31, which picks up external noises (ambient sound) and outputs an external noise signal." Pet. 53 (citing Ex. 1007, 5:34–56, Fig. 4). Petitioner further contends that "a POSITA would have understood *Inanaga*'s microphone 37 facing outward of the tube comprises an external microphone facing away from the user when the earphone is inserted." *Id.* at 54 (citing Ex. 1002¶ 190). Petitioner further contends that "it would have been obvious to incorporate a port into microphone 37, and to orient such a port away from the user to detect desired ambient sounds, for the same reasons discussed" with respect to the challenges applying Platz. *Id.* (citing Ex. 1002 ¶ 190; Pet. § VIII.A.1.b.).

Petitioner further contends that "*Inanaga* discloses an internal microphone unit 36 adjacent to the earphone unit 33 provided on the periphery of the acoustic tube 31 and having a vibration plate 36' facing inward of the tube." *Id.* at 55 (citing Ex. 1007, 5:34–56, Fig. 4). Petitioner further contends that "it would have been obvious to a POSITA to incorporate a port into the microphone 36, and to orient the port to face towards the user, for the same reasons discussed" with respect to the challenges applying Platz. *Id.* (citing Ex. 1002 ¶ 193; Pet. § VIII.A.1.b.).

Dr. Polish testifies that

> it would have been obvious to a person of ordinary skill in the art to incorporate a port into the external microphone 37, and to orient such a port away from the user to detect desired ambient sound, for the same reasons . . . discussed above in Section IX.B.1.b.
>
> . . . .
>
> Moreover, it would have been obvious to a person of ordinary skill in the art to incorporate a port into the internal microphone, and to orient the port towards the user, for the same reasons . . . discussed above in Section IX.B.1.b.

Ex. 1002 ¶¶ 190, 193.

Patent Owner's Argument

Patent Owner contends that Petitioner's "Ground 3 theories fail because it would not have been obvious to change Inanaga's geometry as the Petition proposes" and because "Inanaga's geometry is carefully designed." Prelim. Resp. 16. According to Patent Owner, Petitioner references only its arguments for Platz for explaining why it would have been obvious to incorporate a port Inanaga. *Id.* Patent Owner contends that "[r]egardless of whether that argument has merit with respect to Grounds 1A-1C, it does not transfer to Grounds 3A and 3B, where Inanaga—not Platz—is the reference under consideration." *Id.* at 16–17.

Patent Owner further contends that

> the geometry of Inanaga would not benefit from the change required to satisfy limitation 1[b]. The Petition does not explain what the geometry of Inanaga would be under its proposed modification. Regardless, in order to satisfy limitation 1[b], an axis running through the middle of the port must intersect with the user. Such a change to Inanaga's geometry would not have been obvious. Inanaga discloses why inner microphone unit 36 is disposed where it is:

> ***In order to prevent a change in the acoustic impedance***, the earphone unit 33 and ***the inner microphone unit 36 are mounted on the side of the acoustic tube 31 so that the respective vibration plates 33′ and 36′ are mutually parallel*** and the section area S_1 of the acoustic tube is equal to the respective cross-section areas S_2 and S_3 of the mounting portions of the earphone unit 33 and the microphone unit 36.

Prelim. Resp. 17 (quoting Ex. 1007, 6:5–12 (emphasis added by Patent Owner)) (citing Ex. 2001 ¶ 40; Ex. 1004, 74). According to Patent Owner "the rationale proposed for modifying Platz (which is not prior art), even if correct, does not apply to Inanaga" and "there would be no benefit in changing the orientation of microphone unit 36 so that its axis would intersect with the user" because "disturbing the geometry of the earphone device would negatively impact its acoustic characteristics." *Id.* at 18–19 (citing Ex. 2001 ¶ 44).

For the reasons discussed in detail below, we agree that Petitioner does not sufficiently show limitations 1[a] and 1[b] in its challenges applying Inanaga.

Analysis of Petitioner's Contentions for Challenges Applying Inanaga

Inanaga does not disclose, describe, depict, or mention a "microphone port," and Petitioner, again, does not provide a construction or otherwise explain what a "first microphone port" or a "second microphone port" encompasses. Petitioner again fails to point to a description, definition, figure, or mention, in *any* reference, that explains what these claim terms mean.

For the reasons discussed above in Section III.E.2 with respect to the challenges applying Platz, we determine that Petitioner has not met its burden and does not sufficiently show that Inanaga alone, or in combination with Engle teaches or suggests limitations 1[a] and 1[b] of independent claim 1. As such, Petitioner fails to establish a reasonable likelihood of prevailing with respect to its obviousness challenge to independent claim 1 applying Inanaga. Petitioner also fails to establish a reasonable likelihood of prevailing with respect to its obviousness challenges to claims 2 and 8–11, which depend from claim 1. Petitioner does not cite Engle to cure the deficiencies of Inanaga, and Engle does not cure the deficiencies of Inanaga with respect to independent claim 1.

IV. CONCLUSION

For the foregoing reasons, we conclude that the information presented in the Petition does not establish a reasonable likelihood that Petitioner would prevail in showing that any of the challenged claims of the '666 patent are unpatentable.

V. ORDER

In consideration of the foregoing, it is hereby:

ORDERED the Petition is *denied* as to all challenged claims, and no trial is instituted.

Exhibit OTH-E

IPR2022-01099
Patent 11,244,666 B2

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Exhibit PA-P

US006728385B2

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

(10) **Patent No.: US 6,728,385 B2**
(45) **Date of Patent: Apr. 27, 2004**

(54) **VOICE DETECTION AND DISCRIMINATION APPARATUS AND METHOD**

(75) Inventors: **Olav Kvaløy**, Hundhamaren (NO); **Georg E. Ottesen**, Trondheim (NO); **Viggo Henriksen**, Trondheim (NO); **Sverre Stensby**, Trondheim (NO); **Svein Sørsdal**, Trondheim (NO); **Odd Kr. Ø. Pettersen**, Trondheim (NO); **Jarle Svean**, Trondheim (NO)

(73) Assignee: **Nacre AS**, Trondheim (NO)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 13 days.

(21) Appl. No.: **10/085,097**

(22) Filed: **Mar. 1, 2002**

(65) **Prior Publication Data**

US 2003/0165246 A1 Sep. 4, 2003

(51) **Int. Cl.**[7] **H04R 25/00**

(52) **U.S. Cl.** **381/328**; 381/71.1; 381/71.5; 381/317

(58) **Field of Search** 381/23.1, 71.6, 381/72, 73.1, 312, 314, 315, 317, 318, 320, 321, 322, 330, 331; 704/233; 600/25, 559, 23, 24

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,201,006	A *	4/1993	Weinrich	381/318
5,509,102	A	4/1996	Sasaki	704/219
5,721,783	A *	2/1998	Anderson	381/328
6,023,674	A	2/2000	Mekuria	704/233
6,415,034	B1 *	7/2002	Hietanen	381/71.6
2002/0141602	A1 *	10/2002	Nemirovski	381/328
2002/0143242	A1 *	10/2002	Nemirovski	600/300

* cited by examiner

Primary Examiner—Curtis Kuntz
Assistant Examiner—Brian Ensey
(74) *Attorney, Agent, or Firm*—Young & Thompson

(57) **ABSTRACT**

An apparatus and a method of voice detection and discrimination apparatus for controlling a voice operated system. A protective ear terminal element in the apparatus protects the ear by providing acoustic attenuation. An inner electroacoustic transducer element on an inner side of the ear terminal element detects a first acoustic field and provides a first electronic signal representing the first acoustic field. An outer electroacoustic transducer element on an outer side of the ear terminal element detects a second acoustic field and provides a second electronic signal representing the second acoustic field. An electronics unit is connected with the electroacoustic transducer elements and includes first and second comparison members.

66 Claims, 9 Drawing Sheets

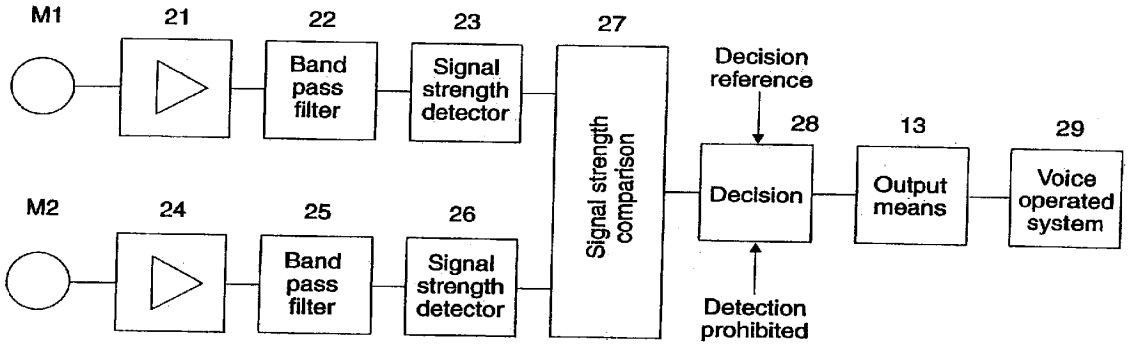


Exhibit PA-P

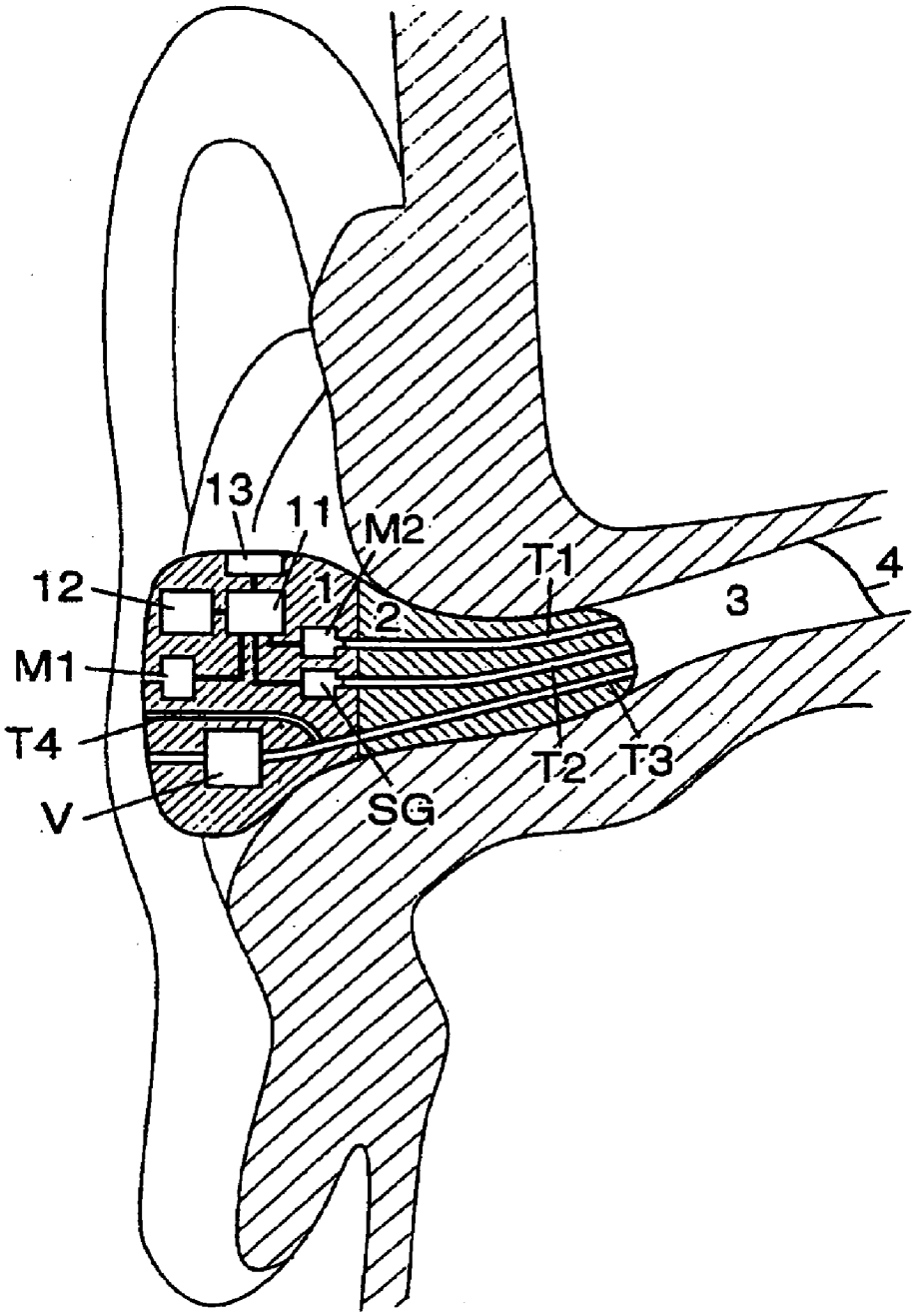


Fig. 1

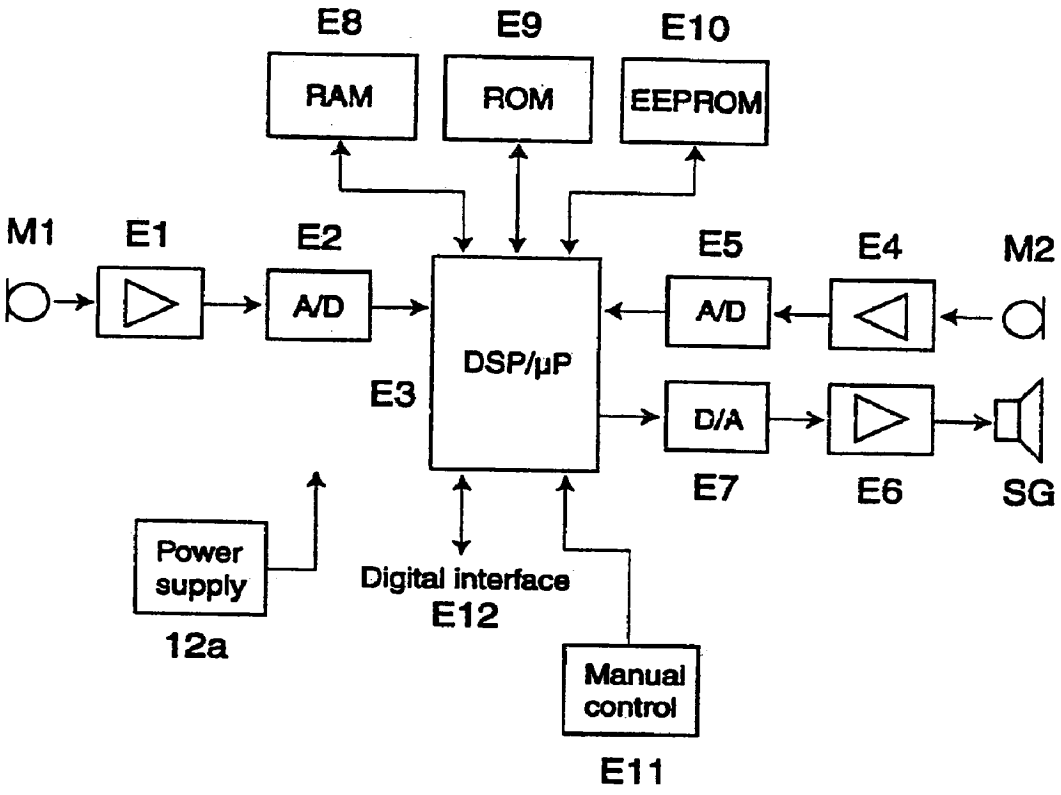

E8
E9
E10
RAM
ROM
EEPROM
M1
E1
E2
E5
E4
M2
A/D
A/D
DSP/µP
E3
D/A
E7
E6
SG
Power supply
12a
Digital interface
E12
Manual control
E11

Fig. 2

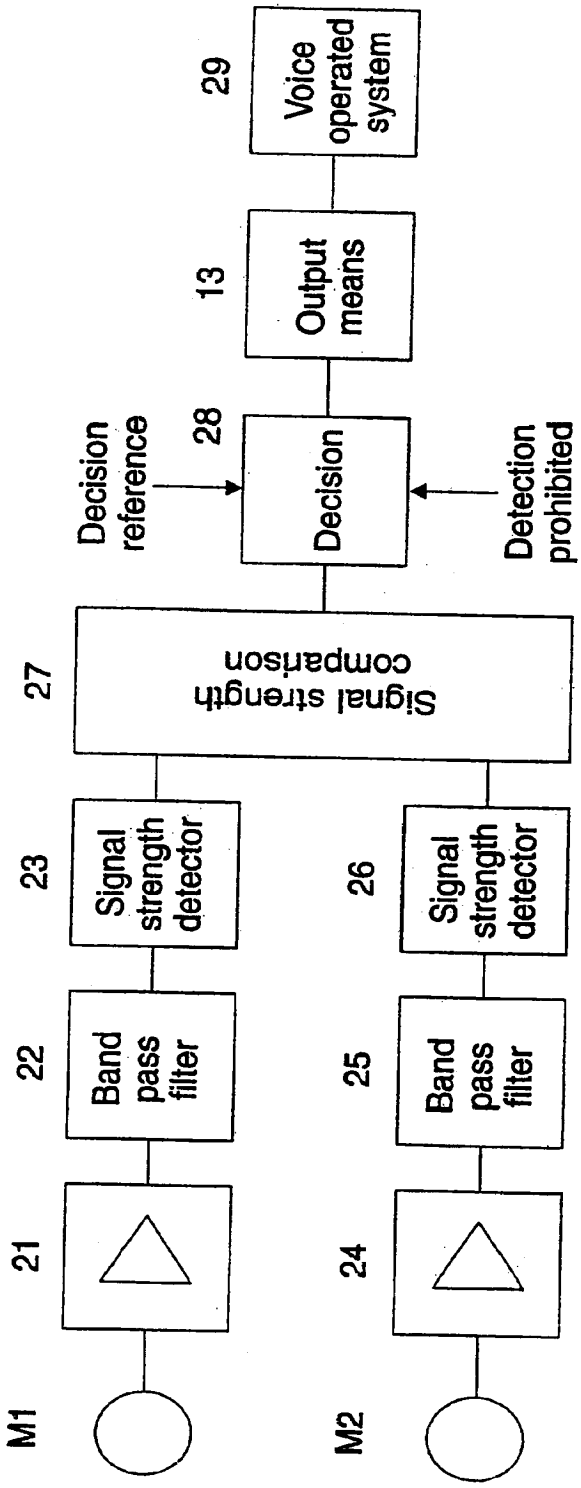

29
Voice operated system
13
Output means
28
Decision
Decision reference
Detection prohibited
27
Signal strength comparison
23
Signal strength detector
26
Signal strength detector
22
Band pass filter
25
Band pass filter
21
24
M1
M2

Fig. 3

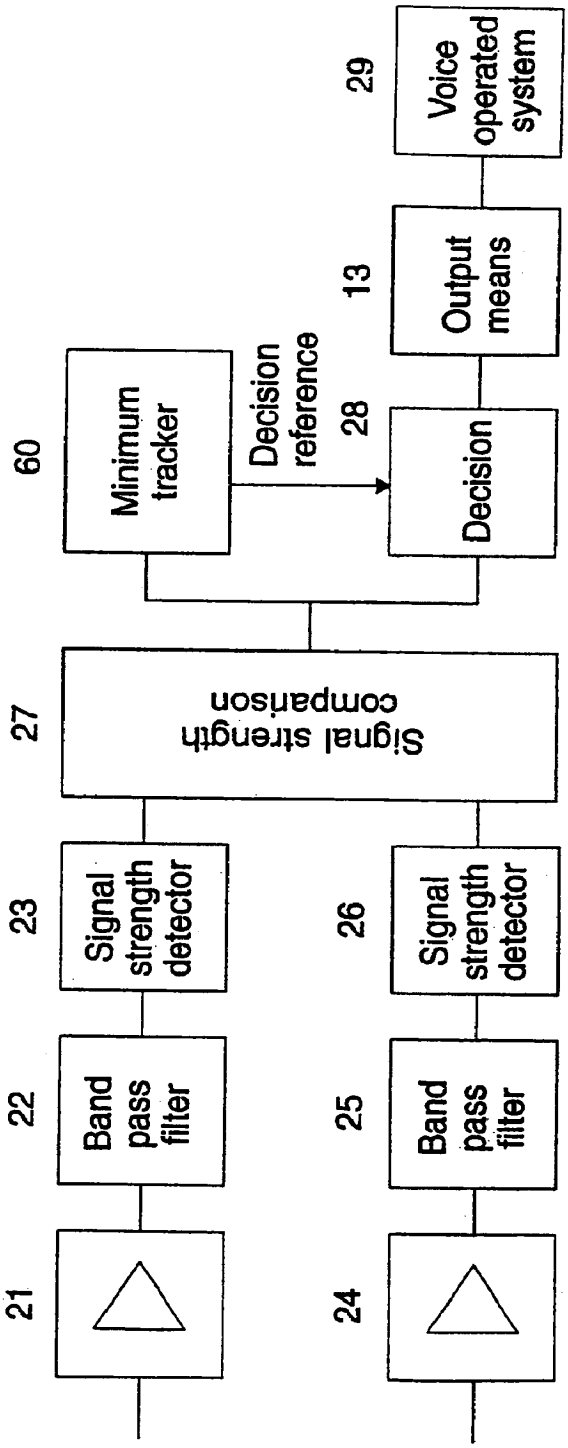

29
Voice operated system
13
Output means
28
Decision
Decision reference
60
Minimum tracker
27
Signal strength comparison
23
Signal strength detector
26
Signal strength detector
22
Band pass filter
25
Band pass filter
21
24

Fig. 4

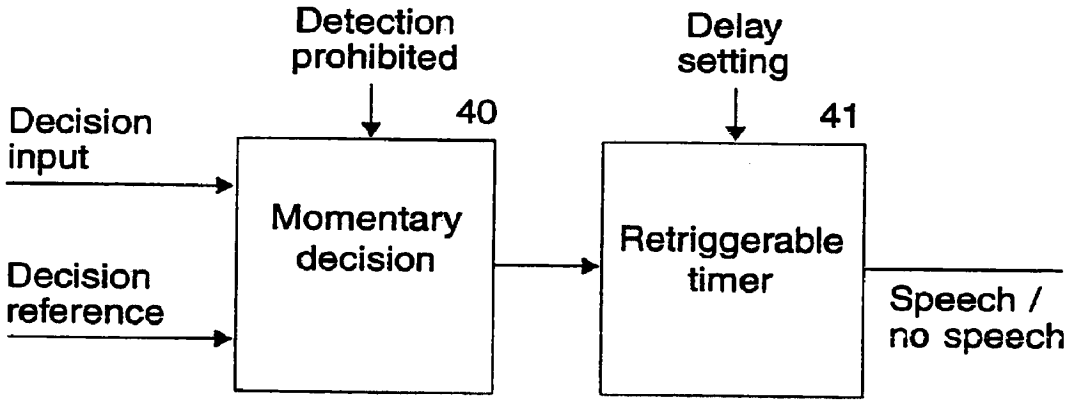

Detection prohibited
40
Delay setting
41
Decision input
Decision reference
Momentary decision
Retriggerable timer
Speech / no speech

Fig. 5

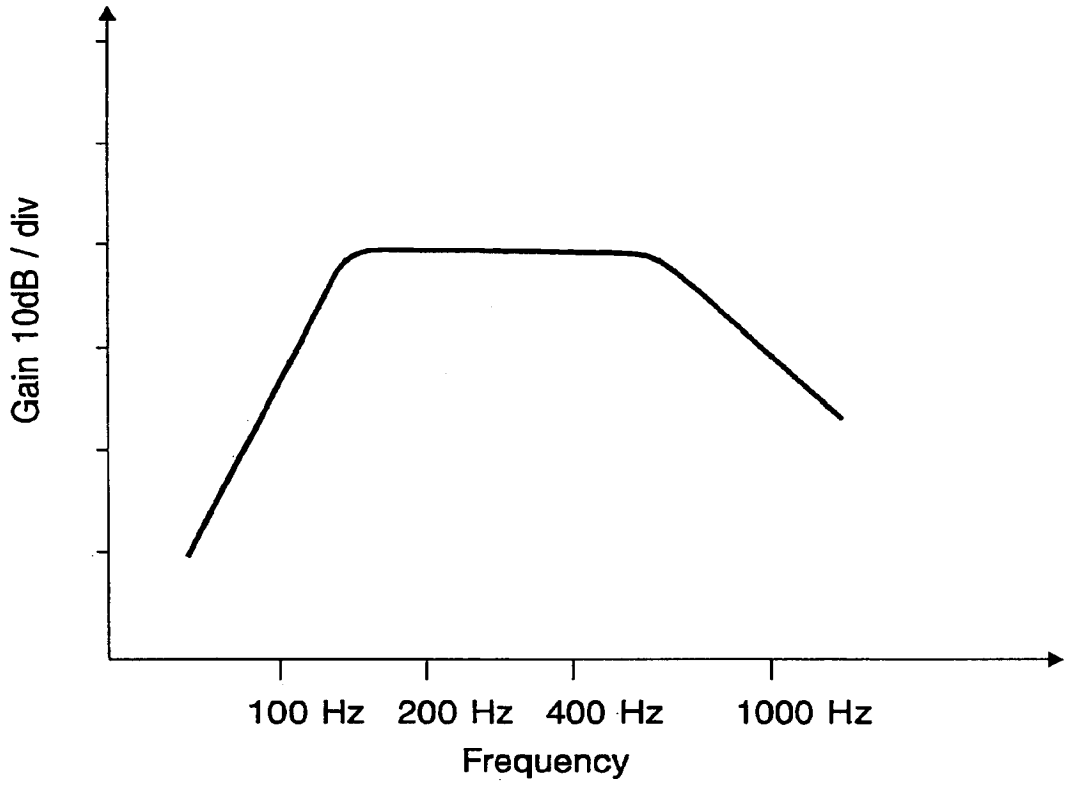


Fig. 6

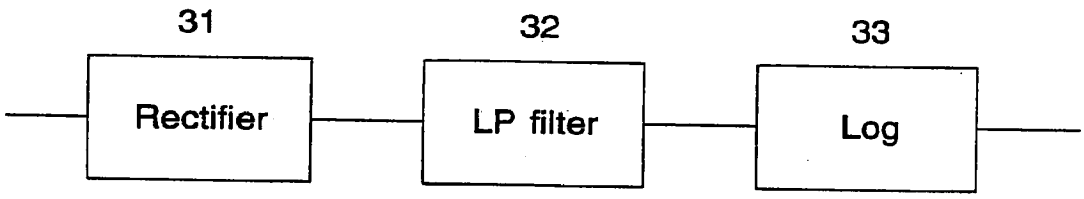


Fig. 7

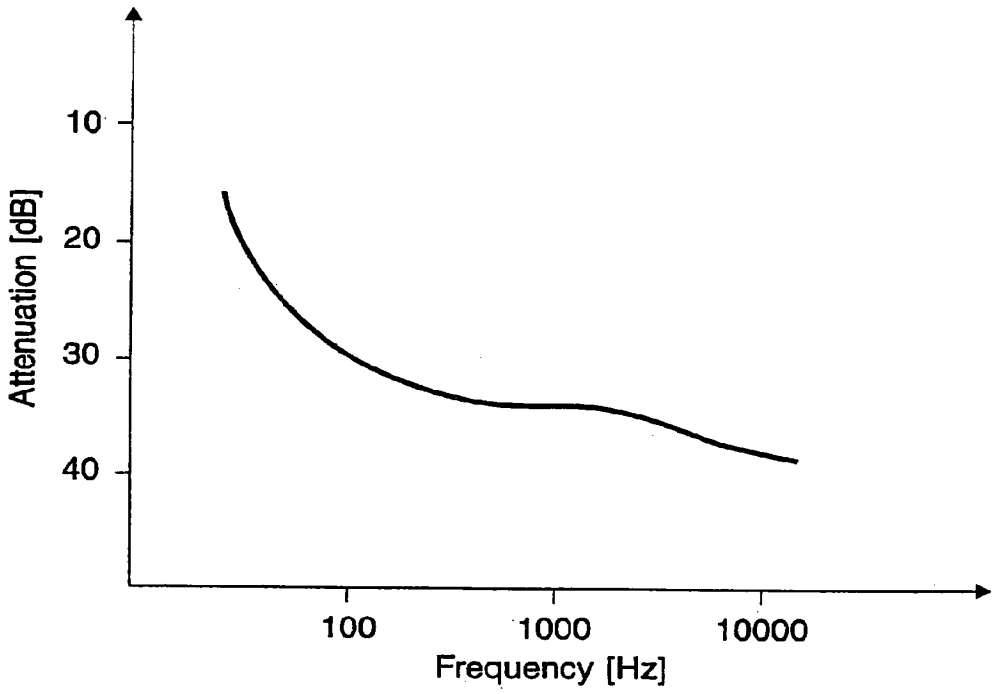


Fig. 8

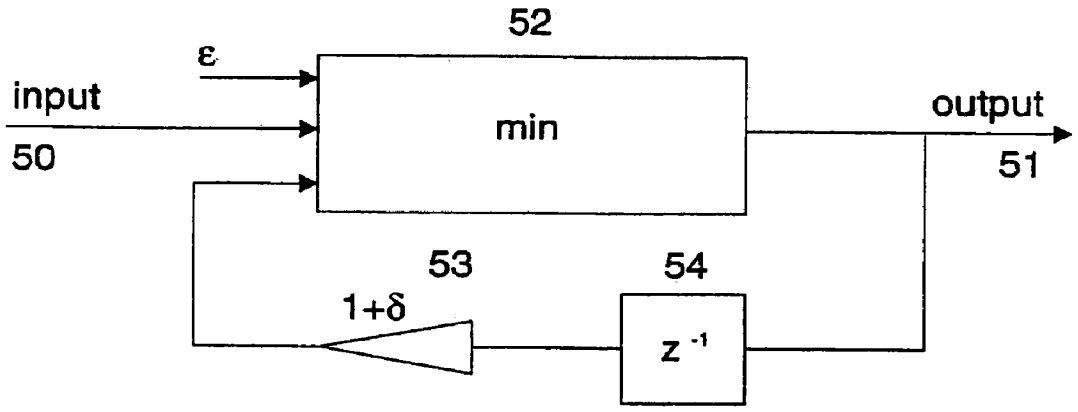


Fig. 9

VOICE DETECTION AND DISCRIMINATION APPARATUS AND METHOD

RELATED APPLICATIONS

The present invention is a further development of the ear terminal described in the international patent applications PCT/NO01/00357, PCT/NO01/00358, PCT/NO01/00359, PCT/NO01/00360, and PCT/NO01/361, which correspond to U.S. application Ser. No.: 09/653,870; Ser. No. 09/653,869; Ser. No. 09/653,868; Ser. No. 09/653,867; Ser. No. 09/653,866; all to be published, and whose subject matter is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention relates to a voice detection and discrimination apparatus in a hearing protection arrangement, and more particularly to a VOX (voice operated transmission/exchange) apparatus, for determining whether an acoustic voice signal is present or absent in a hearing protection arrangement.

The present invention also relates to a method of detecting a voice using the voice detection and discrimination apparatus of the invention in a hearing protection arrangement.

1. Field of Application

Voice activated control is extensively used in communication systems like radio transceivers, intercom systems, recording equipment, etc., and in speech based man-machine-interfaces.

This invention is intended for use in noisy environments, e.g. in environments where some source of acoustic noise is predominant, making it difficult to hear or where damage to the hearing could be at risk. In such environments there could for example be heavy operating machinery or loud vehicle traffic nearby. In other environment there may be large crowds of people. For example in sports stadiums, such as football arenas or the like, where a lot of noise is generated by the audience.

In particular, the primary application for the invention is in situations where it is desirable for people to use a hearing protection arrangement, while still requiring some means of communicating, e.g. to speak with other people or to give commands to voice operated equipment.

2. Prior Art

Present day devices intended to pick up speech from a person in a very noisy environment as a basis for voice detection represent a technological challenge and take several forms. Common types include;

- A microphone in close proximity to the mouth, carried on a microphone boom. The microphone is made with a characteristic emphasising the near field from the mouth. This type is often called "noise cancelling".
- A vibration pickup in contact with the throat, picking up the vibrations of the vocal cord.
- A vibration pickup in contact with the wall of the meatus, the outer ear canal, picking up the vibrations of the tissue in the head.
- A similar pickup in contact with the cheek-bone.
- A microphone picking up the sound in an enclosed space in the inner portion of the meatus.

Detection of voice is based on several techniques:

- Measurement of signal strength of a band-pass filtered microphone/vibration pick-up signal,
- Advanced signal processing on signal picked up by a microphone (A survey of methods are found in Bishnu S. Atal, Lawrence R. Rabiner: "A Pattern Recognition Approach to Voice-Unvoiced-Silence Classification with Applications to Speech Recognition", IEEE Transactions on Acoustics, Speech and Signal Processing, Vol. ASSP-24, No. 3, June 1976, pp 209–212.

Present day devices often fail to work properly in noisy environments. The following types of errors often take place:

- The device is not activated by normal voice
- Noise falsely activates the device in case of no speech

OBJECTS OF THE INVENTION

It is an object of the present invention to provide a voice detection and discrimination apparatus providing much improved performance in noisy environments.

It is a further object of the invention to provide a VOX (voice operated transmission/exchange) apparatus which is capable of detecting and discriminating a voice in a noisy environment.

It is yet an object of the invention to provide a voice detection and discrimination apparatus with improved voice detection capability and which has a reduced false activation due to acoustic noise.

In particular, it is a further object of the invention to provide a VOX (voice operated transmission/exchange) apparatus suitable for use with electronic communication systems being used in noisy environments.

It is also an object of the present invention to provide a method of detecting a voice for the control of a voice operated system, by employing an ear terminal element intended also for protecting the hearing function by providing acoustic attenuation.

SUMMARY OF THE INVENTION

According to the invention, these objectives are achieved in a voice detection and discrimination apparatus for controlling a voice operated system comprising a protective ear terminal element for protecting the ear by providing acoustic attenuation. An inner electroacoustic transducer element on an inner side of the ear terminal element detects a first acoustic field and provides a first electronic signal representing said first acoustic field. An outer electroacoustic transducer element on an outer side of the ear terminal element detects a second acoustic field and provides a second electronic signal representing said second acoustic field. An electronics unit is connected with said electroacoustic transducer elements. The electronics unit comprises first comparison means for comparison of said first and second electronic signals in order to obtain the difference between said two electronic signals. The electronics unit also comprises second comparison means for comparing said difference with given criteria. Output means connected to said electronics unit provides an output signal depending on said second comparison. The output signal is used as an input signal to the voice operated system.

According to the invention the objectives are also achieved with a corresponding method of detecting a persons own voice and for controlling a voice operated system, employing an ear terminal element for protecting the ear by providing acoustic attenuation. The method comprises the following steps: The acoustic signal strength on the inner side of said ear terminal element is detected using a first electroacoustic transducer element. The acoustic signal strength on the outer side of the ear terminal element is detected using a second electroacoustic transducer element.

A difference value representing the difference in the acoustic signal strength between the inner and outer side of the ear terminal element is obtained. Using the value representing the obtained difference and given criteria, it is decided if a voice is present. An output signal depending on the decision is provided using an output means. The output signal is used as an input to a voice operated system.

Further preferable embodiments of the invention are defined in the independent claims.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** illustrates an embodiment of the invention where the voice detection is included in an earplug shaped hearing protective voice communication terminal.

FIG. **2** is a block diagram showing the main functional units of the electronic circuitry of the apparatus according to the invention.

FIG. **3** is a block diagram illustrating one possible signal processing arrangement according to the invention.

FIG. **4** is a block diagram illustrating one possible signal processing arrangement according to the invention where the decision reference signal input is derived from a minimum tracker.

FIG. **5** is a block diagram of the two processing steps of the decision block,

FIG. **6** illustrates a possible frequency response characteristic of the bandpass filters.

FIG. **7** is a block diagram illustrating a preferred embodiment of the signal detection.

FIG. **8** shows a typical sound attenuation characteristic of a hearing protector with a polymer foam sealing section and active noise control (ANR)

FIG. **9** illustrates the operation of a minimum tracker.

DETAILED DESCRIPTION OF THE INVENTION

FIG. **1** illustrates an embodiment according to the invention where the voice detection and discrimination is included in an earplug based hearing protective voice communication terminal **1**,**2**. The earplug **1**,**2** comprises a main section **1** containing two electroacoustic transducer elements M1 and M2 and a sound generator SG. The main section **1** is designed in a way that provides comfortable and secure placement in the concha (the bowl-shaped cavity at the entrance of the ear canal). This may be obtained by using individually moulded ear-pieces **1** that are held in position by the outer ear or by providing at least part of the earplug **1**,**2** with a flexible surrounding pressing against the structure of the outer ear. A sealing section **2** is attached to the main section. The sealing section **2** may be an integral part of the earplug **1**,**2**, or it may be interchangeable. The sound inlet of electroacoustic transducer element M1 is connected to the outside of the earplug **1**,**2**, picking up the external sounds. The electroacoustic transducer element M2 is connected to the inner portion of the meatus **3** by means of an acoustic transmission channel T1. The acoustic transmission channel T1 may contain optional additional acoustic filtering elements. The filtering elements may for instance consist of acoustic resistive elements in the form of porous or sintered inserts, and/or acoustic compliances in the form of small cavities, singly or in combination. An outlet of the sound generator SG is open into the inner portion of the meatus **3** by means of an acoustic transmission channel T2 between the sound generator SG and the inward facing portion of the sealing section **2**. The acoustic transmission channel T2 may also contain optional additional acoustic filtering elements.

When smaller electroacoustic transducer elements M2, and sound generators SG are available, it will be possible to mount the electroacoustic transducer element M2 and the sound generator SG at the innermost part of the sealing section **2**. Then there is no need for the transmission channels T1 and T2.

The two electroacoustic transducer elements M1, M2 and the sound generator SG are connected to an electronics unit **11**, which may be connected to other equipment by a connection interface **13** that may transmit digital or analogue signals, or both, and optionally power.

Electronics **11** and a power supply **12**, e.g. a battery, may be included in main section **1** or in a separate section.

One or both of the electroacoustic transducer elements M1, M2 may in a preferred embodiment be microphones, such as standard miniature electret microphones like the ones used in hearing aids. Recently developed silicon microphones may also be used.

The sound generator SG may in a preferred embodiment be based on the electromagnetic or electrodynamic principle, like sound generators applied in hearing aids.

The main section of the earplug **1**, **2** may be made of standard polymer materials that are used for ordinary hearing aids. The sealing part **2** may be made of a resilient, slowly re-expanding shape retaining polymer foam like PVC, PUR or other materials suitable for earplugs.

For some applications (less extreme noise levels) the earplug may be moulded in one piece **1**,**2** combining the main section **1** and the sealing section **2**. The material for this design may be a typical material used for passive earpugs (Elacin, acryl).

It is also possible to make the earplug **1**,**2** in one piece comprising the main section **1** and the sealing section **2**, all made of a polymer foam mentioned above, but then the channels T1,T2,T3, have to be made of a wall material preventing the channels T1,T2,T3, to collapse when the sealing section **2** is inserted in the meatus **3**. The wall material should be non-porous, typically a plastic or rubber material with a stiffness sufficient to keep the channel open and at the same time allow bending of the channel to conform to the geometry of the meatus.

When the apparatus according to the invention is carried by a user in a noisy environment there will normally be a significant difference in the sound, also referred to as acoustic field or acoustic signal strength between the inside and outside of the apparatus, as detected by the two electroacoustic transducer elements M1 and M2. When the user of the hearing protector speaks, the signal from his voice produces a signal in the inner electroacoustic transducer element M2. The difference in signal strength as detected by the two electroacoustic transducer elements M1,M2 then decreases. When such a decrease in signal difference is detected, this is interpreted as a voice signal being present, and an output for controlling a voice operated system **29** is generated by the output means **13**.

In many applications of an apparatus according to the invention, the surrounding pressure may vary during its use. Hence, an equalisation of pressure between the two sides of the earplug system is required. This is obtained by using a very thin duct T3,T4 or a valve that equalises static pressure differences, while retaining strong low frequency sound attenuation. A safety valve V to take care of rapid decompression may be incorporated in the pressure equalization system T3,T4. The pressure equalisation means T3,T4, V are strictly not required in a basic embodiment of the apparatus according to the invention, but may be an optional feature included in another embodiment of the invention.

FIG. **2** is a block diagram showing the main functional units of the electronic circuitry of the apparatus according to the invention. The electroacoustic transducer element **M1** picks up the ambient sound. A signal from the electroacoustic transducer element **M1** is amplified in the amplifier **E1** and in a basic embodiment of the invention provided directly for the signal processing unit **E3**. The signal from the electroacoustic transducer element **M1** is in a preferable alternative embodiment of the invention sampled and digitised in an analogue to digital converter **E2** and fed to a processing unit **E3** that may be a digital signal processor (DSP), a microprocessor (μP) or a combination of both. A signal from electroacoustic transducer element **M2**, which picks up the sound in the meatus **3** between the isolating section **2** and the tympanum **4**, is amplified in the amplifier **E4**. The amplified signal may either be directly provided for the processing unit **E3** or may be sampled and digitised in the analogue to digital converter **E5** prior to being fed to the processing unit **E3**.

For the case that the apparatus according to the invention is being used to control a voice operated system **29** in the form of a communications system, for example using the techniques described in the abovementioned related applications, it will be useful to include a blocking function, as explained below.

An incoming communication signal may be introduced to the processing unit **E3** through the digital interface **E12**. This communication signal is converted to analogue form in the digital-to-analogue converter **E7** and fed to the analogue output amplifier **E6** that drives the sound generator SG. The sound signal produced by the loudspeaker SG is fed to the tympanum **4** via the canal **T2** into the meatus **3**.

When the incoming communication signal is introduced in the same terminal as is used for voice activated control, it is necessary to apply a blocking function in the form of an additional decision condition signal to the decision process. This additional decision condition signal typically depends on the incoming communication signal. The additional decision condition signal will prohibit or block the detection of the incoming communication signal as if it were the users own voice, during the periods of time when the incoming communication signal is active. This detection prohibition signal is applied to the decision block **28** and the corresponding detail block **40**, in FIGS. **3** and **5**, respectively.

An incoming communication signal may, however, be introduced into an additional hearing protective terminal **(1,2)** located in the ear opposite to the ear accommodating the terminal used for voice activated control. In this case the abovementioned blocking function is strictly not required.

The processing unit **E3** is connected to storage means which may be RAM (Random access memory) **E8**, ROM (read only memory) **E9**, or EEPROM (electrically erasable programmable read only memory) **E10**, or combinations of these. The memories **E8,E9**, and **E10** are in a preferred embodiment of the invention used for storing computer programs, filter coefficients, analysis data and other relevant data.

The storage means **E8,E9,E10** typically contains the criteria to be used by the processing unit **E3** during operation of the device. The criteria may typically comprise data provided during assembly of the apparatus, data provided as part of a calibration procedure, possibly associated with particular users, data obtained from adaptive processes during operation of the device or input data, e.g. provided by a user, via the digital interface **E12**.

The electronic circuitry **11** may be connected to other electrical units by an interface, such as a bi-directional digital interface **E12**. The communication with other electrical units may be performed via a cable or wireless through a digital radio link. The Bluetooth standard for digital short-range radio is one possible candidate for wireless communication for this digital interface **E12**. In a preferred embodiment of the invention, signals that may be transmitted through this interface are:

- program code for the processing unit **E3**
- analysis data from the processing unit **E3**
- synchronisation data when two ear terminals **1,2** are used in a binaural mode
- digitised audio signals in both directions to and from an ear terminal **1,2**.
- control signals for controlling the operation of the ear terminal.
- digital measurement signals for diagnosis of the ear terminal performance.

Based either on the signals received by the electronic circuitry **11** via the communication with other electrical units, on signals stored in the ear terminal element **1,2** itself, or signals detected by the electroacoustic transducer elements **M1,M2** the signal processor **E3** may generate an output signal for the sound generator SG. In the digital version of the invention the digital signal generated in the processing unit **E3** is converted to analogue form in the digital to analogue converter **E7** and fed to the analogue output amplifier **E6** that drives the loudspeaker SG. The sound signal produced by the loudspeaker SG is fed to the tympanum **4** via the canal **T2** into the meatus **3** as described above.

A manual control signal may be generated in the manual control unit **E11** and fed to the processing unit **E3**. The manual control signal may be generated by operating buttons, switches, etc, and may be used to turn the apparatus on and off, to change operation mode, etc. In an alternative embodiment, a voice signal may constitute control signals to the processing unit **E3**. In this case the detected voice signal would typically be compared with a predetermined, e.g. prerecorded, stored representation of a voice signal, such as a digital recording. A manual control signal may also be provided by a remote unit providing output signals adapted to be received by the apparatus, for example via the interface **13**.

The electric circuitry is powered by the power supply **12***a* that may be a primary or rechargeable battery arranged in the earplug or in a separate unit, or it may be powered via a connection to another equipment, e.g. a communication radio.

The block diagram in FIG. **3** illustrates one possible signal processing arrangement according to the invention. The signal processing may be all analogue, or the amplified signals from **M1** and **M2** may be A/D-converted and all filtering and signal processing is performed in the digital domain as illustrated by FIG. **2**. The signal from electroacoustic transducer element **M1** is amplified in amplifier **21** and filtered in a bandpass filter **22** before it is fed to a signal strength detector. Likewise the signal from electroacoustic transducer element **M2** is amplified in amplifier **24** and filtered in bandpass filter **25** before it is fed to a signal strength detector **26**. The outputs from the two signal strength detectors **23,26** are compared in the signal strength comparison unit **27** which also provides as an output a difference signal representing the difference in signal strength from the two electroacoustic transducer elements.

This difference signal output is input to the decision block **28**. The difference signal has a negative dB value. When the

difference signal is less negative than a certain limit a decision is made that the user of the equipment is speaking. When the difference signal is more negative than this limit the user is considered not to be speaking. The difference signal is typically in the range of −20 to −40 dB. Typically a suitable value of the limit will be in the range 2 to 10 dB less negative than a typical value of the difference signal for noise alone, for any single apparatus.

The limit is typically stored in the storage means **E8**,**E9**, **E10** of the apparatus and may be a predetermined value for any single apparatus. The limit is input as the decision reference to to the decision block **28**, as indicated in FIG. **3**.

However, the limit may be generated during use of the device to accomodate for slow drift in the performance of the apparatus. An adaptive process may be performed in the processing unit **E3** in order to obtain this adapted limit. This limit could also be generated in a calibration procedure being performed at regular intervals.

In a similar manner, the limit may be generated during use of the device to accomodate for individual differences between users of the apparatus.

In an alternative the decision reference signal input may be derived using a minimum tracker **60** as illustrated in FIG. **4**. The minimum tracker takes an input signal from the output of the signal strength comparison block **27**.

A possible characteristic of the bandpass filters is shown in FIG. **6**. The diagram shows frequency response with upper and lower cut-off frequencies of 150 Hz and 700 Hz respectively, a highpass slope of +18 dB/octave and a lowpass slope of −6 dB/octave. This frequency characteristic selects a frequency range where the signal strength of the user's voice in the enclosed space in the inner part of the meatus is high. At the same time it suppresses low frequency noise that may be dominating in typical environments (vehicles, factories, etc.)

One preferred embodiment of the signal strength detection **23**,**26** is illustrated in FIG. **7**. The bandpass filtered signal from each of the bandpass filters **23**,**25** is rectified in a rectifier unit **31** and passed on to a lowpass filter **32**. A suitable time constant of the lowpass filter is 10 ms. The output signal of the lowpass filter **32** is input to a logarithmic converter **33**. The logarithm of the signal from **32** is in a digital version of the invention calculated and output to the comparison block **27**. In an analog version the logarithm of the signal from **32** is obtained using an analog logarithmic converter **33**. The outputs from the logarithmic converters **33** are fed to the signal strength comparison unit **27**.

In the signal strength comparison unit **27** there may be performed a running subtraction of the logarithmic value that is output from the signal strength detector **23** from the logarithmic value that is output from the signal strength detector **26**.

If such a signal subtraction is performed when there is no voice signal produced by the user, the result of the subtraction provides a measure of the attenuation of the hearing protecting function of the apparatus (**1**,**2**) according to the invention.

If in addition such a running subtraction is performed prior to use of the apparatus, it is possible to obtain a calibration value, being a typical attenuation value of the attenuation of the hearing protecting function of the apparatus (**1**,**2**) according to the invention. Such a calibration value would preferably be stored internally in storage means (**E8**,**E9**,**E10**) in the apparatus. In some applications such a calibration could be performed for an intended user of the apparatus.

When such a calibration operation has been performed for an apparatus and for a particular user, a later signal subtraction performed when there is no voice signal produced by the user, the result of the subtraction will be a verification of the continued correct operation of the sound attenuation function of the hearing protecting function of the apparatus (**1**,**2**) according to the invention. Correct operation is verified if the result of the subtraction is roughly equal to the value obtained in the calibration operation.

If a calibration operation is performed in a controlled environment with a controllable noise signal generator, it is possible to obtain a calibrated frequency dependent characteristic attenuation of the hearing protecting function of the apparatus (**1**,**2**) according to the invention.

The decision block **28** makes a decision based on the fact that the signal strength difference (calculated in **27**) between electroacoustic transducer elements **M1** and **M2** due to external sounds alone, is independent of the sound character and sound level. It is only dependent on the sound attenuating properties of the hearing protector. These properties are normally independent of sound strength, but dependent on the frequency of the sound. A typical sound attenuation characteristic of a hearing protector with a polymer foam sealing section and active noise control (ANR) is shown in FIG. **8**.

When the user of the hearing protector, being in a noisy environments, speaks, the signal from his voice produces a strong signal in electroacoustic transducer element **M2** (especially in the frequency range 100 Hz–1 kHz), and the difference, as measured by block **27** in FIG. **3**, is diminished. When the frequency dependent attenuation of the hearing protector is known, this gives a very precise functioning of the voice detector compared to a detector that is based on a signal level alone. The detector may act on a limit that is just a few dB less than the attenuation of the hearing protector, without risking false detection due to external noise.

The attenuation characteristic of the hearing protector may be known a priori, or it may be estimated prior to normal use or during use as explained above. A running estimate of the attenuation may be performed by employing a minimum tracker **60** on the signal from the signal strength comparison **27** in FIG. **3**.

The result of the signal strength comparison **28** will typically be a constant or slowly varying signal due mainly to the characteristics of the hearing protector **1**,**2**, with fairly short peaks added due to the speech signal when the user speaks. The speech peaks will typically have a duration of 10–30 milliseconds. The interval between the peaks will typically have a duration of 10–500 milliseconds. The decision block **28** may accordingly contain two signal processing steps, the first step **40** being a momentary decision comparing the composite input signal with a reference signal, the second step **41** being a retriggerable timer with a fixed delay of typically 500 milliseconds. The purpose of the retriggerable delay is to bridge the interval between the speech peaks. The output from the retriggerable delay constitutes the final signal signifying the presence of speech.

FIG. **9** illustrates the operation of the minimum tracker **60**. The minimum tracker **60** outputs the minimum value of the input signal as a function of time. When the input signal is higher than the output signal, the output value increases upwards slowly until it reaches the input value.

The minimum tracker **60** is preferably implemented as a digital filter with a first input **50** to a minimum (min) function block **52** having an output **51**. In a feedback loop of the min function block **52** a sample delay block **54** provides a sample delay 1/z, a multiplication block **53** with a multiplication factor 1+δ provides a multiplication function, the output of which provides a second input to the

min function block **52**. A third input to the min unction block **52** is provided by a minimum value input **55** set to a minimum value ϵ. The min function block **52** outputs the smallest value of its three inputs, as indicated in FIG. **9**.

The minimum value ϵ provided to the minimum value input **55** is used to avoid the minimum tracker **60** dropping down to a value of zero. If the minimum tracker **60** enters the zero value, it will not recover from this value. The initial value in the delay should thus be ϵ. The range of the positive constants δ and ϵ depend on the number format (integer or floating point) and the number of bits in the implementation of the digital filter.

Due to the syllabic nature of speech the output of the minimum tracker **60** represents a measure of the acoustic attenuation of the hearing protector. The limit in the decision block **28** may then be adjusted according to the running estimate of the acoustic attenuation.

Part of the electronics unit **11**,E**3** and the output means **13** may be contained in a unit being separate from, but connected to said ear terminal **1**,**2**. In some situations it may be required to use signal processing units located in a separate unit, for example due to limited space in the ear terminal element, or due to additional processing functions in auxiliary signal processing units. The output means may in some applications require radio transmitters with output powers undesirable near the users head. In this case part of the output means could be arranged some distance away the head, but connected using a suitable communications interface.

The processing unit E**3** may comprise signal analysis means for detecting the presence of speech components, such as words, in the signal from the inner electroacoustic transducer element M**2**. Upon detecting certain components in the signal particular commands, instructions or code may be fetched from the storage means E**8**,E**9**,E**10** for transmission to the voice operated system using the output means **13**. The signal analysis means may also comprise means for determining the duration, frequency content, and amplitude of the signal from the inner electroacoustic transducer element M**2**. In particular, the signal analysis means comprises means for separating the voice signal from the total signal detected by the inner electroacoustic transducer element. The signal analysis is typically realised as software modules performing a combination of signal processing functions, such as digitial filtering.

What is claimed is:

1. Voice detection and discrimination apparatus for controlling a voice operated system comprising a protective ear terminal element (**1**,**2**) for protecting the ear by providing acoustic attenuation,

an inner electroacoustic transducer element (M**2**) on an inner side of the ear terminal element (**1**,**2**) for detecting a first acoustic field and providing a first electronic signal representing said first acoustic field,

an outer electroacoustic transducer element (M**1**) on an outer side of the ear terminal element (**1**,**2**) for detecting a second acoustic field and providing a second electronic signal representing said second acoustic field,

an electronics unit (**11**,E**3**) connected with said electroacoustic transducer elements (M**1**,M**2**) and comprising first comparison means (**27**) for comparison of said first and second electronic signals to obtain the difference between said two electronic signals, and second comparison means (**28**) for comparing said difference with given criteria, and

output means (**13**,E**12**) connected to said electronics unit for providing an output signal depending on said second comparison, said output signal is an input signal to the voice operated system (**29**),

wherein the criteria comprises a predetermined difference in acoustic signal strength as detected by the two electroacoustic transducer elements (M**1**,M**2**) in a noisy environment.

2. Apparatus according to claim **1** wherein the output signal comprises instructions, commands, code for said voice operated system, such as a communication system.

3. Apparatus according to claim **1** wherein the criteria are predetermined and stored in storage means (E**8**,E**9**,E**10**) in the apparatus.

4. Apparatus according to claim **1** comprising processing means (E**3**) for adapting the criteria during use of the apparatus.

5. Apparatus according to claim **1** wherein the predetermined difference in acoustic signal strength as detected by the two electroacoustic transducer elements (M**1**,M**2**) in a noisy environment is in the range −20 to −40 dB, and a predetermined allowable decrease of the said difference in acoustic signal strength is in the range of 2 to 10 dB.

6. Apparatus according to claim **1** comprising a first electronic filter (**22**) for filtering the electronic signal from the inner electroacoustic transducer element (M**2**).

7. Apparatus according to claim **1** comprising a second electronic filter (**25**) for filtering the electronic signal from the outer electroacoustic transducer element (M**1**).

8. Apparatus according to claim **6** comprising electronic bandpass filters (**22**,**25**), typically with a passband in the range 150–700 Hz.

9. Apparatus according to claim **8** comprising a signal strength detector (**23**,**26**) connected with said bandpass filters (**22**,**25**) for providing inputs for a signal strength comparison block (**27**).

10. Apparatus according to claim **9** comprising a decision block (**28**) connected with said signal strength comparison block (**27**) for providing said output signal to said voice operated system (**29**).

11. Apparatus according to claim **1**

wherein at least a part of the electronics unit (**11**,E**3**) and the output means (**13**) is contained in a unit being separate from, but connected to said ear terminal (**1**,**2**).

12. Apparatus according to claim **1**

wherein said ear terminal element (**1**,**2**) comprises an ear plug section (**1**) and a sealing section (**2**) forming a hearing protector (**1**,**2**) for placement in an ear and for protection of the hearing function.

13. Voice detection and discrimination apparatus for controlling a voice operated system comprising

a protective ear terminal element (**1**,**2**) for protecting the ear by providing acoustic attenuation,

an inner electroacoustic transducer element (M**2**) on an inner side of the ear terminal element (**1**,**2**) for detecting a first acoustic field and providing a first electronic signal representing said first acoustic field,

an outer electroacoustic transducer element (M**1**) on an outer side of the ear terminal element (**1**,**2**) for detecting a second acoustic field and providing a second electronic signal representing said second acoustic field,

an electronics unit (**11**,E**3**) connected with said electroacoustic transducer elements (M**1**,M**2**) and comprising first comparison means (**27**) for comparison of said first and second electronic signals to obtain the difference between said two electronic signals, and second comparison means (**28**) for comparing said difference with given criteria, and

output means (13,E12) connected to said electronics unit for providing an output signal depending on said second comparison, said output signal is an input signal to the voice operated system (29)

wherein the signal strength comparison means (27) comprises means for estimating the acoustic attenuation of the ear terminal element (1,2) for establishing said predetermined difference.

14. Apparatus according to claim 13 comprising a minimum tracker (60) connected with the detectors (23,26) for obtaining running estimates of the said acoustic attenuation.

15. Apparatus according to claim 1,

wherein the electronics unit (11,E3) comprises signal analysis means (E3) for extracting a voice signal from the inner electroacoustic transducer element (M2), for transmission to the voice operated system.

16. Apparatus according to claim 15,

comprising signal analysis means (E3) for detecting the presence of particular speech components, such as words, in the signal from the inner electroacoustic transducer element (M2), for forming commands, instructions or code for the voice operated system (29).

17. Apparatus according to claim 1

wherein the electroacoustic transducer element(s) (M1, M2) is(are) microphones.

18. Method of detecting a voice and for controlling a voice operated system, employing an ear terminal element (1,2) for protecting the ear by providing acoustic attenuation comprising the following steps;

detecting (M2) the acoustic signal strength on the inner side of said ear terminal element,

detecting (M1) the acoustic signal strength on the outer side of said ear terminal element,

obtaining (23,26,27) a difference value representing the difference in the acoustic signal strength between the inner and outer side of said ear terminal element,

deciding (28) if a voice is present using said value representing said difference and given criteria (28),

providing an output signal depending on said decision using output means (13,E12), the output signal being an input to the voice operated system (29), and

determining if said difference in the acoustic signal strength (27) between the inner and outer side of said ear terminal element (1,2) has decreased more than a given amount.

19. Method according to claim 18 comprising

providing an output signal (13) in the form of a control signal, commands, instructions, code for said voice operated system (29), such as a communication system.

20. Method according to claim 18 comprising

writing and reading predetermined criteria to and from storage means (E8,E9,E10) in the ear terminal element (1,2).

21. Method according to claim 18 comprising

determining if said difference in the acoustic signal strength (27) between the inner and outer side of said ear terminal element (1,2) has decreased more than a given amount, typically 2–10 dB, from a predetermined amount, typically −20 to −40 dB.

22. Method according to claim 18 comprising

filtering (22,25) the signals from the two electroacoustic transducer elements (M2,M1) using bandpass filters (22,25).

23. Method according to claim 18 comprising;

analysing (E3) the voice signal for determining its characteristic, such as duration, frequency, and amplitude,

providing said output signal (13) depending on the said determined voice characteristics.

24. Method according to claim 23 comprising;

obtaining a value representing the running average of the difference in the acoustic signal level between the inner and outer side of the ear terminal element (1,2).

25. Method according to claim 18 comprising

adjusting the said criteria according to the running average of the said difference.

26. Method according to claim 18 comprising

extracting information from the acoustic signals being detected by the electroacoustic transducer elements (M1,M2).

27. Method according to claim 26 comprising

analysing the signal from the electroacoustic transducer elements (M1,M2) for the presence of particular speech components, such as words.

28. Method according to claim 18 comprising

prior to normal use, performing a calibration operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present.

29. Method according to claim 18 comprising

during normal operation, performing a verification operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present, and the obtained attenuation is compared with a predetermined stored attenuation value.

30. Voice detection and discrimination apparatus for controlling a voice operated system comprising

a protective ear terminal element (1,2) for protecting the ear by providing acoustic attenuation,

an inner electroacoustic transducer element (M2) on an inner side of the ear terminal element (1,2) for detecting a first acoustic field and providing a first electronic signal representing said first acoustic field,

an outer electroacoustic transducer element (M1) on an outer side of the ear terminal element (1,2) for detecting a second acoustic field and providing a second electronic signal representing said second acoustic field,

an electronics unit (11,E3) connected with said electroacoustic transducer elements (M1,M2) and comprising first comparison means (27) for comparison of said first and second electronic signals to obtain the difference between said two electronic signals, and second comparison means (28) for comparing said difference with given criteria,

output means (13,E12) connected to said electronics unit for providing an output signal depending on said second comparison, said output signal is an input signal to the voice operated system (29), and

a decision block with a detection prohibition function in order to enable a blocking function in the ear terminal when an incoming communication signal is introduced in the ear terminal.

31. Apparatus according to claim 13 wherein the output signal comprises instructions, commands, code for said voice operated system, such as a communication system.

32. Apparatus according to claim 13 wherein the criteria are predetermined and stored in storage means (E8,E9,E10) in the apparatus.

33. Apparatus according to claim 13 comprising processing means (E3) for adapting the criteria during use of the apparatus.

34. Apparatus according to claim 13 wherein the predetermined difference in acoustic signal strength as detected by the two electroacoustic transducer elements (M1,M2) in a noisy environment is in the range −20 to −40 dB, and a predetermined allowable decrease of the said difference in acoustic signal strength is in the range of 2 to 10 dB.

35. Apparatus according to claim 13 comprising a first electronic filter (22) for filtering the electronic signal from the inner electroacoustic transducer element (M2).

36. Apparatus according to claim 13 comprising a second electronic filter (25) for filtering the electronic signal from the outer electroacoustic transducer element (M1).

37. Apparatus according to claim 35 comprising electronic bandpass filters (22,25), typically with a passband in the range 150–700 Hz.

38. Apparatus according to claim 37 comprising a signal strength detector (23,26) connected with said bandpass filters (22,25) for providing inputs for a signal strength comparison block (27).

39. Apparatus according to claim 38 comprising a decision block (28) connected with said signal strength comparison block (27) for providing said output signal to said voice operated system (29).

40. Apparatus according to claim 13

wherein at least a part of the electronics unit (11,E3) and the output means (13) is contained in a unit being separate from, but connected to said ear terminal (1,2).

41. Apparatus according to claim 13

wherein said ear terminal element (1,2) comprises an ear plug section (1) and a sealing section (2) forming a hearing protector (1,2) for placement in an ear and for protection of the hearing function.

42. Apparatus according to claim 13,

wherein the electronics unit (11,E3) comprises signal analysis means (E3) for extracting a voice signal from the inner electroacoustic transducer element (M2), for transmission to the voice operated system.

43. Apparatus according to claim 42,

comprising signal analysis means (E3) for detecting the presence of particular speech components, such as words, in the signal from the inner electroacoustic transducer element (M2), for forming commands, instructions or code for the voice operated system (29).

44. Apparatus according to claim 13

wherein the electroacoustic transducer element(s) (M1, M2) is(are) microphones.

45. Method of detecting a voice and for controlling a voice operated system, employing an ear terminal element (1,2) for protecting the ear by providing acoustic attenuation comprising the following steps;

detecting (M2) the acoustic signal strength on the inner side of said ear terminal element,

detecting (M1) the acoustic signal strength on the outer side of said ear terminal element,

obtaining (23,26,27) a difference value representing the difference in the acoustic signal strength between the inner and outer side of said ear terminal element,

deciding (28) if a voice is present using said value representing said difference and given criteria (28),

providing an output signal depending on said decision using output means (13,E12), the output signal being an input to the voice operated system (29), and

adjusting the said criteria according to the running average of the said difference.

46. Method according to claim 45 comprising

providing an output signal (13) in the form of a control signal, commands, instructions, code for said voice operated system (29), such as a communication system.

47. Method according to claim 45 comprising

writing and reading predetermined criteria to and from storage means (E8,E9,E10) in the ear terminal element (1,2).

48. Method according to claim 45 comprising

determining if said difference in the acoustic signal strength (27) between the inner and outer side of said ear terminal element (1,2) has decreased more than a given amount, typically 2–10 dB, from a predetermined amount, typically −20 to −40 dB.

49. Method according to claim 45 comprising

filtering (22,25) the signals from the two electroacoustic transducer elements (M2,M1) using bandpass filters (22,25).

50. Method according to claim 45 comprising;

analysing (E3) the voice signal for determining its characteristic, such as duration, frequency, and amplitude,

providing said output signal (13) depending on the said determined voice characteristics.

51. Method according to claim 50 comprising;

obtaining a value representing the running average of the difference in the acoustic signal level between the inner and outer side of the ear terminal element (1,2).

52. Method according to claim 45 comprising

extracting information from the acoustic signals being detected by the electroacoustic transducer elements (M1,M2).

53. Method according to claim 52 comprising

analysing the signal from the electroacoustic transducer elements (M1,M2) for the presence of particular speech components, such as words.

54. Method according to claim 45 comprising

prior to normal use, performing a calibration operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present.

55. Method according to claim 45 comprising

during normal operation, performing a verification operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present, and the obtained attenuation is compared with a predetermined stored attenuation value.

56. Method of detecting a voice and for controlling a voice operated system, employing an ear terminal element (1,2) for protecting the ear by providing acoustic attenuation comprising the following steps;

detecting (M2) the acoustic signal strength on the inner side of said ear terminal element,

detecting (M1) the acoustic signal strength on the outer side of said ear terminal element,

obtaining (23,26,27) a difference value representing the difference in the acoustic signal strength between the inner and outer side of said ear terminal element,

deciding (28) if a voice is present using said value representing said difference and given criteria (28),

providing an output signal depending on said decision using output means (13,E12), the output signal being an input to the voice operated system (29), and

during normal operation, performing a verification operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present, and the obtained attenuation is compared with a predetermined stored attenuation value.

57. Method according to claim 56 comprising

providing an output signal (13) in the form of a control signal, commands, instructions, code for said voice operated system (29), such as a communication system.

58. Method according to claim 56 comprising

writing and reading predetermined criteria to and from storage means (E8,E9,E10) in the ear terminal element (1,2).

59. Method according to claim 56 comprising

determining if said difference in the acoustic signal strength (27) between the inner and outer side of said ear terminal element (1,2) has decreased more than a given amount, typically 2–10 dB, from a predetermined amount, typically −20 to −40 dB.

60. Method according to claim 56 comprising;

filtering (22,25) the signals from the two electroacoustic transducer elements (M2,M1) using bandpass filters (22,25).

61. Method according to claim 56 comprising;

analysing (E3) the voice signal for determining its characteristic, such as duration, frequency, and amplitude,

providing said output signal (13) depending on the said determined voice characteristics.

62. Method according to claim 61 comprising;

obtaining a value representing the running average of the difference in the acoustic signal level between the inner and outer side of the ear terminal element (1,2).

63. Method according to claim 56 comprising

adjusting the said criteria according to the running average of the said difference.

64. Method according to claim 56 comprising

extracting information from the acoustic signals being detected by the electroacoustic transducer elements (M1,M2).

65. Method according to claim 64 comprising

analysing the signal from the electroacoustic transducer elements (M1,M2) for the presence of particular speech components, such as words.

66. Method according to claim 56 comprising

prior to normal use, performing a calibration operation where an estimate of the attenuation of the hearing protector is obtained by determining the difference in the sound levels as detected by the two electroacoustic transducer elements in a period where no voice but only noise is present.

* * * * *

Exhibit PAT-J


US010810989B2

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

(10) **Patent No.: US 10,810,989 B2**
(45) **Date of Patent: Oct. 20, 2020**

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

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(51) **Int. Cl.**

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H04R 1/10	(2006.01)
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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)

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See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,596,902	A *	6/1986	Gilman	H04R 25/505 381/320
5,721,783	A	2/1998	Anderson	
5,867,581	A	2/1999	Obara	
6,415,034	B1	7/2002	Hietanen	
6,754,359	B1 *	6/2004	Svean	H04R 1/1083 381/313

(Continued)

Primary Examiner — Andrew L Sniezek

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(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.

17 Claims, 9 Drawing Sheets

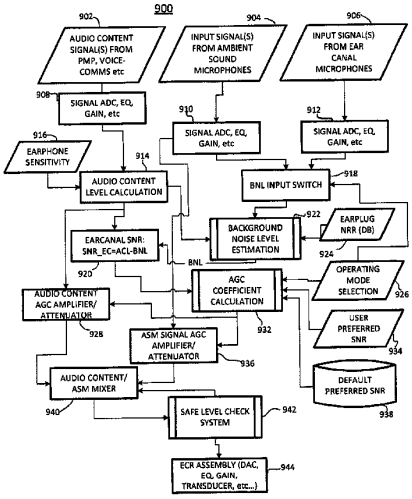


Exhibit PAT-J

(56) **References Cited**

U.S. PATENT DOCUMENTS

7,006,650	B1	2/2006	Wild	
7,929,713	B2	4/2011	Victorian	
8,077,892	B2	12/2011	Dijkstra	
8,098,832	B2	1/2012	Lin	
8,213,629	B2	7/2012	Goldstein et al.	
8,229,127	B2	7/2012	Jorgensen	
8,774,433	B2	7/2014	Goldstein	
8,917,894	B2 *	12/2014	Goldstein	H04R 1/1016 381/328
9,055,377	B2	6/2015	Kinsbergen	
9,137,611	B2	9/2015	Unno	
9,232,322	B2	1/2016	Fang	
9,357,314	B2	5/2016	Goorevich	
9,613,622	B1	4/2017	Melanson	
10,134,377	B2 *	11/2018	Goldstein	G10K 11/17827
10,535,334	B2 *	1/2020	Goldstein	H04R 3/002
2001/0046304	A1	11/2001	Rast	
2005/0058301	A1	3/2005	Brown	
2005/0058313	A1	3/2005	Victorian et al.	
2006/0002574	A1	1/2006	Shennib et al.	
2006/0045282	A1	3/2006	Reber	
2006/0083388	A1	4/2006	Rothchild	
2006/0222194	A1	10/2006	Bramslow et al.	
2006/0262938	A1	11/2006	Gauger, Jr. et al.	
2007/0116310	A1	5/2007	Ludvigsen	
2007/0189544	A1	8/2007	Rosenberg	
2007/0223717	A1 *	9/2007	Boersma	H04M 1/6058 381/74
2008/0159547	A1	7/2008	Schuler et al.	
2008/0181442	A1	7/2008	Goldstein et al.	
2008/0240458	A1	10/2008	Goldstein et al.	
2008/0260191	A1	10/2008	Victorian et al.	
2011/0195676	A1	8/2011	Victorian et al.	
2012/0020485	A1	1/2012	Visser et al.	
2016/0021469	A1	1/2016	Victorian et al.	

* cited by examiner

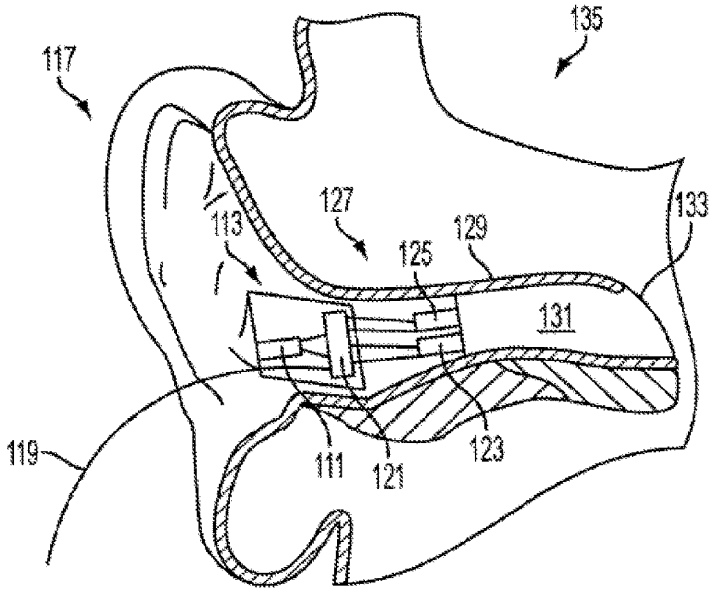


FIG. 1 100

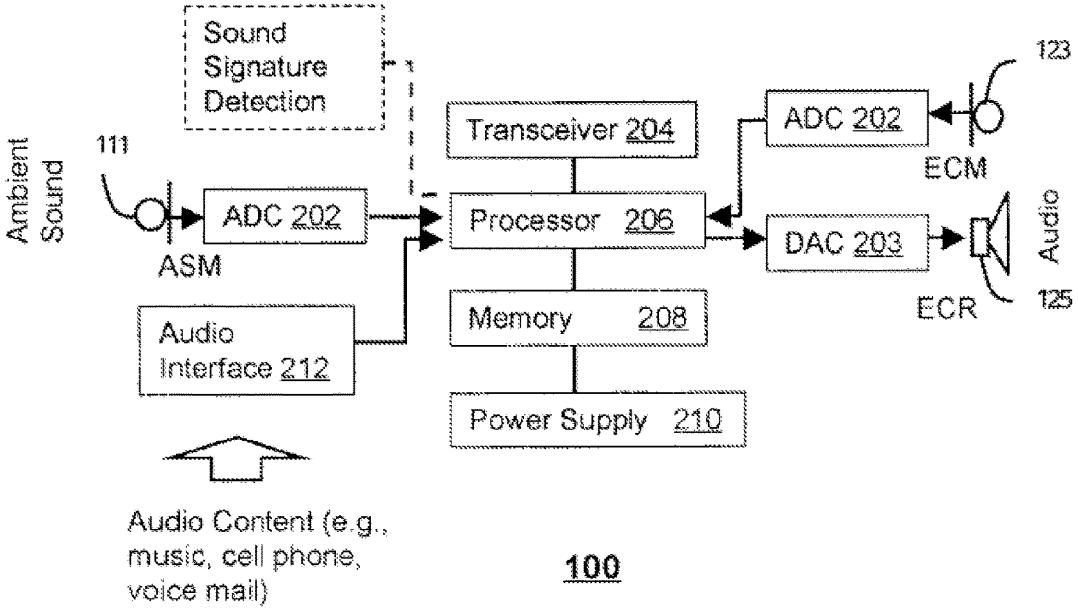
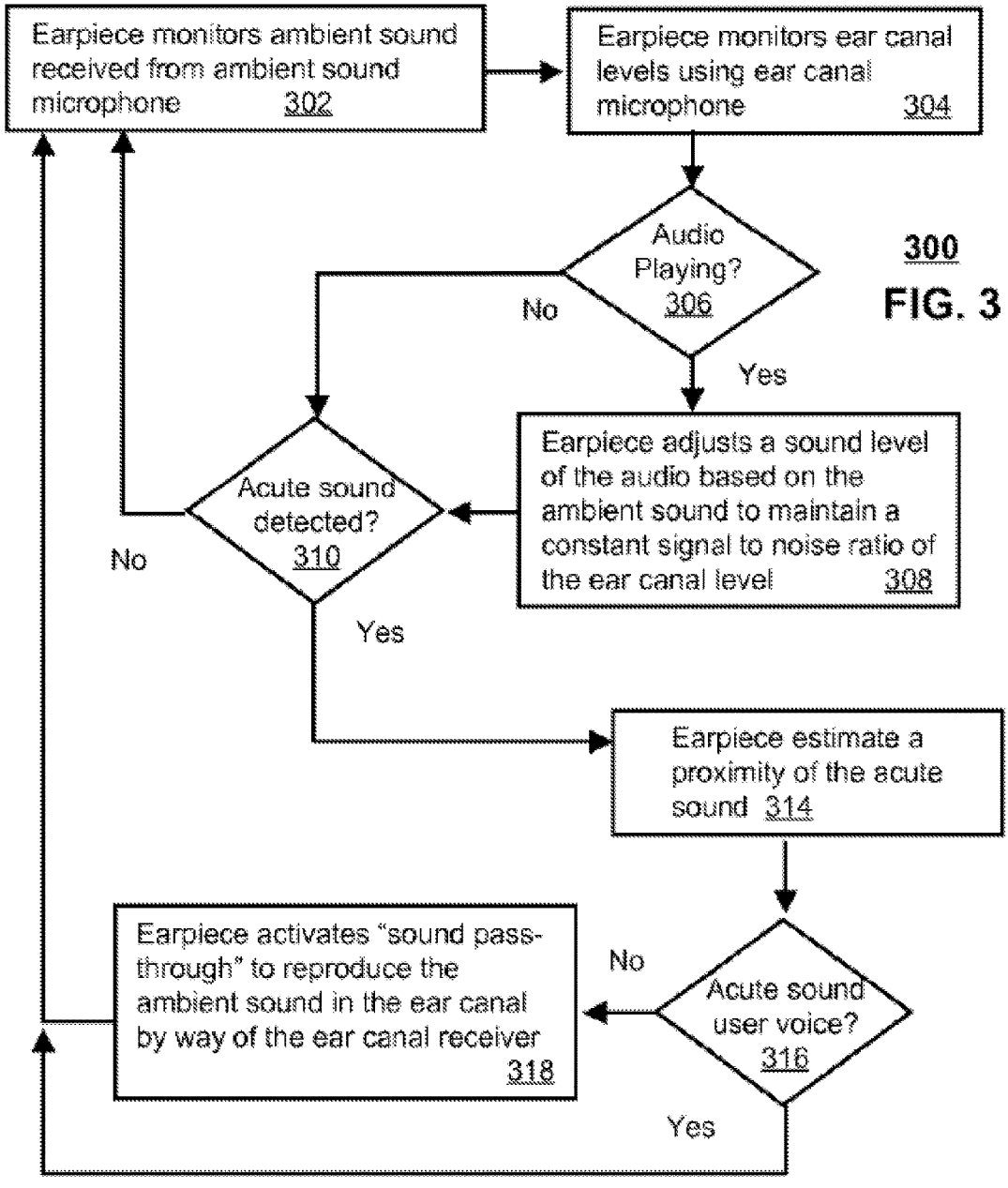


FIG. 2


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

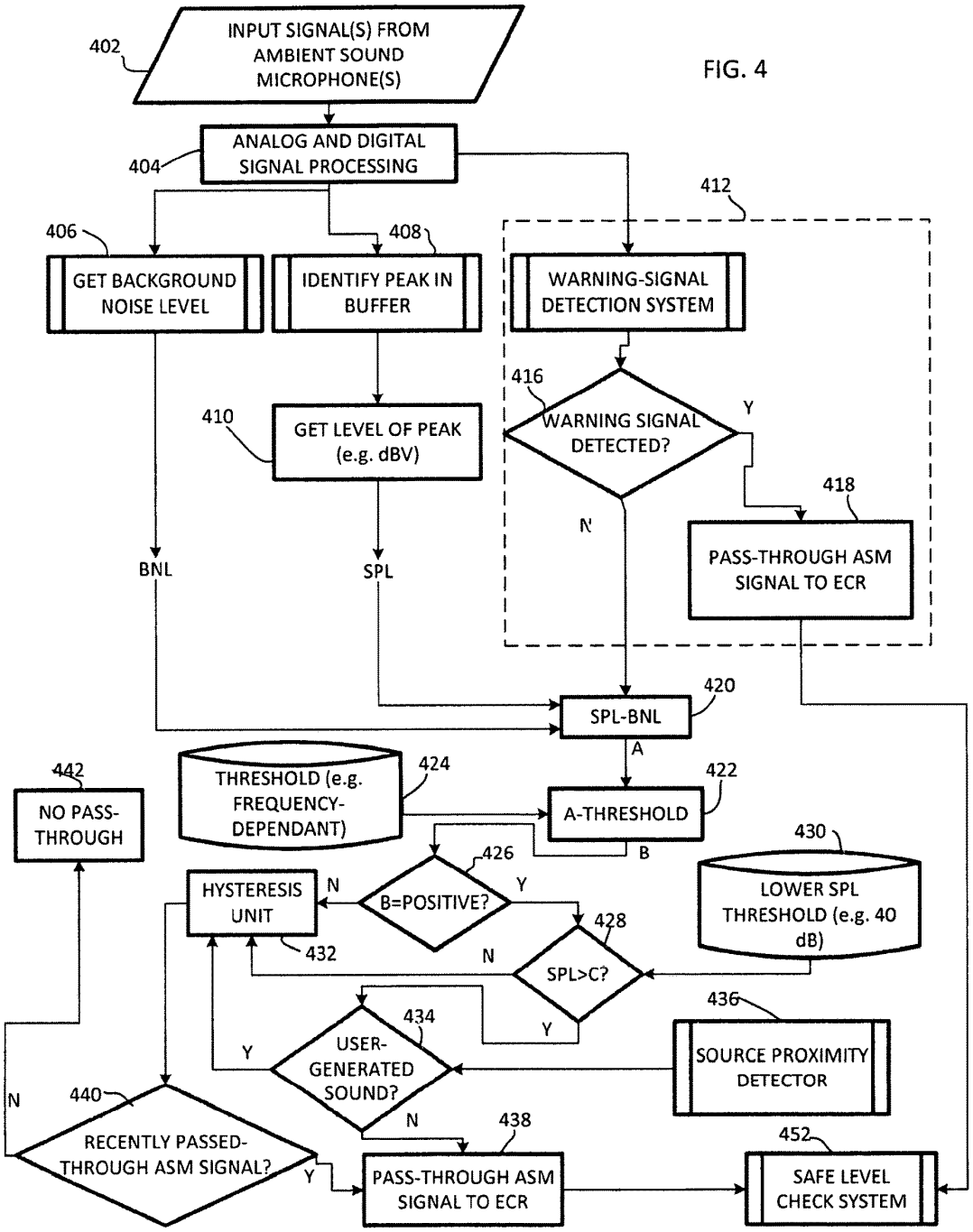
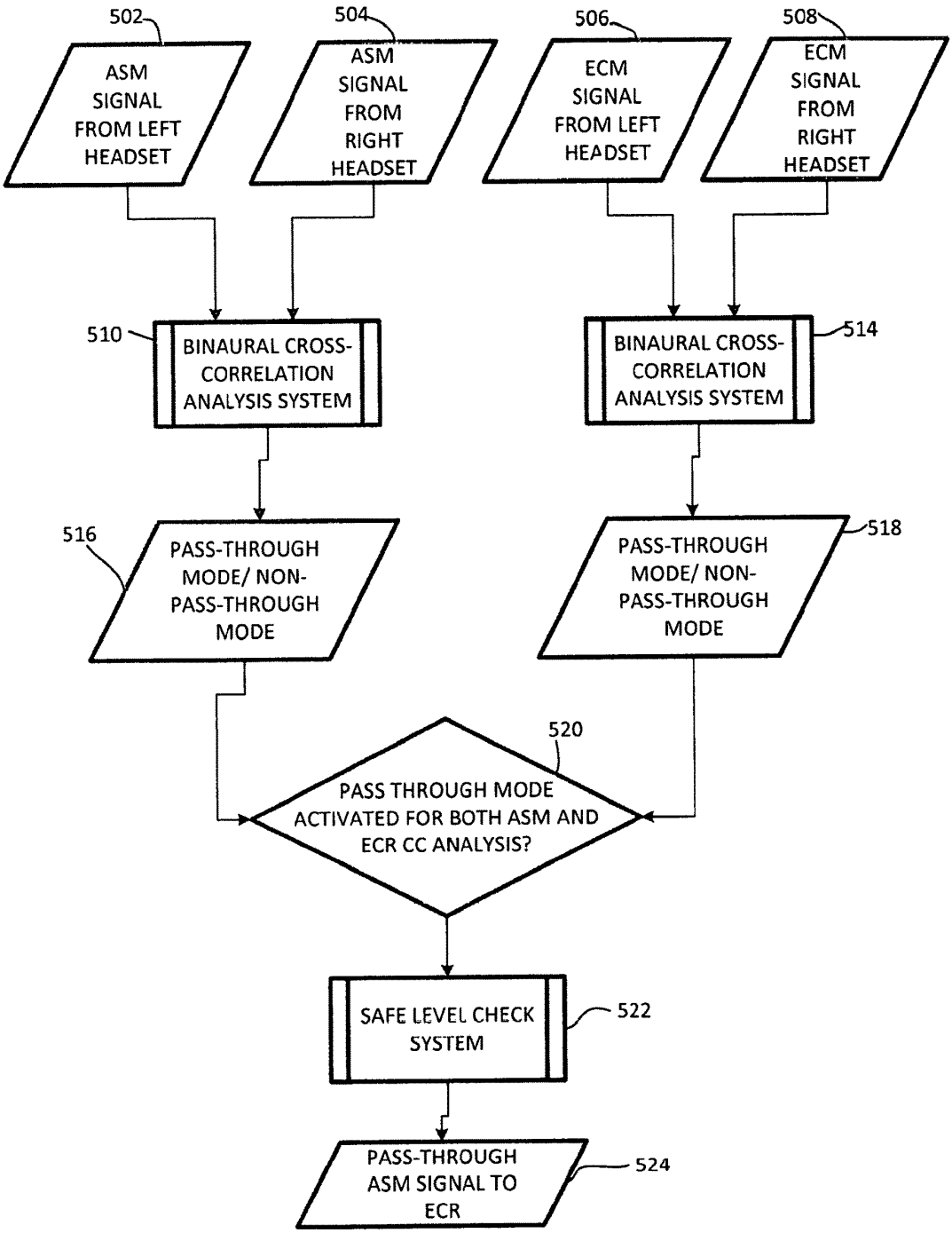

402
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONE(S)
404
ANALOG AND DIGITAL SIGNAL PROCESSING
406
GET BACKGROUND NOISE LEVEL
408
IDENTIFY PEAK IN BUFFER
410
GET LEVEL OF PEAK (e.g. dBV)
412
WARNING-SIGNAL DETECTION SYSTEM
416
WARNING SIGNAL DETECTED?
Y
N
418
PASS-THROUGH ASM SIGNAL TO ECR
BNL
SPL
420
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

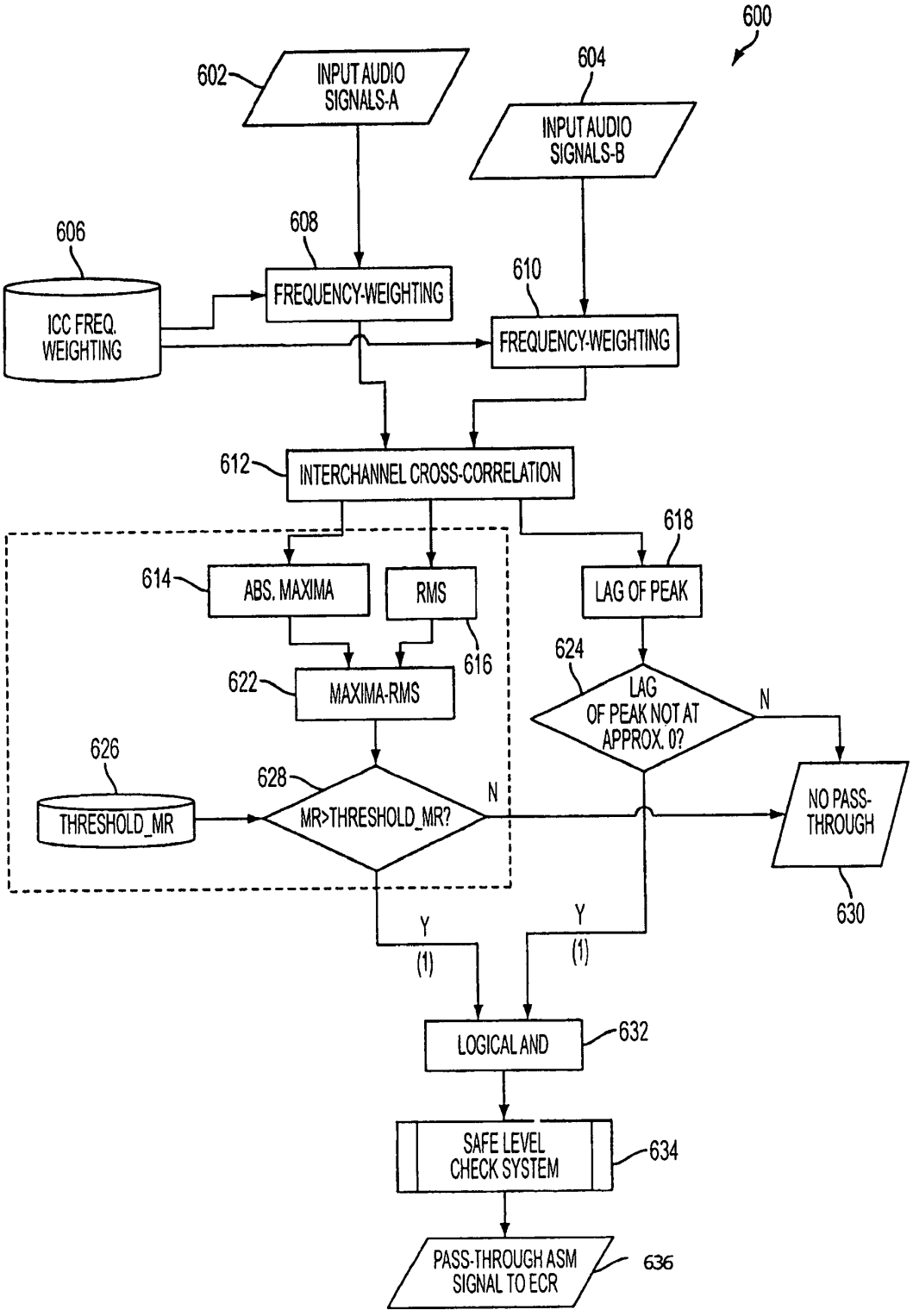
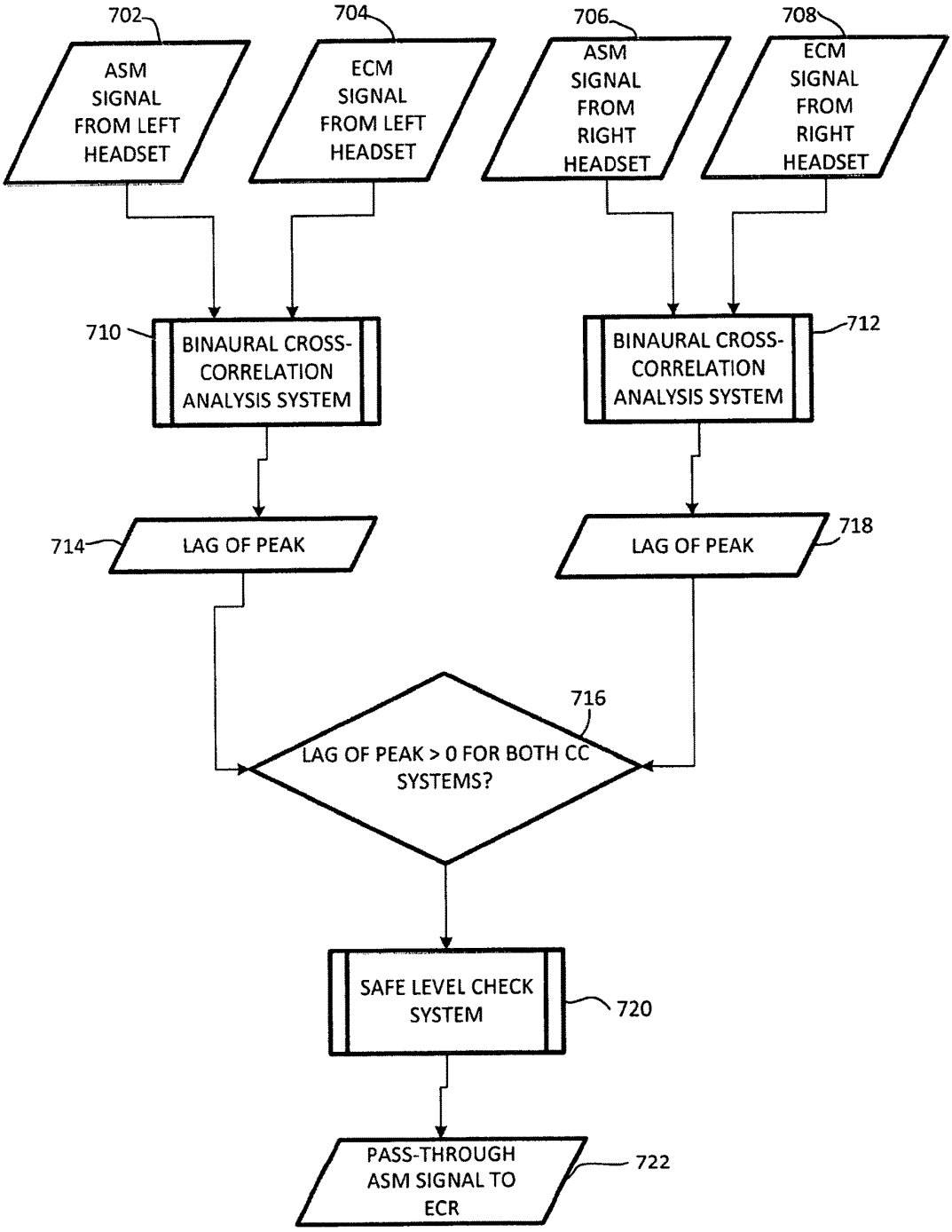

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH
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

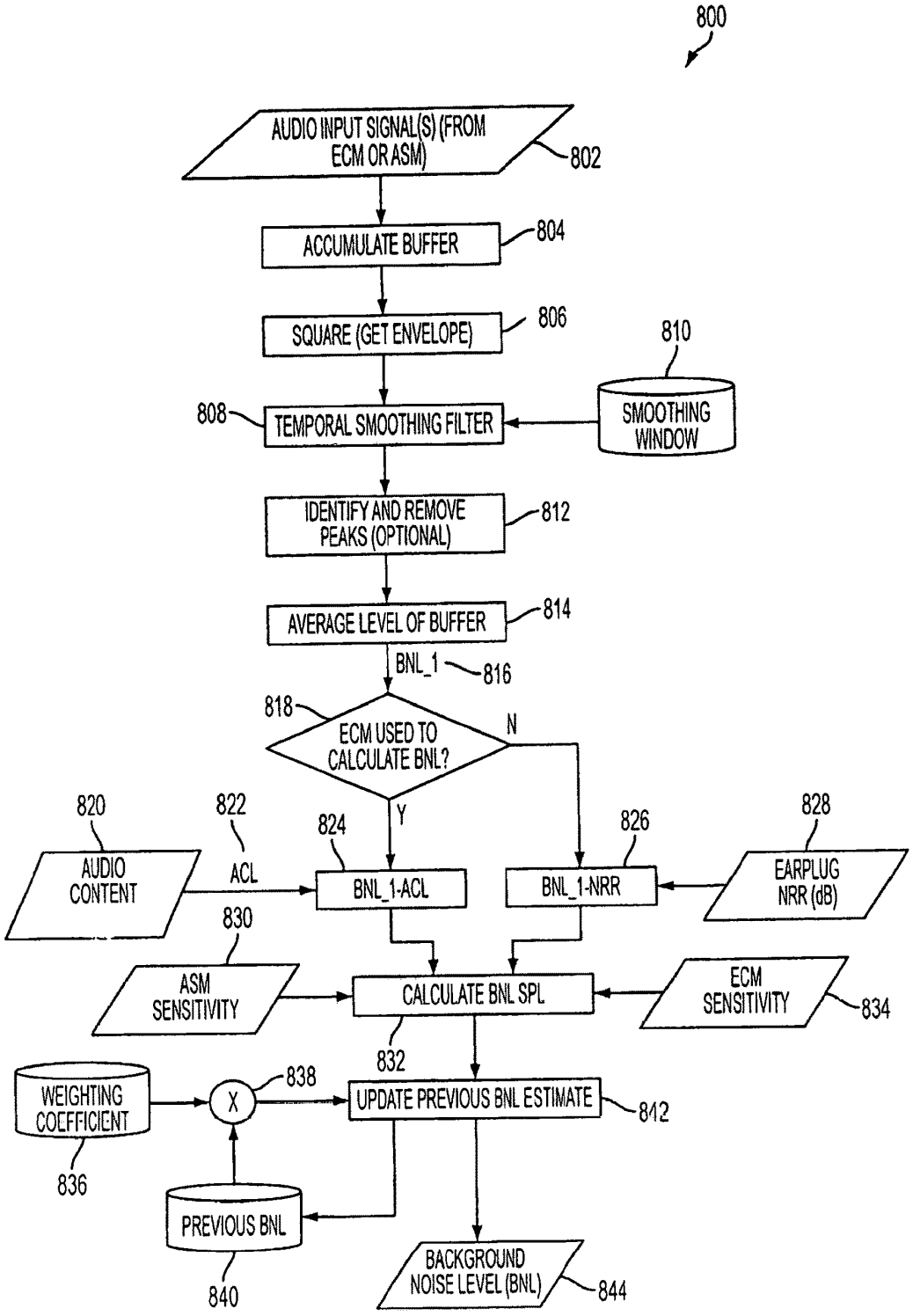

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT
822
ACL
824
BNL_1-ACL
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

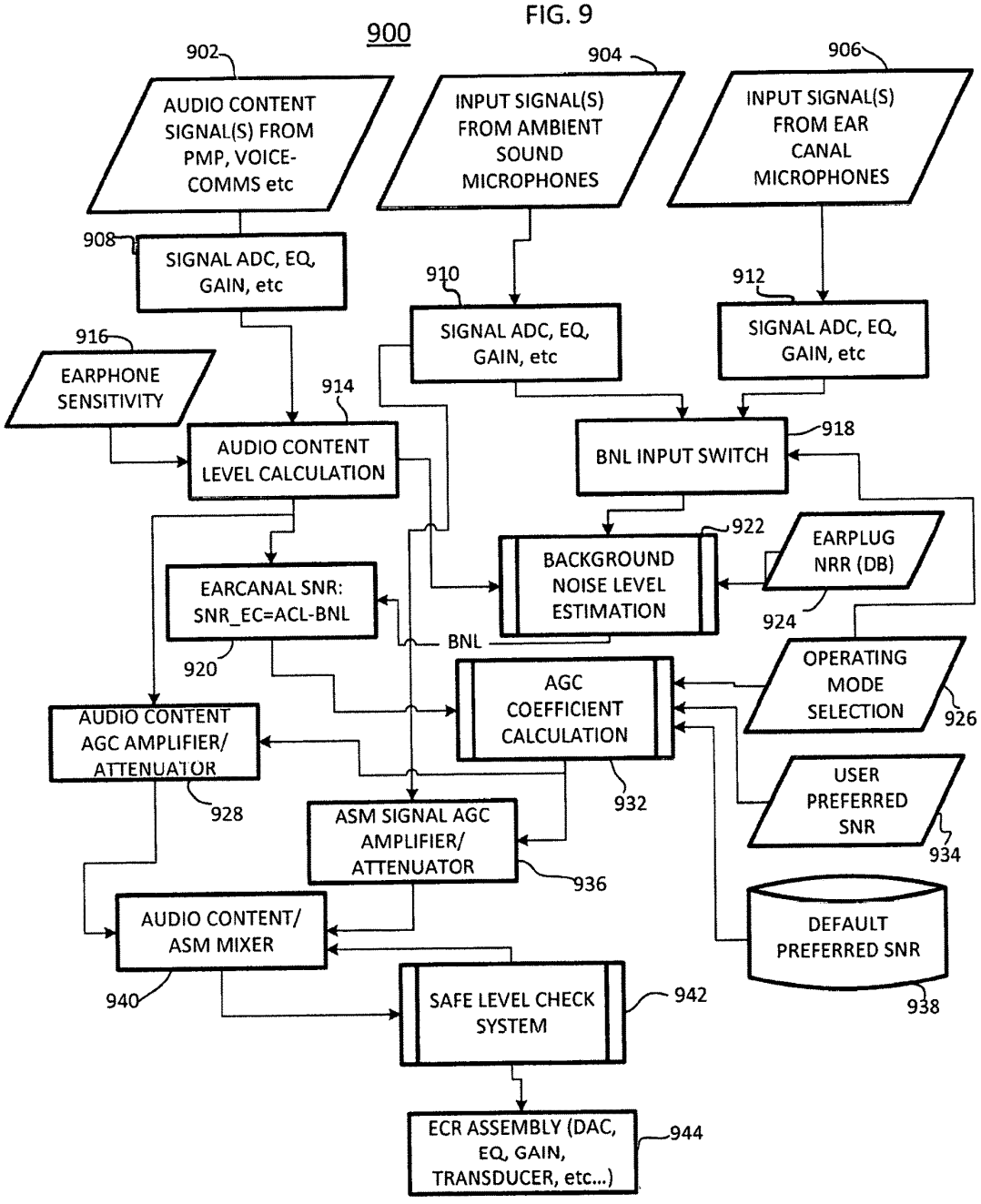
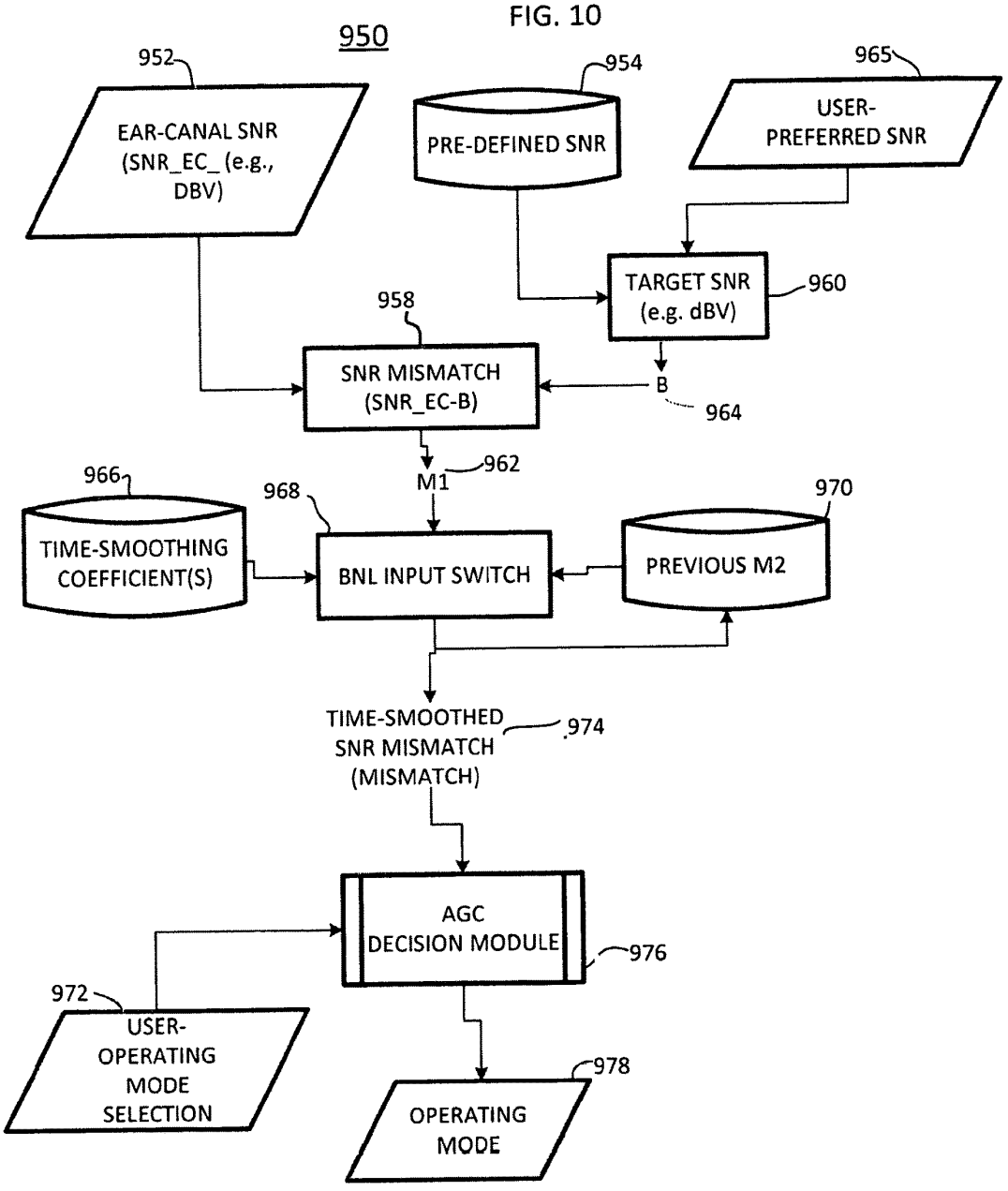

900
902
AUDIO CONTENT SIGNAL(S) FROM PMP, VOICE-COMMS etc
904
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES
906
INPUT SIGNAL(S) FROM EAR CANAL MICROPHONES
908
SIGNAL ADC, EQ, GAIN, etc
910
SIGNAL ADC, EQ, GAIN, etc
912
SIGNAL ADC, EQ, GAIN, etc
916
EARPHONE SENSITIVITY
914
AUDIO CONTENT LEVEL CALCULATION
918
BNL INPUT SWITCH
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. 16/193,568, filed 16 Nov. 2018. U.S. patent application Ser. No. 16/193,568 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) Ill 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 Ill 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 Ill, 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 Ill 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 Ill 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 Ill 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 Ill 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 aRMS 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.

What is claimed:

1. An earpiece configured to pass through sound from an ambient sound microphone comprising:

a housing, where the housing has a tip end and an ambient environment end;

a processor;

a first microphone configured to measure a first acoustic environment closer to the tip end than the ambient environment end, generating a first microphone signal;

a second microphone configured to measure a second acoustic environment closer to the ambient environment end than the tip end, generating a second microphone signal; and

a speaker configured to play an audio content signal, where the processor is coupled to the first microphone, the second microphone and the speaker, and where the processor is configured, if the earpiece has been placed into pass-through mode, to:

modify the first microphone signal to generate a first modified signal;

modify the second microphone signal to generate a second modified signal;

mix the first modified signal, the audio content signal and the second modified signal to generate a mixed signal; and

send the mixed signal to the speaker.

2. The earpiece according to claim **1**, where the earpiece is placed into pass-through mode when the processor detects an acute sound by examining the temporal or spectral characteristics of a portion of the second microphone signal and the examined portion exceeds a threshold value.

3. The earpiece according to claim **2**, where the first microphone is an ear canal microphone.

4. The earpiece according to claim **3**, where the second microphone is an ambient sound microphone.

5. The earpiece according to claim **4**, where the first modified signal is configured to reduce the noise in the first acoustic environment.

6. The earpiece according to claim **5**, where noise is defined as the sound in the first acoustic environment that is not a result of playing the audio content and playing the second modified signal.

7. The earpiece according to claim **6**, where the processor is configured to modify the audio content signal prior to generating the mixed signal so that the mixed signal is generated by mixing the first modified signal, the second modified signal and the modified audio content signal.

8. The earpiece according to claim **7**, further including:

a safe level check system.

9. The earpiece according to claim **8**, where the safe level check system checks if the mixed signal would exceed a safe threshold level if played by the speaker.

10. The earpiece according to claim **9**, where the safe level check system reduces the amplitude of the mixed signal if the mixed signal would exceed a safe threshold level if played by the speaker.

11. The earpiece according to claim **4**, further including a tip, attached to the tip end of the housing.

12. The earpiece according to claim **11**, where the tip is configured to acoustically seal an ear canal when inserted therein.

13. The earpiece according to claim **12**, where the tip provides at least 10 dB acoustic isolation from 500 Hz to 4000 Hz.

14. The earpiece according to claim **7**, where a modified audio signal is a zero signal.

15. The earpiece according to claim **1**, where the earpiece has been placed into pass-through mode by a user operated mode selection.

16. The earpiece according to claim **15**, where the first microphone is an ear canal microphone.

17. The earpiece according to claim **16**, where the second microphone is an ambient sound microphone.

* * * * *

Exhibit PA-R

US008150084B2

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

(10) **Patent No.: US 8,150,084 B2**
(45) **Date of Patent: Apr. 3, 2012**

(54) **HEARING AID AND A METHOD OF PROCESSING A SOUND SIGNAL IN A HEARING AID**

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(73) Assignee: **Widex A/S**, Lynge (DK)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 904 days.

(21) Appl. No.: **11/281,390**

(22) Filed: **Nov. 18, 2005**

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Related U.S. Application Data

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

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

(58) **Field of Classification Search** 381/380–381, 381/312–331; 379/430

See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,073,366 A		2/1978	Estes	
4,311,206 A	*	1/1982	Johnson	181/135
5,058,171 A		10/1991	Wolf	
6,327,370 B1	*	12/2001	Killion et al.	381/313
6,424,721 B1		7/2002	Hohn	
6,473,512 B1	*	10/2002	Juneau et al.	381/328
6,539,096 B1		3/2003	Sigwanz	
6,882,736 B2	*	4/2005	Dickel et al.	381/317
7,003,127 B1	*	2/2006	Sjursen et al.	381/322
7,181,030 B2	*	2/2007	Rasmussen et al.	381/312

FOREIGN PATENT DOCUMENTS

EP	0695108 A2	1/1996
EP	0695108 B1	4/2000
EP	1 196 009 A2	4/2002
WO	WO 02/17838 A1	3/2002

OTHER PUBLICATIONS

Chinese Office Action dated Jul. 3, 2009 for Appln. 03826498I6 with English Translation.

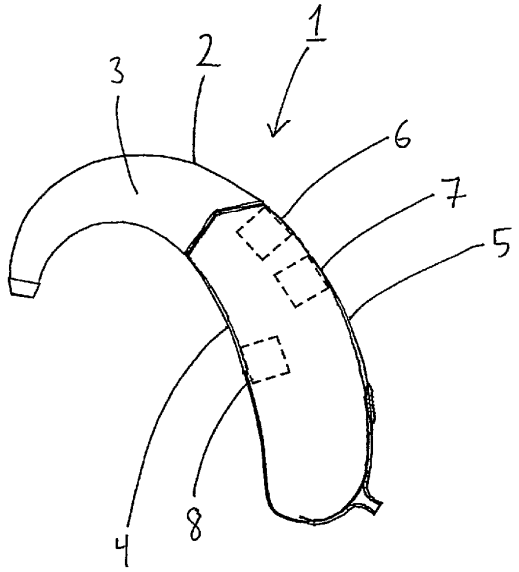
* cited by examiner

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(57) **ABSTRACT**

A hearing aid (**1**) includes a housing (**2**), at least a first microphone for receiving ambient sound, processing circuitry for processing a signal from the first microphone (**6**, **7**) and an output transducer. At least one further microphone (**8**) is disposed in the hearing housing at a location, which, during normal use of the hearing aid, is sheltered against wind influences.

23 Claims, 2 Drawing Sheets



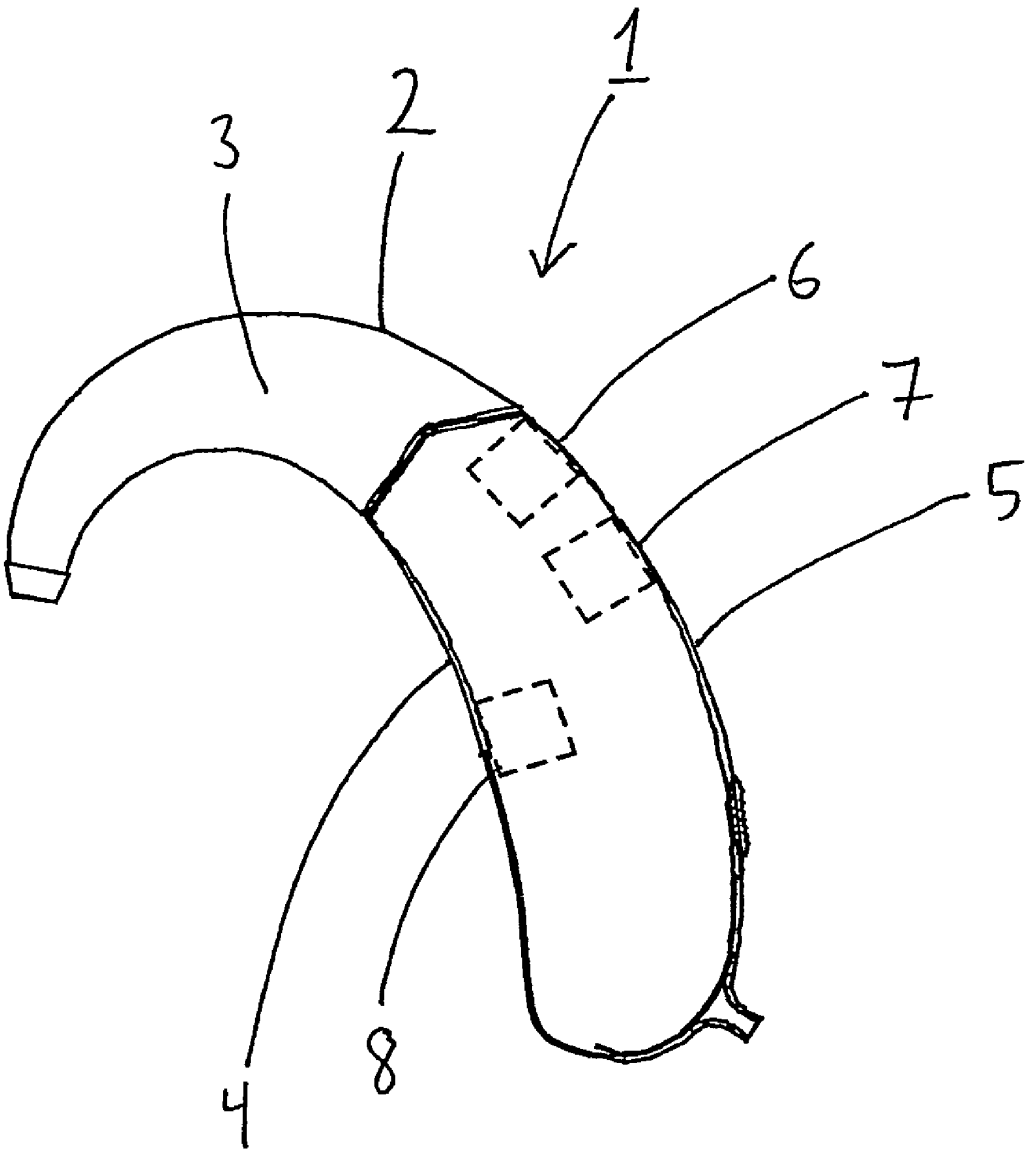


Fig. 1

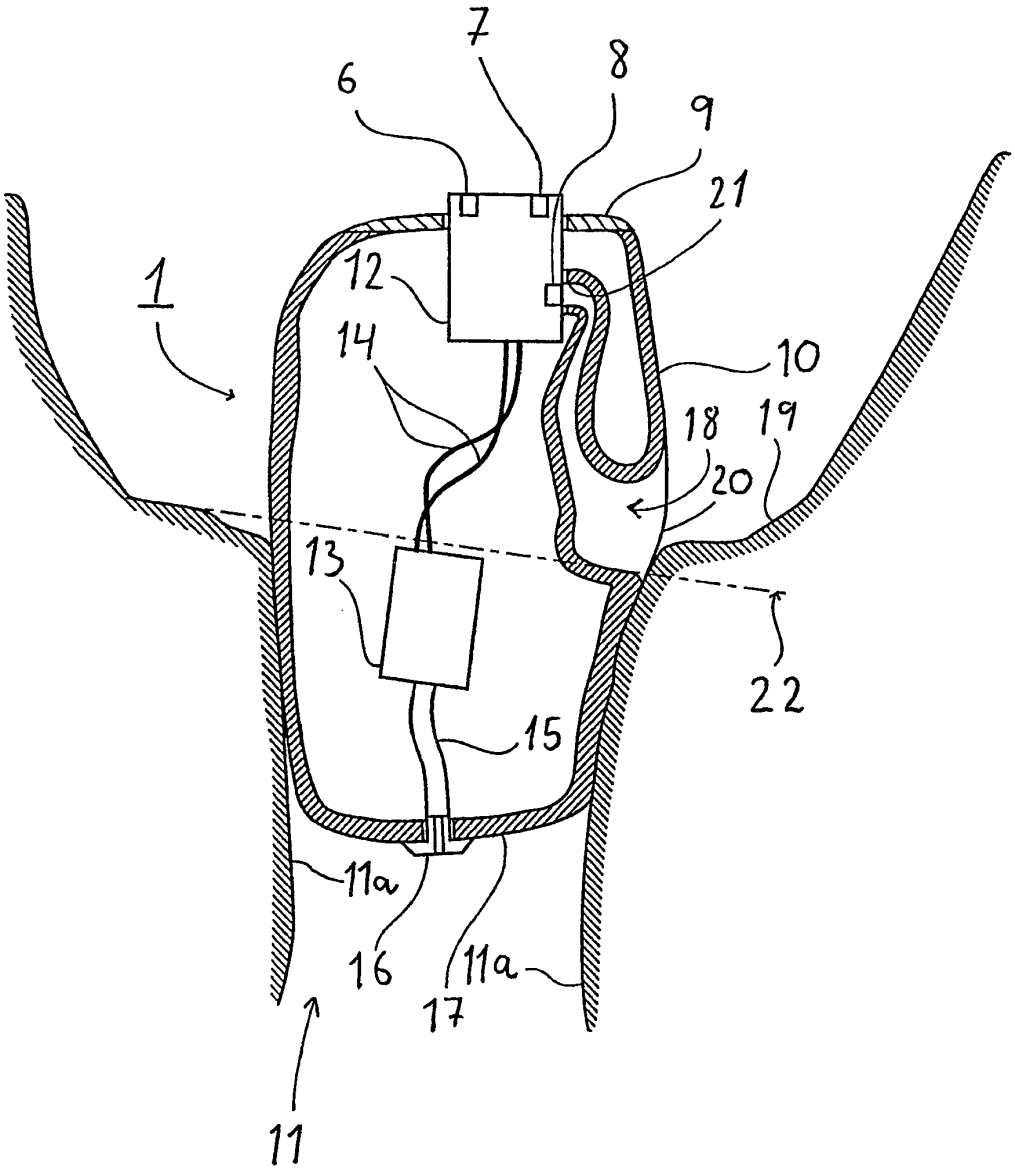


Fig. 2

HEARING AID AND A METHOD OF PROCESSING A SOUND SIGNAL IN A HEARING AID

RELATED APPLICATIONS

The present application is a continuation-in-part of application No. PCT/DK2003/000330, filed on May 19, 2003, in Denmark and published as WO2004/103020 A1.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention generally relates to hearing aids. The invention, more specifically relates to hearing aids and to methods of acquiring and processing sound signals in hearing aids. The invention, still more specifically, relates to a hearing aid
comprising a housing, and at least a first and a second microphone for receiving ambient sound.

2. The Prior Art

It is a known problem that transducers, such as microphones, in electronic devices give rise to problems with wind noise.

In hearing aids there are two prime sources of this wind noise. One is the direct influence of the wind on the membrane of the transducer. Another is acoustic noise from turbulence around the head, the outer ear, and the housing of the hearing aid itself.

In an ear with normal hearing the first source is not very pronounced because the membrane, i.e. the eardrum, is buried deep in the inner ear at the end of the ear canal.

This is not the case in hearing aids. Conversely, in hearing aids the transducers are, for various reasons such as frequency characteristics, sound pressure sensitivity, directional phase characteristics etc., mounted as far in the open as possible in the attempt to achieve an optimum acoustic response of the transducer.

In many cases the wind influence directly on the membrane results in so large signals from the transducer to the signal processing circuitry in the hearing aid that saturation occurs. When parts of the processing circuitry such as analogue/digital converters (ADCs) in a digital hearing aid, amplifier stages in an analogue hearing aid or even the microphones themselves saturate, the output signal produced by the hearing aid will be impaired.

Attempts have been made to reduce the problems with wind noises in hearing aids.

US-A-2002/0037088 refers to several prior art attempts, and states inter alia that in order to reduce wind noise it is known to fit the microphone openings so as to protect them from the wind as much as possible. This is however generally incompatible with the desire of having the microphones mounted as far in the open as possible. In particular, the direction sensitivity, which is achieved by using two or more microphones at predetermined locations on a hearing aid, would suffer if the microphones were not in the open. Moreover, US-A-2002/0037088, does not elaborate on how to protect these openings, as it takes another approach to overcoming wind noises than external measures.

U.S. Pat. No. 3,875,349 provides a hearing aid possessing a first microphone having spherical sensitivity characteristics and a second microphone with directional sensitivity characteristics, and a switch for switching the amplifier selectively to either one of both microphones. The switch may be automatically controlled depending on the signal. This document seeks to resolve different listening situations and does not deal with wind influences.

U.S. Pat. No. 5,201,006 shows a hearing aid with a primary and a secondary microphone.

The primary microphone serves to provide the signal to be amplified. The secondary microphone is provided for feedback compensation purposes and is adapted to receive sound through a secondary duct, said sound mainly originating from parts of the shell between the shaped portion and the wall of the auditory meatus. As the purpose is feedback compensation, the secondary microphones are not intended to receive any ambient sounds.

SUMMARY OF THE INVENTION

It is the object of the present invention to provide a hearing aid and a method of acquiring and processing sound, which overcomes the above problems.

According to the invention, in a first aspect, this object is achieved by a hearing aid comprising a housing, a first microphone for receiving ambient sound, a second microphone for receiving ambient sound, processing means for processing a signal from at least one of the first and the second microphone, and an output transducer, wherein said housing has a faceplate and a shell portion, the shell portion being adapted for partial insertion in the ear canal of a user, and wherein said second microphone has a sound port located at a surface of said shell portion at a location that is, during normal use of the hearing aid, sheltered against wind influences.

The second microphone disposed at a location, which, during normal use of the hearing aid, is sheltered against wind influences, can be used under circumstances, where the main microphone or microphones of the hearing aid are disturbed by wind influence.

The invention, in a second aspect, provides a hearing aid comprising a housing, a first microphone for receiving ambient sound, a second microphone for receiving ambient sound, processing means for processing a signal from at least one of the first and the second microphone, and an output transducer, wherein said housing has a concave surface adapted for wearing behind the ear of a user, and wherein said second microphone is disposed on said concave surface.

Having the microphone located in this position allows it to lie in the gap between the ear and the housing during use, whereby it is protected from the wind.

Having the microphone located within the housing also allows for a further preferred embodiment wherein the hearing aid comprises an electronics module and said second microphone is located on the electronics module. This has the advantage that no wires need to be drawn from the electronics module to the second microphone, thus obviating the need for the delicate work of drawing the wires, which is generally performed manually.

According to a further embodiment said second microphone comprises 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. The use of a canal provides further design options, such as special shapes for tuning the wind noise influence on the microphone.

Preferably, according to still a further embodiment, said canal has an inlet in said surface of the housing and said inlet is positioned so in said surface that it lies essentially in the transition between the concha and the ear canal, when said hearing aid is positioned in the ear for which the hearing aid is adapted. By having the inlet of the canal at this place, which

is as deep as practically possible in the ear, the ear itself aids in protecting the second microphone from the wind.

According to another preferred embodiment the lower cut-off frequency of said second microphone is substantially higher than the lower cut-off frequency of said at least one first microphone. This is advantageous as the majority of the energy in the noise spectrum from wind noises is found in and below the low frequency part of the audible spectrum. Saturation caused by the signal from this microphone will thus occur at a higher level than the saturation caused by the signal(s) from the main microphone(s). Saturation caused by these can thus be used to indicate the presence of wind noise.

Preferably the cut off-frequency of said second microphone is above approximately 1 kHz. The cut-off frequencies for the main microphones in modern hearing aids may be as low as 20 Hz. The sensitivity of the further microphone will thus, depending on the frequency of course, be several dB lower for frequencies below 1 kHz.

According to yet another embodiment the hearing aid comprises means for detecting saturation of the amplification circuitry and disabling said at least one first microphone in case saturation has been detected. Such detection could then be used to switch to the second microphone.

The invention, in a third aspect, provides a method of acquiring and processing a sound signal in a hearing aid, comprising acquiring a first sound signal through a first microphone opening placed in the hearing aid at a first position, which is selected for good acoustic rendering on conditions not exposed to wind, acquiring a second sound signal through a second microphone opening placed in the hearing aid at a second position, which is selected to be, during normal use, at least partially sheltered against direct wind exposure, detecting the conditions of wind exposure by analysis of at least one of the first and the second sound signals, selecting for processing on conditions not exposed to wind mainly the first sound signal, and selecting for processing on conditions exposed to wind mainly the second sound signal.

Hereby is achieved that good rendering is maintained under conditions where the first microphone is not subjected to wind influences.

According to an embodiment of the method, it comprises selecting for the first microphone a directional microphone.

Hereby is achieved that the directional sensitivity is maintained under conditions where the main microphone is not subjected to wind influences.

According to a second embodiment the method comprises selecting for the second microphone a microphone with less sensitivity to low frequencies than is the case with the first microphone.

Hereby is achieved that the hearing aid is even less prone to wind influences, because the second microphone or the circuit connected thereto will saturate later than the first microphone.

According to a different embodiment the method comprises customizing the hearing aid to adapt it to the auditory meatus of an individual user.

Hereby it is achieved that the second microphone is protected in the best possible way against wind influences.

According to another embodiment of the invention the method comprises flaring the second microphone opening to reduce any turbulence by crosswind.

The invention, in a fourth aspect, provides a method of acquiring and processing a sound signal in a hearing aid, comprising acquiring a first sound signal through a first microphone opening placed in the hearing aid at a first position, which is selected for good acoustic rendering on conditions not exposed to wind, acquiring a second sound signal through a second microphone opening placed in the hearing aid at a second position, which is selected to be, during normal use, recessed to a position essentially in the transition between the concha and the ear canal, in order to be at least partially sheltered against direct wind exposure, detecting the conditions of wind exposure by analysis of level of noise of at least one of the first and the second sound signals, selecting for processing on conditions not exposed to wind mainly the first sound signal, and selecting for processing on conditions exposed to wind mainly the second sound signal.

Further features and embodiments will appear from the detailed part of the specification.

BRIEF DESCRIPTION OF THE DRAWINGS

For better understanding of the invention a detailed description will be given based on non-limiting exemplary embodiments illustrated in the figures, where

FIG. **1** is a schematic side view, partially in section, illustrating typical positions of the microphones in a BTE hearing aid (BTE=behind-the-ear), and

FIG. **2** is a partial cross section schematically illustrating typical positions of the microphones in an ITE hearing aid (ITE=in-the-ear).

DETAILED DESCRIPTION OF THE INVENTION

In FIG. **1** a BTE hearing aid **1** is illustrated. The hearing aid comprises a housing **2**. The housing **2** essentially comprises four sides, viz. two substantially flat parallel sides **3** and two curved sides, where the curved sides comprise a convex side **5** and a concave side **4**. During use, the substantially flat sides **3** are located between the pinna and the head. The convex side **5** is exposed behind the ear, and the concave surface **4** faces the transition between the outer ear and the head.

A front microphone **6** and a rear microphone **7** are located on the convex surface **5**. In the following the front microphone **6** and the rear microphone **7** will generally be referred to as main microphones. In this respect it should be noted that hearing aids without direction sensitivity could have only one main microphone. The two main microphones **6** and **7** are spaced apart for obtaining direction sensitivity of the hearing aid **1**.

As already mentioned, the main microphones **6** and **7** are placed as far in the open as possible in order to obtain good characteristic. They are thus highly subjected to wind influences, and hence to wind-related noise.

A further microphone **8** is located on the concave surface **4** of the hearing aid. During use, i.e. when the hearing aid **1** is placed behind the ear of the wearer, a cavity is formed by the pinna, the head, the transition between these, and the concave surface **4** of the hearing aid housing. The further microphone **8** faces this cavity and is thus well protected from direct wind influences. In order to further suppress wind influences, the microphone opening of the second microphone **8** in the housing **2** may be flared (not shown).

Experiments have shown that even though the frequency characteristics of the further microphone **8** may be inferior to the frequency characteristics of the main microphones it is still acceptable, in particular when considering the alternative of a signal from the main microphones **6** and **7** that is influenced by wind noise or distorted, e.g. due to saturation.

FIG. **2** illustrates another embodiment of the invention. In the description of this embodiment the same reference numerals as in FIG. **1** have been used for the same or corresponding features.

The hearing aid in FIG. **2** has an ITE housing comprising an upper faceplate **9** and a hollow shell part **10** adapted for partial insertion into the ear canal **11** of the wearer where it is held against the ear canal wall **1** la. Typically the faceplate **9** is a standard item comprising an electronics module **12**, whereas the hollow shell part **10** is manufactured individually to the user. Methods for the manufacturing of custom shells, e.g. by casting, are well known in the field. In a preferred embodiment, the shell part is manufactured by means of Computer Aided Manufacturing using a 3D-printing method, in the following referred to as CAMISHA®. This method is described in WO-A-02/078233 and in U.S. Pat. No. 5,487,012.

The main microphones **6** and **7** are mounted in the electronics module **12**, which protrudes slightly from the faceplate **9**. Preferably, the electronics module **12** also contains all the necessary signal processing circuitry, except the output transducer **13**.

The output transducer **13** is located separately in the hearing aid housing and connected to the electronics module **12** by means of flexible leads **14**. The sound is emitted from the output transducer **13** through a transducer outlet tube **15** ending in a transducer outlet plug **16** in the bottom **17** wall of the hollow shell part **10** facing the inner part of the ear canal.

In the embodiment of FIG. **2** the electronics module **12** furthermore comprises a further microphone **8**. The further microphone **8** is located on a part of the electronics module **12** spaced from the faceplate and not facing the exterior. That is to say, the further microphone comprises an electronic part located on a part of the electronics module **12**, which, in the assembled hearing aid **1**, faces the cavity in the interior of the hearing aid **1**. In the illustrated embodiment the hearing aid housing comprises a canal **18** from inlet **20** in the surface of shell part **10** to the further microphone **8** within the housing. Hereby the effective sound port of the further microphone is located at the shell surface while the electronic part of the further microphone is located in the electronics module.

The canal **18** is preferably integrally formed with the shell part of the hearing aid housing. In a CAMISHA® manufacturing method, a computer model of the auditory canal is established. In an interactive optimising procedure, an operator selects the electronics module and decides the optimum location and orientation of the electronics module, as well as of the shell in the canal. The shell will then be printed tailored so as to contact parts of the sides of the auditory canal. However, an outer part of the shell will not contact the auditory canal. It is a part of the customizing to decide which part will contact the canal and to establish the demarcation line to the outer shell part.

Because the CAMISHA® shell part **10** of the housing of the hearing aid **1** is individually manufactured, it is possible to locate the inlet **20** of the canal **18** just outside the demarcation line **22** in such a way in the shell part **10** of the housing that it is just exposed, when the hearing aid is inserted to the wearing position in the ear canal **11**. Thereby it is achieved that the inlet **20** of the canal **18** is located as deeply as possible in the ear without the ear canal wall **1** la blocking the inlet **20**. The demarcation line, demarcating the contacting portion form the non-contacting shell portion is illustrated with the dash-dot line **22** in FIG. **2**. The recessed location of the inlet **20** achieves sheltering against direct wind exposure by the housing in combination with the head and the pinna of the hearing aid user. Preferably the inlet **20** forming the outer microphone opening is flared, as can be seen in FIG. **2**.

The other end of the canal **18** ends in an outlet **21** at a position in the hearing aid corresponding to the position of the further microphone **8**. Because the further microphone **8** is located on the electronics module **12** it has a well-defined position with respect to the faceplate **9**. This allows the canal **18** to be integrally formed with the CAMISHA® shell part **10** during the individualized computer aided manufacturing.

Thus, by the use of the CAMISHA® shell part **10** it is possible to locate the canal **18** optimally with respect to the positions of its inlet **20** and outlet **21**. That is to say, to have the inlet **20** retracted or recessed as deeply as possible in the ear, where least wind influences are present, and to have the outlet **21** arranged to feed the sound into the further microphone **8**, which is placed on the electronics module **12**, obviating the need for further wires to be drawn. In the preferred embodiment, the inlet **20** to the microphone is flared. Flaring the inlet reduces the noise input into the canal by any crosswind.

Though the present invention has in the above been embodied in ITE and BTE hearing aids only, the skilled person would realise that it would also in a CIC hearing aid (CIC=completely in the canal) be possible to arrange a further microphone within the housing with a canal leading to it.

Evidently, only little useful directional information can be obtained from the retracted microphone **8**. Switching directly to the further microphone **8** alone, when wind noise or saturation is detected, would therefore entail the sacrifice of directional information. However, most of the noise energy from the wind-generated noises is low frequent. Accordingly, in one embodiment high pass filtering is used to extract high frequency information from the main microphones **6** and **7** and low pass filtering to extract the low frequency information without wind noise from the further microphone **8**. This may be achieved in several ways e.g. by introduction of fixed high and low pass filters upon detection of saturation, or by smooth switching. This then allows suppression of wind noise from the main microphones **6** and **7**, while maintaining at least some degree of directional sensitivity. In other embodiments, the signals from the respective microphones can be combined or balanced according to a weighting factor, selected according to the prevailing conditions.

Depending on the type of hearing aid, various criteria for detection of wind noise exist. One is low frequency comparison between the signals from the further microphone **8** and the main microphones **6** and **7**. Another is detection of saturation of the analogue/digital converter. Typically in wind noise conditions the slew rate could be detected, as the signal would erratically switch from full positive signal to full negative signal and vice versa. Still another criterion could be the detection of the low-frequency part of the signal from the main microphone exceeding a predetermined threshold. Yet another criterion could be the level of uncorrelated noise in the dual microphone parts of a directional microphone exceeding a predetermined threshold.

Since most of the signal energy of the wind noise is low frequent it is advantageous to have a different cut-off frequency for the further microphone, as compared to the main microphones **6** and **7**, which themselves preferably are matched to each other. This allows the further microphone to saturate much later than the main microphones, even if they were to be exposed to the same noise.

We claim:

1. A hearing aid comprising a housing, a first microphone for receiving ambient sound, a second microphone for receiving ambient sound, processing means for processing a signal from at least one of the first and the second microphone, and an output transducer, wherein said housing has a faceplate and a shell portion, the shell portion being adapted for partial insertion in the ear canal of a user, and wherein said hearing aid is customized to adapt it to the auditory meatus of said user such that said second microphone has a sound port

located at a surface of said shell portion at a location that is, during normal use of the hearing aid, sheltered against wind influences, wherein said port is positioned essentially at the transition between the concha and the ear canal, when said hearing aid is positioned in the intended normal use position.

2. The hearing aid according to claim **1**, wherein said processing means is adapted for detecting conditions of wind exposure by analysis of a sound signal from at least one of the first and the second microphones, for selecting for processing on conditions not exposed to wind mainly the signal from the first microphone, and for selecting for processing on conditions exposed to wind mainly the signal from the second microphone.

3. The hearing aid according to claim **1**, wherein said processing means comprises an electronics module, and wherein said second microphone is located on said electronics module.

4. The hearing aid according to claim **1**, wherein said second microphone comprises an electronic part and a canal from said sound port to said electronic part.

5. The hearing aid according to claim **1**, wherein the lower cut-off frequency of said second microphone is substantially higher than the lower cut-off frequency of said first microphone.

6. The hearing aid according to claim **1**, wherein the cut off-frequency of said second microphone is above approximately 1 kHz.

7. The hearing aid according to claim **2**, wherein the processing means comprises amplification circuitry, and wherein the processing means is adapted for detecting conditions of wind exposure by detecting saturation of the amplification circuitry.

8. A behind-the-ear type hearing aid comprising a housing with curved sides including a convex side and a concave side, at least a first microphone for receiving ambient sound, a second microphone for receiving ambient sound, processing means for processing a signal from at least one of the first and the second microphone, and an output transducer, wherein

a first microphone opening for said first microphone is formed on the convex side of the hearing aid housing, and

a second microphone opening for said second microphone is disposed on said concave side of said hearing aid housing so as to shelter against wind influences by the housing in combination with the head and the pinna of the hearing aid user.

9. A method of acquiring and processing a sound signal in a hearing aid having a housing with curved sides including a convex side and a concave side, the method comprising acquiring a first sound signal through a first microphone opening formed on the convex side of the hearing aid housing, acquiring a second sound signal through a second microphone opening formed on the concave side of said housing so as to be sheltered against wind influences by the housing in combination with the head and pinna of the hearing aid user, detecting the conditions of wind exposure by analysis of at least one of the first and the second sound signals, said analysis comprising detection of the low frequency part of the signal from the first microphone exceeding a predetermined threshold, selecting for processing on conditions not exposed to wind mainly the first sound signal, and selecting for processing on conditions exposed to wind mainly the second sound signal.

10. The method according to claim **9**, comprising acquiring the first sound signal by a directional microphone.

11. The method according to claim **9**, comprising acquiring the second sound signal by a microphone with less sensitivity to low frequencies than is the case with the first microphone.

12. The method according to claim **9**, comprising customizing the hearing aid to adapt it to the auditory meatus of an individual user.

13. The method according to claim **9**, comprising flaring the second microphone port to reduce any turbulence by cross-wind.

14. The method according to claim **9**, wherein said analysis further comprises low frequency comparison between the first and second sound signals.

15. The method according to claim **9**, wherein said analysis further comprises detection of saturation by the signal of an analogue/digital converter.

16. The method according to claim **9**, wherein said analysis further comprises detection of the slew rate of at least one of the first and the second sound signals.

17. A method of acquiring and processing a sound signal in a hearing aid comprising a housing having a faceplate and also having a shell portion adapted for partial insertion in the ear canal of the hearing aid user, said method comprising:

acquiring a first sound signal by a directional microphone through a first microphone opening placed in the hearing aid at a first position, which is selected for good acoustic rendering on conditions not exposed to wind,

acquiring a second sound signal through a second microphone opening placed in the hearing aid at a second position, wherein said hearing aid is customized to adapt it to the auditory meatus of said user such that said second position is on a surface of said shell portion at a location that is, during normal use of the hearing aid, sheltered against wind influences, wherein said second position is essentially at the transition between the concha and the ear canal, when said hearing aid is positioned in the intended normal use position,

detecting the conditions of wind exposure by analysis of at least one of the first and the second sound signals,

selecting for processing on conditions not exposed to wind mainly the first sound signal, and

selecting for processing on conditions exposed to wind mainly the second sound signal,

wherein the first sound signal comprises a dual signal from two parts of said directional microphone, and

wherein the analysis of at least one of the first and the second sound signals comprises detection of the level of uncorrelated noise in the dual signal exceeding a predetermined threshold.

18. A method of acquiring and processing a sound signal in a hearing aid comprising a housing having a faceplate and also having a shell portion adapted for partial insertion in the ear canal of the hearing aid user, said method comprising:

acquiring a first sound signal through a first microphone opening placed in the hearing aid at a first position, which is selected for good acoustic rendering on conditions not exposed to wind,

acquiring a second sound signal through a second microphone opening placed in the hearing aid at a second position, wherein said hearing aid is customized to adapt it to the auditory meatus of said user such that said second position is on a surface of said shell portion at a location that is, during normal use of the hearing aid, sheltered against wind influences, wherein said second position is essentially at the transition between the concha and the ear canal, when said hearing aid is positioned in an intended normal use position,

detecting the conditions of wind exposure by analysis of a level of noise of at least one of the first and the second sound signals,

selecting for processing on conditions not exposed to wind mainly the first sound signal, and

selecting for processing on conditions exposed to wind mainly the second sound signal,

wherein said analysis of the level of noise of at least one of the first and second sound signals comprises at least one of a low frequency comparison between the signals from the first microphone and the second microphone, and a slew rate detection.

19. The hearing aid according to claim **1**, wherein the hearing aid comprises an electronics module and said second microphone is located on said electronics module.

20. The method according to claim **9** wherein the hearing aid comprises an electronics module and said second microphone is located on the electronics module.

21. A hearing aid according to claim **1**, wherein said second microphone comprises a port in the surface of the housing and a canal from the port to an electronic part of the microphone within that housing.

22. A method according to claim **9**, wherein said second microphone comprises a port in the surface of the housing and a canal from the port to an electronic part of the microphone within that housing.

23. A method of fabricating a hearing aid having a housing, a first microphone for receiving ambient sound, a second microphone for receiving ambient sound, processing means for processing a signal from at least one of the first and the second microphone, and an output transducer, wherein said housing has a faceplate and a shell portion, the shell portion being adapted for partial insertion in the ear canal of a user, said method comprising customizing said hearing aid to adapt it to the auditory meatus of said user such that said second microphone has a sound port located at a surface of said shell portion at a location that is, during normal use of the hearing aid, sheltered against wind influences, so that said port is positioned essentially at the transition between the concha and the ear canal, when said hearing aid is positioned in the intended normal use position.

* * * * *

Exhibit PA-T

US 20060083388A1

(19) **United States**

(12) **Patent Application Publication** (10) **Pub. No.: US 2006/0083388 A1**

Rothschild (43) **Pub. Date: Apr. 20, 2006**

(54) **SYSTEM AND METHOD FOR SELECTIVELY SWITCHING BETWEEN A PLURALITY OF AUDIO CHANNELS**

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(57) **ABSTRACT**

A system and method for selectively switching between a plurality of audio channels are provided. The system and method of the present disclosure will enable a user to listen to desired audio content, e.g., music with a total lack of distraction while selectively allowing certain important and selected sounds to interrupt the audio content. The system for selectively switching between a plurality of audio channels includes an audio content device (ACD) for supplying audio content on a first channel; at least one microphone input device for generating an acoustic signal from sound external to the system on a second channel; an audio signal processing control unit (ASPCU) for receiving the audio content on the first channel and the acoustic signal on the second channel and for selectively switching between the first and second channel; and an audio output device (AOD) for audibly producing sounds from the selected channel.

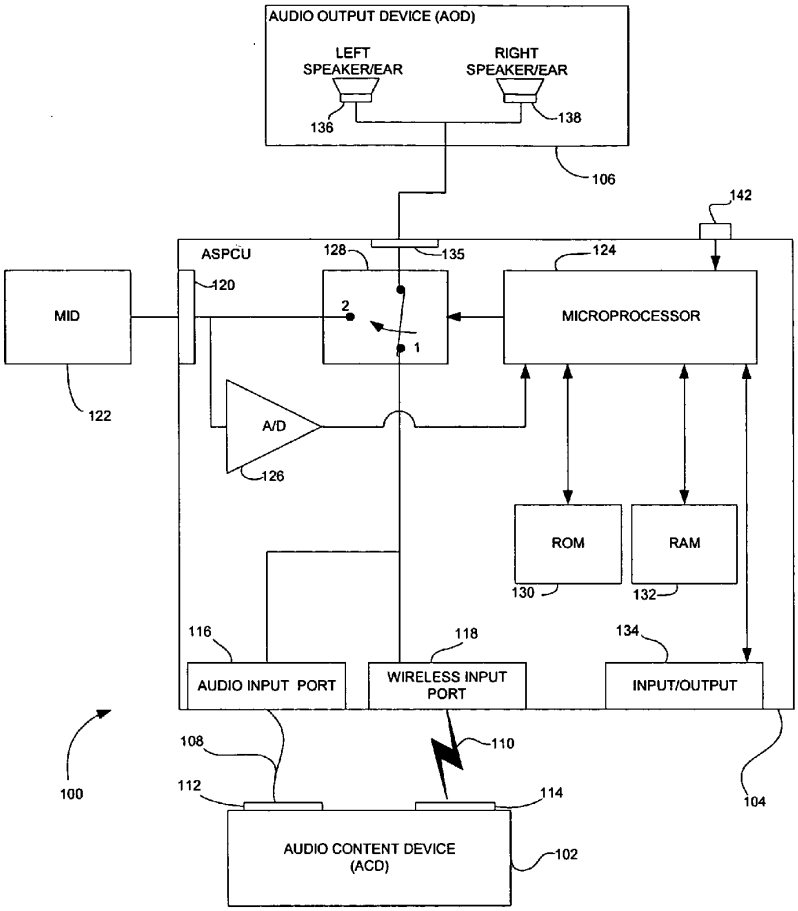


Exhibit PA-T

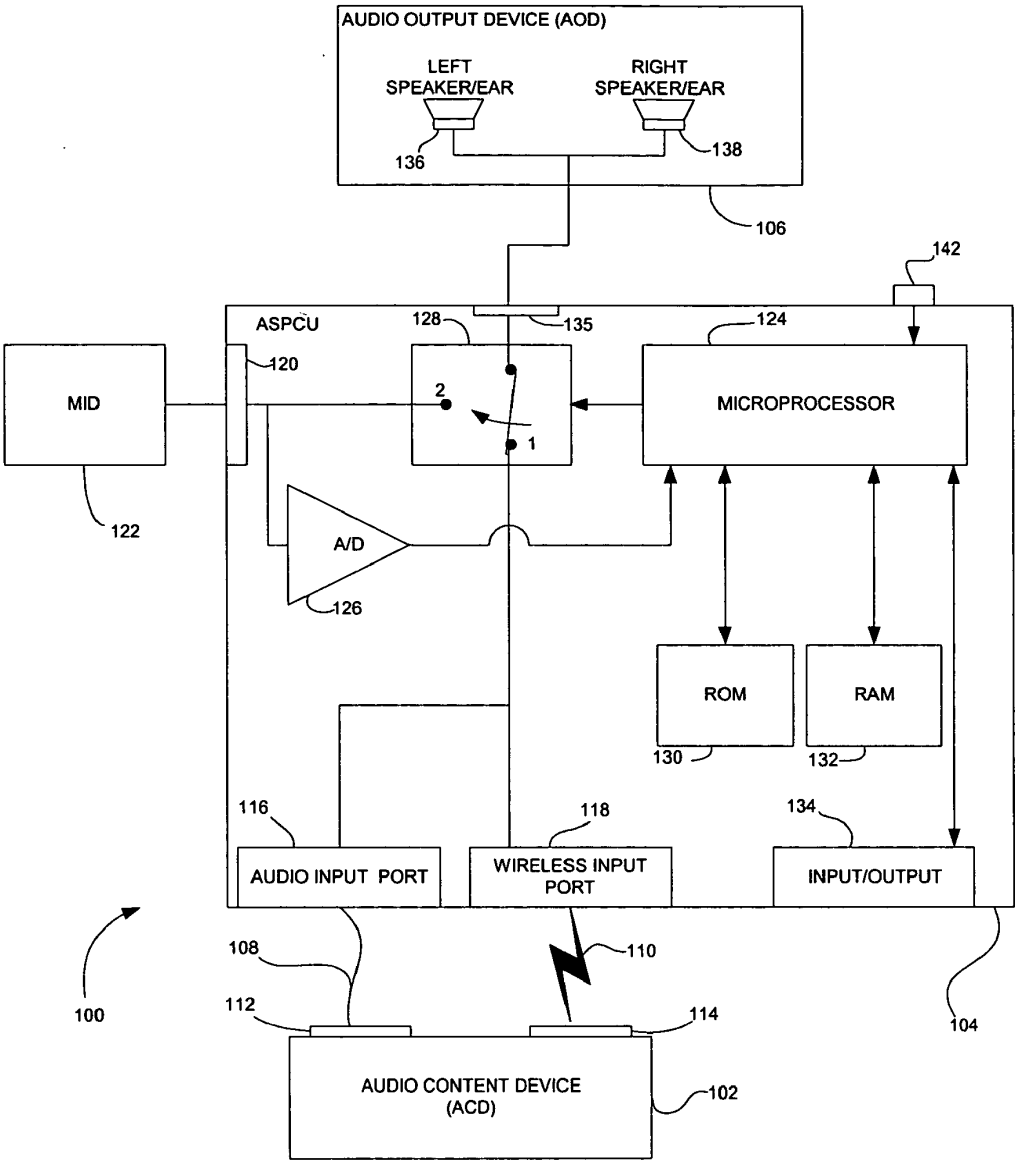


FIG. 1

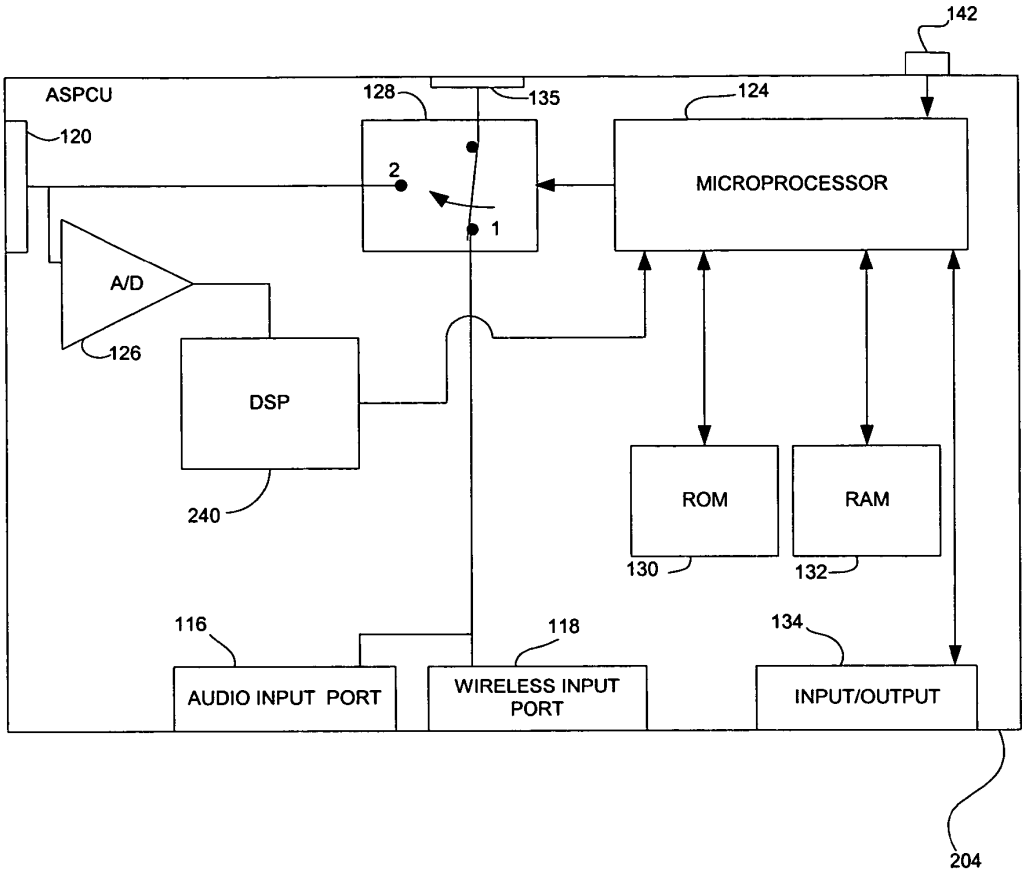

142
ASPCU
120
128
135
124
2
1
MICROPROCESSOR
A/D
126
DSP
240
ROM
RAM
130
132
116
118
134
AUDIO INPUT PORT
WIRELESS INPUT PORT
INPUT/OUTPUT
204

FIG. 2

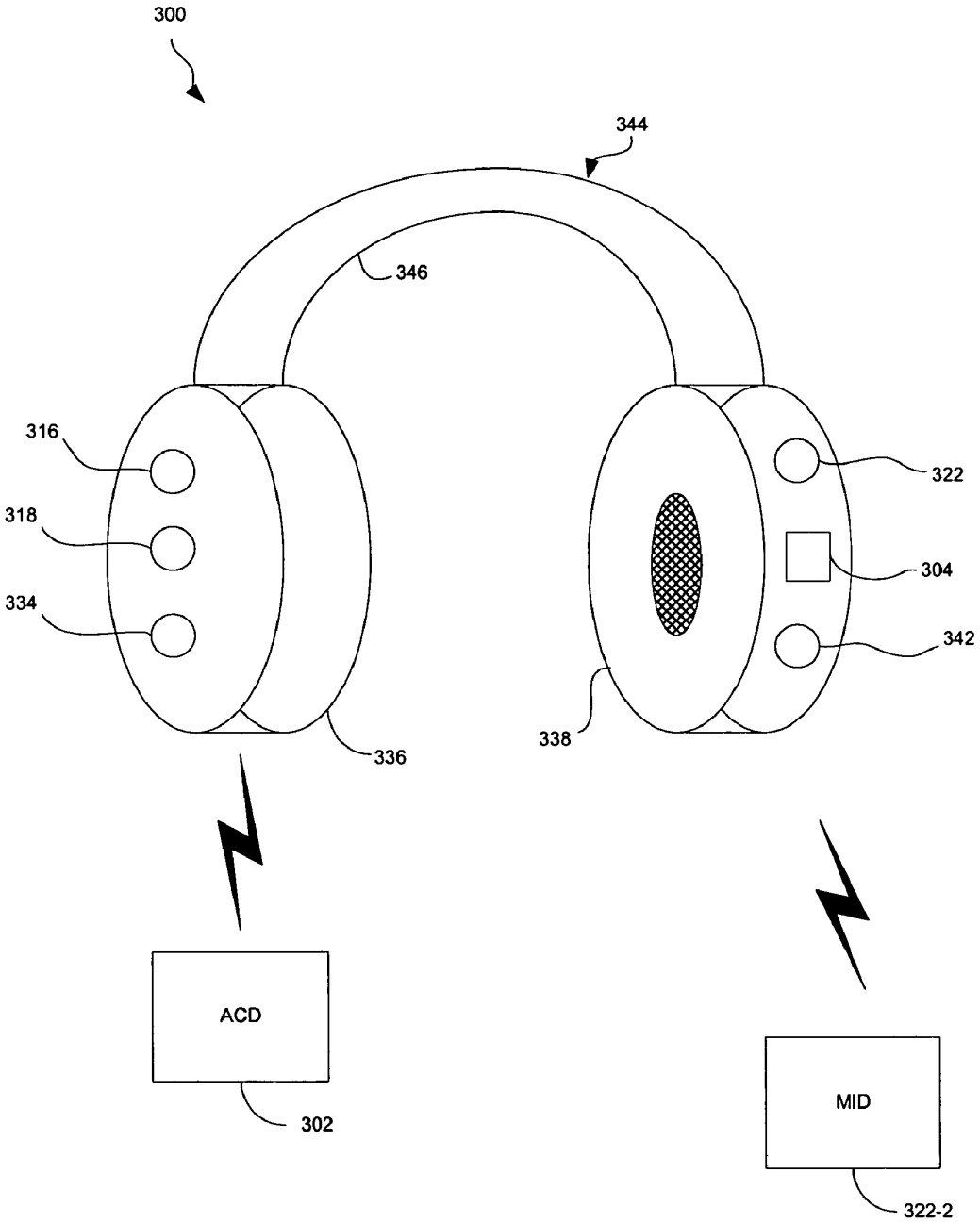


FIG. 3

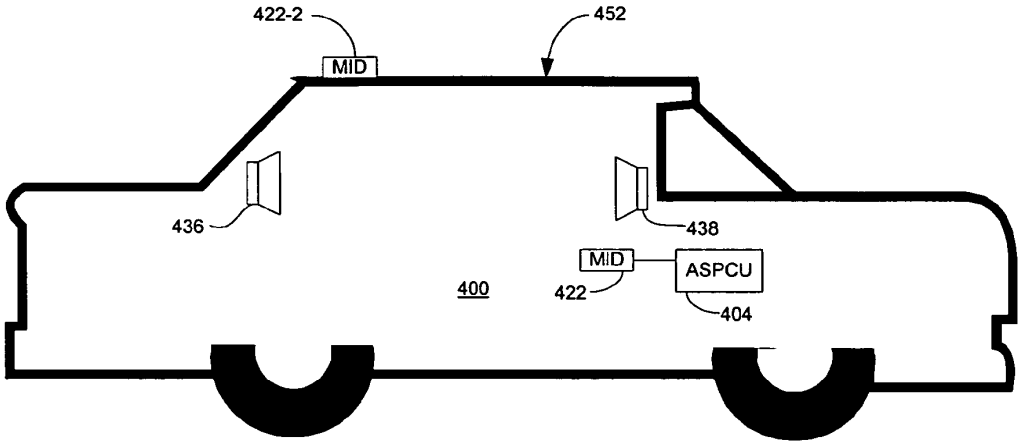


FIG. 4

SYSTEM AND METHOD FOR SELECTIVELY SWITCHING BETWEEN A PLURALITY OF AUDIO CHANNELS

BACKGROUND

[0001] 1. Field

[0002] The present disclosure relates generally to data processing and audio communications systems, and more particularly, to systems and methods for selectively switching between a plurality of audio channels.

[0003] 2. Description of the Related Art

[0004] Audio and video content devices have become more numerous in the past several years with device proliferation and miniaturization. Content devices (both audio and video) have become increasingly portable, and various devices including portable DVD players, cellular telephones, portable radios and televisions, MP3 audio players, network audio players, CD players, portable computers, tape cassette players, PDAs, minidisk players, among others, are now commonplace. The users of these devices take these devices into a variety of environments and use the devices for both business and pleasure. The device users frequently have a desire to enjoy the audio content in virtual exclusion of sounds other then the desired audio content.

[0005] Recently, digital noise reduction sound processing, and better headphone designs to isolate sound, have allowed audio headphone users to enhance their audio content listening experience. However, increasingly, the audio headphone devices are used in environments where various noise channels outside the contained headphone environment exist simultaneously. For instance, in an automobile, one source of sound would be the sound coming from the audio device (in this example music content), while another source would be the sounds from outside the car, while a third would be the sounds from within the car. The user frequently has a desire and a need to be able to process all of these distinct sound channels, but the challenge is in processing only the channels that the user has the need to listen to at the specific time the user needs to listen to them. 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.

[0006] Therefore, a need exists for systems and methods for selectively switching between various audio channels, for example, between desired audio content and predetermined sound patterns. A need also exists for a system which switches between a plurality of audio channels where one of the audio channels is remote from the user.

SUMMARY

[0007] A system and method for selectively switching between a plurality of audio channels with or without user input are provided. The system will receive and audibly produce desired audio content to a user, but will interrupt the audio content when predetermined sound patterns are detected and subsequently play the sound patterns to the user. The system and method of the present disclosure will allow the user to hear external noises (e.g., horns, emergency vehicles, people, etc) outside the user's listening environment, when needed by means of selective switched sound processing. The selective sound processing will selectively allow certain sounds or voices to immediately interrupt and override the audio content, e.g., music. The resulting experience with the system and method of the present disclosure is one where the user can enjoy audio content with a total lack of distraction, until the system and method selectively allows certain important and selected sounds to interrupt the audio content.

[0008] According to an aspect of the present disclosure, an apparatus for selectively switching between audio channels is provided. The apparatus includes a first audio input connection for receiving audio content; a second audio input connection for receiving an acoustic signal; a controller for receiving the audio content and acoustic signal and for determining whether to output the audio content or acoustic signal; and an output connection for outputting the determined signal. The apparatus further includes a switching mechanism having a first and second position controlled by the controller, wherein in the first position the first audio input connection is coupled to the output connection and in the second position the second audio input connection is coupled to the output connection.

[0009] According to another aspect of the present disclosure, a system for selectively switching between a plurality of audio channels is provided. The system includes an audio content device (ACD) for supplying audio content on a first channel; at least one microphone input device for generating an acoustic signal from sound external to the system on a second channel; an audio signal processing control unit (ASPCU) for receiving the audio content on the first channel and the acoustic signal on the second channel and for selectively switching between the first and second channel; and an audio output device (AOD) for audibly producing sounds from the selected channel.

[0010] According to a further aspect of the present disclosure, a headphone for selectively switching between a plurality of audio channels is provided. The headphone includes an audio content device (ACD) for supplying audio content on a first channel; at least one microphone input device for generating an acoustic signal from sound external to the headphone on a second channel; an audio signal processing control unit (ASPCU) for receiving the audio content on the first channel and the acoustic signal on the second channel and for selectively switching between the first and second channel; and first and second speakers for audibly producing sounds from the selected channel.

[0011] In yet another aspect of the present invention, a method for selectively switching between a plurality of audio channels in an audio device is provided. The method includes the steps of supplying audio content on a first channel of the audio device; generating an acoustic signal from sound external to the audio device on a second channel; receiving the audio content on the first channel and the acoustic signal on the second channel and selectively

switching between the first and second channel; and audibly producing sounds from the selected channel. The method further includes the steps of determining if the acoustic signal matches a predetermined pattern; and if the acoustic signal matches the predetermined pattern, selecting the second channel to be audibly produced, wherein the predetermined pattern is digitized human speech or digitized emergency sounds.

BRIEF DESCRIPTION OF THE DRAWINGS

[0012] The above and other aspects, features, and advantages of the present disclosure will become more apparent in light of the following detailed description when taken in conjunction with the accompanying drawings in which:

[0013] **FIG. 1** is a diagram of a system for selectively switching between a plurality of audio channels in accordance with an embodiment of the present disclosure;

[0014] **FIG. 2** is diagram of an audio signal processing control unit (ASPCU) for selectively switching between a plurality of audio channels in accordance with an embodiment of the present disclosure;

[0015] **FIG. 3** is a diagram of an exemplary headphone employing a system for selectively switching between a plurality of audio channels in accordance with an embodiment of the present disclosure; and

[0016] **FIG. 4** is a diagram of a vehicle employing a system for selectively switching between a plurality of audio channels in accordance with an embodiment of the present disclosure.

DETAILED DESCRIPTION

[0017] Preferred embodiments of the present disclosure will be described hereinbelow with reference to the accompanying drawings. In the following description, well-known functions or constructions are not described in detail to avoid obscuring the present disclosure in unnecessary detail. Throughout the figures like reference numerals represent like elements.

[0018] A system and method for selectively switching between a plurality of audio channels are provided. The system and method of the present disclosure will enable a user to listen to desired audio content, e.g., music with a total lack of distraction while selectively allowing certain important and selected sounds to interrupt the audio content.

[0019] Referring to **FIG. 1**, a system for selectively switching between a plurality of audio channels is shown. The system **100** generally includes an audio content device (ACD) **102** for supplying audio content on a first channel, an audio signal processing control unit (ASPCU) **104** for selectively switching between the audio content supplied from the ACD **102** and other audio sources, e.g., a microphone input device **122**, on a second channel and an audio output device (AOD) **106** for audibly producing sounds from the selected channel.

[0020] The ACD **102** may be any device that produces and delivers an audio signal to the ASPCU **104**. Conventional audio content devices include but are not limited to portable DVD players, cellular or mobile telephones, portable radios and televisions, MP3 audio players, network audio players, CD players, portable computers, tape cassette players, personal digital assistants (PDAs), minidisk players, among others. It should be noted that some ACD devices are analog signal devices, while other are based on digital signal processing.

[0021] The audio content device (ACD) **102** may be coupled to the ASPCU **104** via hardwired **108** or wireless connection **110**. If a hardwired connection is employed, the ACD **102** will include the appropriate output connection **112**, e.g., an RCA jack, a USB port, a FireWire port (IEEE 1394), serial port, parallel port, etc. If a wireless connection **110** is employed, the ACD **102** will include a wireless port **114** with an appropriate encoder and transmitter to wirelessly transmit audio content to the ASPCU **104**. The wireless connection will operate under any of the various known wireless protocols including but not limited to Bluetooth™ interconnectivity, infrared connectivity, radio transmission connectivity including computer digital signal broadcasting and reception commonly referred to as Wi-X or 80211.X (where x denotes the type of transmission), or any other type of communication protocols or systems currently existing or to be developed for wirelessly transmitting data.

[0022] To receive the audio content from the ACD **102**, the ASPCU **104** will include at least one audio input port, e.g., an audio input port **116** for hardwired connections and/or a wireless input port **118** for wireless connections. It is to be appreciated that if a wireless connection is employed, the wireless input port **118** of the ASPCU **104** will include conventional circuitry to process the incoming audio content, e.g., a receiver, decoder, demodulator, etc. It is also to be appreciated that depending on the type of ACD **102** used, the input ports **116**, **118** of the ASPCU **104** may include further circuitry, e.g., analog-to-digital converters (ADC), digital-to-analog converters (DAC), for converting the incoming signals to an appropriate format to be either processed and/or audibly produced for a user.

[0023] The ASPCU **104** is adapted to received and process sounds and/or acoustic signals other then the desired audio content from the ACD **102**. The ASCPCU **104** will listen for predetermined sounds and, if necessary, interrupt the audio content being received from the ACD **102** and play the externally generated sound to the user. The ASPCU **104** includes a second audio input port **120** adapted to receive sounds and/or acoustic signals generated externally from system **100**. Preferably, a microphone input device (MID) **122** will be coupled to the second audio port **120** for receiving sound and generating an acoustic signal to the ASPCU **104**. The MID **122** may be coupled to the ASPCU **104** by the various hardwired and wireless connections described above. If a wireless MID **122** is employed, the wireless MID will include an encoder/modulator for generating an electrical acoustic signal from sound and a transmitter/antenna combination to transmit the acoustic signal to the ASPCU **104**. Correspondingly, in the wireless embodiment, the second audio input port **120** will include a receiver and decoder for receiving and decoding the transmitted signal.

[0024] The ASPCU **104** will include a microprocessor **124** for receiving the acoustic signal from the MID **122** and for determining whether the acoustic signal should interrupt the audio content being played to allow the user to hear the acoustic signal. Preferably, an output of the MID **122** will be coupled to the ASPCU **104** by an analog-to-digital converter

126 for converting the acoustic signal generated by the MID into a digital form that can be processed by the microprocessor **124**. The ASPCU **104** will further include a switching mechanism **128** having at least a first input coupled to the audio input port **116**, **118**, a second input coupled to the MID **122** and an output coupled to the AOD **106**. The switching mechanism **128**, e.g., a relay, transistor, etc, is controlled by the microprocessor **124** to allow the audio content from the ACD **102**, e.g., a first channel, or the acoustic signal from the MID **122**, e.g., a second channel, to be played to the user. Initially, upon starting of the ASPCU **104**, the switching mechanism **128** will default to the first position to allow any audio content received by the audio input port **116**, **118** to be transmitted to the AOD **106** to be played to the user, via an audio output port **135**. During use, the microprocessor **124** will continuously monitor acoustic signals coming from the MID **122**. If any acoustic signal matches a predetermined digital pattern, the microprocessor **124** will transmit an output signal to the switching mechanism **128** to set the switching mechanism **128** to the second position. In the second position of the switching mechanism **128**, the acoustic signal picked up by the MID **122** will be output to the AOD **106** and played to the user.

[**0025**] The audio output device (AOD) **106** may be any device known in the art to audibly produce sound from electrical signals, for example, a speaker, headphones, an ear bud, etc. Preferably, the AOD **106** will include left **136** and right **138** speakers/ear devices for individual playing separate channels of audio content to produce stereo sound and/or for individual playing sound from the input channels of the ASPCU **104**. The AOD **106** may further include an amplifier (not shown) for amplifying the signal to be played, or alternatively, the amplifier may be disposed in the ASPCU **104**. The AOD **106** may receive the signals to be played wirelessly as described above, and in this embodiment, the output port **135** will have the necessary wireless components.

[**0026**] The ASPCU **104** may also include conventional digital noise reduction processing circuitry that will allow the ASPCU to process and reduce noise from both the ACD **102** and the MID **122**.

[**0027**] As described above, the microprocessor **124** will be constantly monitoring the ACD **102** and the MID inputs for certain digital patterns, e.g., preselected sound patterns, that have been preset into the microprocessor's processing instructions. These executable instructions will be loaded into the microprocessor during an initialization routine from random access memory (ROM) **103**. These digital patterns will represent audio sounds that have been digitized. This presetting to recognize certain audio sounds could be programmed from inception by the manufacturer of the system of the present disclosure, or alternatively, could be programmed by the user of the system who would program the ASPCU by means of a computer or other programming device coupled to the ASPCU via input/output port **134**. These user-defined patterns will be stored in random access memory (RAM) **132**, e.g., internal flash memory, compact flash cards, smartmedia cards, memory stick, a microdrive, etc. These present sounds could include but are not limited to various human voice patterns denoting various words, various human voice patterns denoting stress or emergency, various noise patterns denoting emergency sounds including police cars, ambulances, fire engines, or other sound patterns that the manufacturer, or alternatively, the user wants the ASPCU to recognize.

[**0028**] Conventional computer software programs exist that allow human speech patterns to be converted from a MID or other comparable device, to digital signals (or digital code) and then to allow those signals (code) to be recognized as human words, also known as speech recognition technology. However, since speech recognition technology requires a large amount of processing power, the system's reaction time to external sounds may be slower than required to be effective. Referring to **FIG. 2**, another embodiment of the ASPCU **204** is illustrated which increases the processing speed of the ASPCU. The ASPCU **204** includes a digital signal processor (DSP) **240** which is functionally similar to a microprocessor but performs one function. Here, the DSP **240** includes a speech recognition algorithm for receiving an acoustic signal from the MID and A/D converter **126** and for determining whether it matches a preset pattern. If the DSP **240** determines a match has occurred, the DSP **240** will transmit a signal to the microprocessor **124** or to the switching mechanism **128** directly. By moving the speech recognition functionality to the DSP **240**, the DSP **240** will react quicker than the microprocessor **124** since this is its only function and the microprocessor **124** will be less taxed in performing other functions of the system. Therefore, the overall system response time will be quicker.

[**0029**] It is to be understood that the present disclosure may be implemented in various forms of hardware, software, firmware, special purpose processors, or a combination thereof. A system bus couples the various components shown in **FIGS. 1 and 2** and may be any of several types of bus structures including a memory bus or memory controller, a peripheral bus, and a local bus using any of a variety of bus architectures. The system also includes an operating system and micro instruction code. The various processes and functions described herein may either be part of the micro instruction code or part of an application program (or a combination thereof) which is executed via the operating system.

[**0030**] It is to be further understood that because some of the constituent system components and method steps depicted in the accompanying figures may be implemented in software, the actual connections between the system components (or the process steps) may differ depending upon the manner in which the present disclosure is programmed. Given the teachings of the present disclosure provided herein, one of ordinary skill in the related art will be able to contemplate these and similar implementations or configurations of the present disclosure.

[**0031**] When the user utilizes the system of the present disclosure, the user would normally be hearing only sound from the ACD **102** which would be delivered to the user via the AOD **106**, e.g., a speaker or headphone. The ACD content would continue to be heard by the user until ASPCU **104**, **204** recognizes one of the sound patterns that would come from the MID **122**, and the microprocessor **124** of the ASPCU would instruct the switching mechanism **128** to stop the digital output, or alternatively the analog output, of the ACD and quickly switch the user to the MID output. The ASPCU **104**, **204** will allow the user to set a time delay for the switch over from the ACD to the MID. After the switch

occurs between the ACD **102** and MID **122**, the ASPCU will switch back to the audio content from the ACD after either a time delay preprogrammed by the user or after the user manually instructed the ASPCU to switch again to the ACD. This manual instruction could be communicated by speech recognition which would allow the MID to signal the ASPCU and thus the ASPCU to switch back to the ACD or by means of a switch **142** that the user could press located on the AOD **106** or the ASPCU, or any other means that would instruct the ASPCU to make the switch between the MID and the ACD.

[**0032**] In another embodiment, instead of completely switching from the ACD **102** to the MID **122**, the ASPCU **104**, **204** may lower the volume of the audio content coming from the ACD and play the sounds from the MID at a higher volume. Alternatively, the ASPCU may supply the audio content from the ACD **102** to one output channel, e.g., left speaker **136**, and the sound from the MID **122** to the second output channel, e.g., right speaker **138** so the user may simultaneously hear both channels.

[**0033**] In a further embodiment, upon the system **100** switching to the input from the MID **122** or other external sound, the ASPCU **104** may instruct the audio content device **102** to pause from supplying the audio content, or if the source content is live, e.g., radio transmission, satellite transmission, television transmission, etc., the ASPCU **104** may buffer the received audio content in conventional memory buffers or RAM **132**. When the system switches back to the audio content device or first channel, the ASPCU will either instruct the audio content device to unpause and resume supplying audio content from the point of interruption or, alternatively, will play the audio content stored in the memory buffer.

[**0034**] Referring to **FIG. 3**, another embodiment of the present disclosure is illustrated. The system **300** is embodied in a standard headphone enclosure **344** which is to be used to deliver the audio content to the user. Headphone **344** includes an audio output device in the form of a left speaker **336** and a right speaker **338** coupled together by a band **346** which supports the headphone on the user. It is noted that conventional headphone enclosures come in various shapes and sizes and types and that the present disclosure should not be limited to the headphone illustrated in **FIG. 3**. The ASPCU **304** may be disposed in either of the speaker housings **336**, **338** and the various input/output devices may be located on either speaker housings **336**, **338** and/or on the band **346**. For this illustration, audio input port **316**, wireless input port **318** and input/output port **334** are disposed on the left speaker housing **336**; microphone input device **322** and switch **342** are disposed on the right speaker housing **338**.

[**0035**] An application of the embodiment shown in **FIG. 3** is best illustrated in the context for sound devices that are used in work environments. If a worker wishes to enjoy audio content in a totally immersive environment by wearing headphone **344** connected to an audio content device **302** and yet when another worker needs to get the attention of the first subject worker, the system **300** would allow the other worker to be heard while interrupting and replacing the audio content. Thus, when the worker using the subject device is not needed by coworkers, he can enjoy his audio content without any ambient noise or distraction, and yet when he is needed by other workers, the ASPCU **304** will immediately interrupt that audio content to alert the worker. These audio interruptions can be intelligently and automatically selected based on user programming, and subsequently function with or without user defined input. For instance, the user could program the headphone device **344** so that a certain word would trigger the audio content to be interrupted. When a coworker says this word, the device would automatically switch the outside sound channel (which in this case would be the work environment) into the headphone. Further, the user could also select certain key sounds (for instance, an alarm bell) so that when the ASPCU **304** recognizes this sound, sounds from within the users work environment would automatically replace the previous audio content the user had been listening to.

[**0036**] In a further embodiment, the system **300** may employ multiple MIDs. For example, MID **332-2** may be placed remotely from where the user or ASPCU **304** is located. In this embodiment, the ASPCU may include multiplexing circuitry to receive multiple inputs from the multiple MIDS. Furthermore, the ASPCU may include a digital signal processor employing noise detection technology for determining which of the plurality of MIDs is active and subsequently controlling the multiplexer to receive the active MID.

[**0037**] As another example of the advantages of the present disclosure, **FIG. 4** illustrates another embodiment of the system of the present disclosure where a user while driving a vehicle can listen to audio content from any source while at the same time being able to selectively hear interior occupants of the vehicle and outside sounds and noises. These exterior sounds would include emergency vehicles, loud and abrupt warning noises, human voices, and other pre-selected noises. Furthermore, the audio content may also be interrupted by select passenger noises. These noises could be selected by the user or defaulted by the manufacturer. An example of these noises would be the word "help", or the word "interrupt" spoken by any person in the vehicle. A loud and abrupt sound could also be used as a trigger to interrupt the audio content. As shown in **FIG. 4**, a vehicle **452** will include ASPCU **404** coupled to the vehicle's audio system. Preferably, the ASPCU **404** will be disposed in the dashboard for facilitating connection to the vehicle's audio system, e.g., radio, CD player, etc. A first MID **422** will be disposed in the passenger cabin to detect speech uttered by passengers of the vehicle. A second MID **422-2** will be located on an outside surface of the vehicle and may be part of an external antenna used for the vehicle's radio or cellular phone. Upon detection of a predetermined pattern by the ASPCU **404**, audio content being played over the vehicle's front speaker **438** and rear speaker **436** will be interrupted and sound detected by either the first or second MID **422**, **422-2** will be played over the speakers **436**, **438**.

[**0038**] In another less preferred embodiment of the present disclosure, no interior environment noises would interrupt the selected audio content. This application of the present disclosure would not only include automobiles but other transportation devices including boats, motorcycles/scooters, personal transportation devices such as the Segway device, aircraft, and other transportation devices.

[**0039**] While the disclosure has been shown and described with reference to certain preferred embodiments thereof, it will be understood by those skilled in the art that various

changes in form and detail may be made therein without departing from the spirit and scope of the disclosure as defined by the appended claims.

What is claimed is:

1. An apparatus for selectively switching between audio channels, the apparatus comprising:

a first audio input connection for receiving audio content;

a second audio input connection for receiving an acoustic signal;

a controller for receiving the audio content and acoustic signal and for determining whether to output the audio content or acoustic signal; and

an output connection for outputting the determined signal.

2. The apparatus as in claim 1, further comprising a switching mechanism having a first and second position controlled by the controller, wherein in the first position the first audio input connection is coupled to the output connection and in the second position the second audio input connection is coupled to the output connection.

3. The apparatus as in claim 1, wherein the first audio input connection is a hardwired port for receiving audio content through a wire.

4. The apparatus as in claim 1, wherein the first audio input connection is a wireless port for receiving audio content wirelessly.

5. The apparatus as in claim 1, wherein the second audio input connection includes at least one microphone input device for generating an acoustic signal from sound external to the apparatus.

6. The apparatus as in claim 5, wherein the at least one microphone input device is located remotely from the second audio input connection.

7. The apparatus as in claim 6, wherein the remote microphone input device communicates to the second audio input connection wirelessly.

8. The apparatus as in claim 2, wherein the controller receives the acoustic signal, determines if the acoustic signal matches a predetermined pattern and, if the acoustic signal matches the predetermined pattern, positions the switching mechanism to couple the second audio input connection to the output connection.

9. The apparatus as in claim 8, wherein the predetermined pattern is digitized human speech.

10. The apparatus as in claim 8, wherein the predetermined pattern is digitized emergency sounds or preselected sound patterns.

11. The apparatus as in claim 8, further comprising a memory device for storing the audio content received while the second audio input connection is coupled to the output connection.

12. The apparatus as in claim 8, wherein the controller positions the switching mechanism back to couple the first audio input connection to the output connection after a predetermined period of time.

13. The apparatus as in claim 8, further comprising a switch for allowing the user to manually position the switching mechanism.

14. The apparatus as in claim 1, wherein the output connection further comprises an audio output device for audibly producing the determined signal.

15. A system for selectively switching between a plurality of audio channels, the system comprising:

an audio content device (ACD) for supplying audio content on a first channel;

at least one microphone input device for generating an acoustic signal from sound external to the system on a second channel;

an audio signal processing control unit (ASPCU) for receiving the audio content on the first channel and the acoustic signal on the second channel and for selectively switching between the first and second channel; and

an audio output device (AOD) for audibly producing sounds from the selected channel.

16. The system as in claim 15, wherein the ASPCU further comprises a switching mechanism having a first and second position, wherein in the first position the first channel is coupled to the audio output device and in the second position the second channel is coupled to the audio output device.

17. The system as in claim 15, wherein the at least one microphone input device is located remotely from the ASPCU.

18. The system as in claim 17, wherein the remote microphone input device communicates to the ASPCU wirelessly.

19. The system as in claim 16, wherein the ASPCU receives the acoustic signal, determines if the acoustic signal matches a predetermined pattern and, if the acoustic signal matches the predetermined pattern, positions the switching mechanism to couple the second channel to the audio output device.

20. The system as in claim 19, further comprising a memory device for storing the audio content received while the second channel is coupled to the audio output device.

21. The system as in claim 19, wherein when the second channel is coupled to the audio output device, the ASPCU pauses the audio content supplied from the audio content device.

22. The system as in claim 19, wherein the predetermined pattern is digitized human speech.

23. The system as in claim 19, wherein the predetermined pattern is digitized emergency sounds or preselected sound patterns.

24. The system as in claim 19, wherein the ASPCU positions the switching mechanism back to couple the first channel to the audio output device after a predetermined period of time.

25. The system as in claim 19, further comprising a switch for allowing the user to manually position the switching mechanism.

26. The system as in claim 15, wherein the audio content device is located remotely from the ASPCU and communicates to the ASPCU wirelessly.

27. A headphone for selectively switching between a plurality of audio channels, the headphone comprising:

an audio content device (ACD) for supplying audio content on a first channel;

at least one microphone input device for generating an acoustic signal from sound external to the headphone on a second channel;

an audio signal processing control unit (ASPCU) for receiving the audio content on the first channel and the

acoustic signal on the second channel and for selectively switching between the first and second channel; and

first and second speakers for audibly producing sounds from the selected channel.

28. The headphone as in claim 27, wherein the ASPCU further comprises a switching mechanism having a first and second position, wherein in the first position the first channel is coupled to the first and second speakers and in the second position the second channel is coupled to the first and second speakers.

29. The headphone as in claim 27, wherein the at least one microphone input device is located remotely from the headphone.

30. The headphone as in claim 29, wherein the remote microphone input device communicates to the ASPCU wirelessly.

31. The headphone as in claim 28, wherein the ASPCU receives the acoustic signal, determines if the acoustic signal matches a predetermined pattern and, if the acoustic signal matches the predetermined pattern, positions the switching mechanism to couple the second channel to the first and second speakers.

32. The headphone as in claim 31, further comprising a memory device for storing the audio content received while the second channel is coupled to the first and second speakers.

33. The headphone as in claim 31, wherein when the second channel is coupled to the first and second speakers, the ASPCU pauses the audio content supplied from the audio content device.

34. The headphone as in claim 31, wherein the predetermined pattern is digitized human speech.

35. The headphone as in claim 31, wherein the predetermined pattern is digitized emergency sounds or preselected sound patterns.

36. The headphone as in claim 31, wherein the ASPCU positions the switching mechanism back to couple the first channel to the first and second speakers after a predetermined period of time.

37. The headphone as in claim 31, further comprising a switch for allowing the user to manually position the switching mechanism.

38. The headphone as in claim 27, wherein the ASPCU receives the acoustic signal, determines if the acoustic signal matches a predetermined pattern and, if the acoustic signal matches the predetermined pattern, couples the first channel to the first speaker and couples the second channel to the second speaker.

39. The headphone as in claim 28, wherein the audio content device is located remotely from the headphone and communicates to the ASPCU wirelessly.

40. A method for selectively switching between a plurality of audio channels in an audio device, the method comprising the steps of:

supplying audio content on a first channel of the audio device;

generating an acoustic signal from sound external to the audio device on a second channel;

receiving the audio content on the first channel and the acoustic signal on the second channel and selectively switching between the first and second channel; and

audibly producing sounds from the selected channel.

41. The method as in claim 40, further comprising the steps of:

determining if the acoustic signal matches a predetermined pattern; and

if the acoustic signal matches the predetermined pattern, selecting the second channel to be audibly produced.

42. The method as in claim 41, wherein the predetermined pattern is digitized human speech.

43. The method as in claim 41, wherein the predetermined pattern is digitized emergency sounds.

44. The method as in claim 41, further comprising the step of switching back to the first channel after a predetermined period of time.

45. The method as in claim 41, further comprising the step of storing the received audio content when the second channel is selected.

46. The method as in claim 41, further comprising the step of pausing the audio content when the second channel is selected.

* * * * *

Exhibit PA-B

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(54) **SYSTEM AND METHOD FOR SELECTIVE CONTROL OF ACOUSTIC ISOLATION IN HEADSETS**

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(22) Filed: **Apr. 24, 2001**

Related U.S. Application Data

(63) Non-provisional of provisional application No. 60/199,283, filed on Apr. 24, 2000.

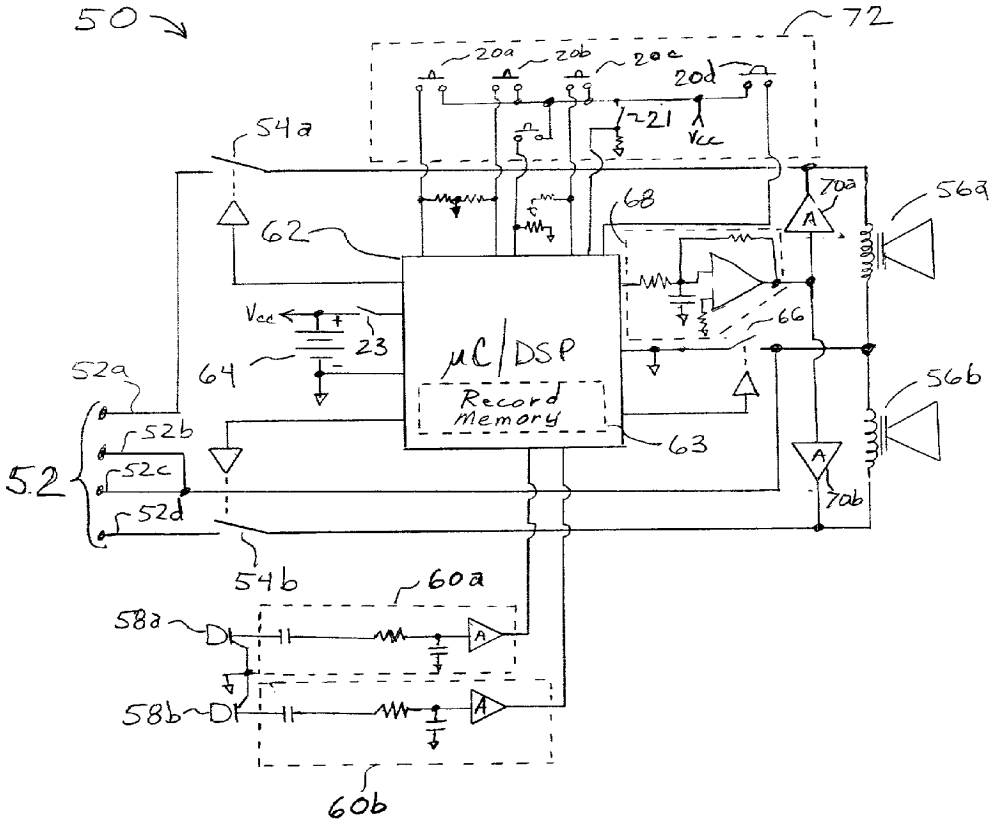
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(57) **ABSTRACT**

An apparatus and method for providing controlled acoustic isolation within various forms of headsets. The invention provides manual and automatic mechanisms for changing the amount of acoustic isolation provided by the headsets. Sounds in the environment that the user wishes to be made aware of can be programmed into a set of stored sound selection characteristics. In response to correlation of the stored sound characteristics with sounds in the external environment the headset decreases acoustic isolation by coupling signals from one or more external microphones to the audio conversion elements within the earpieces. Alternatively, the apparatus can respond to sounds to be blocked by increasing acoustic isolation. A manual control may be activated by the user to decrease acoustic isolation at their discretion.



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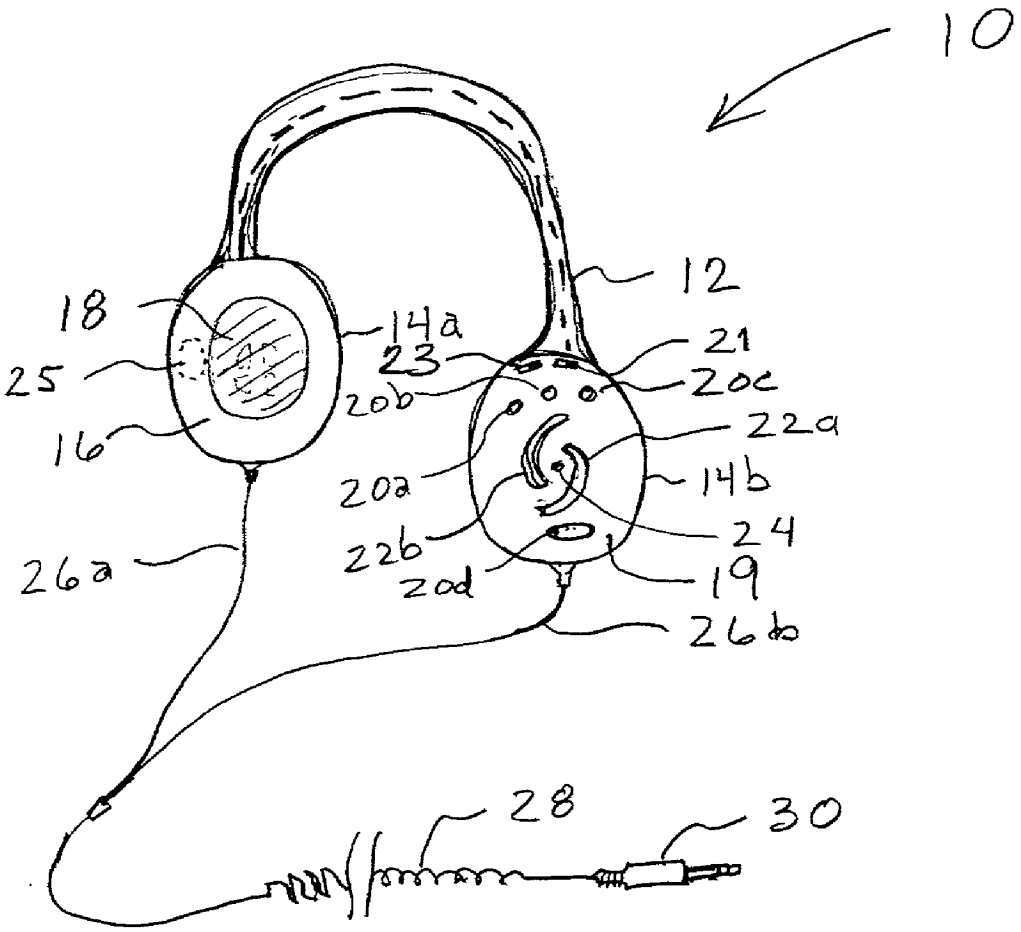


FIG. 1

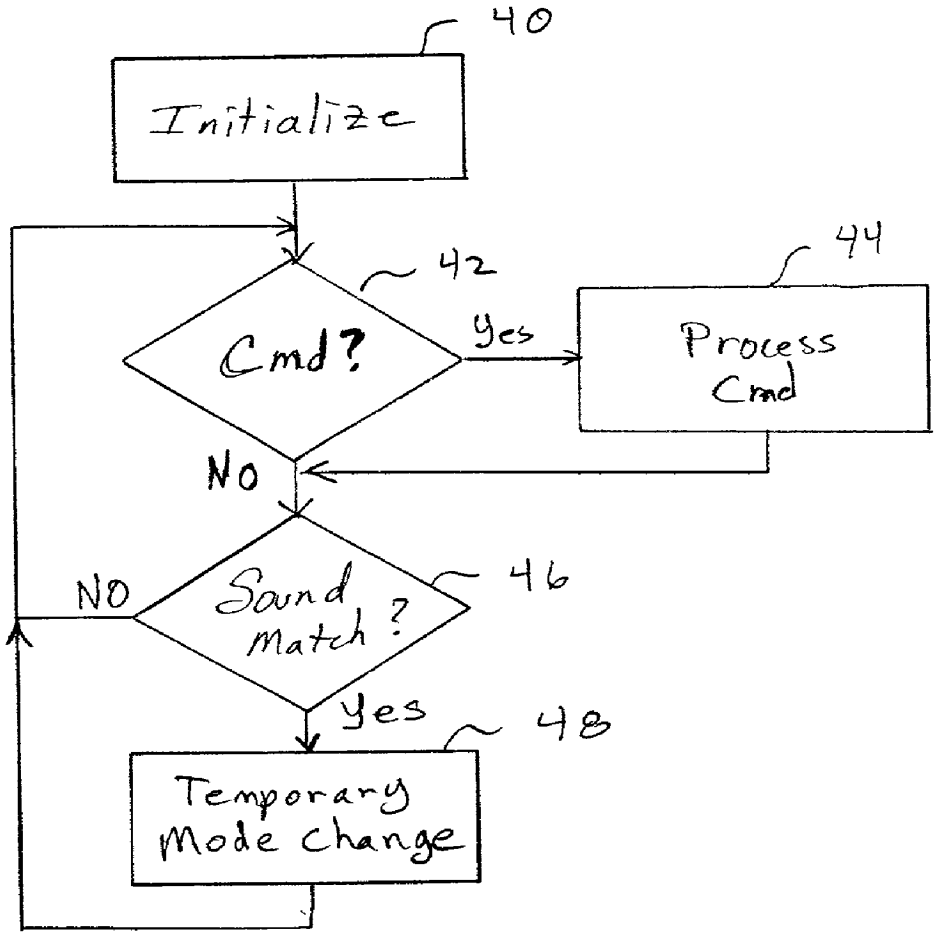

40
Initialize
42
Cmd?
Yes
44
Process Cmd
No
NO
Sound Match ?
46
Yes
48
Temporary Mode Change

FIG. 2

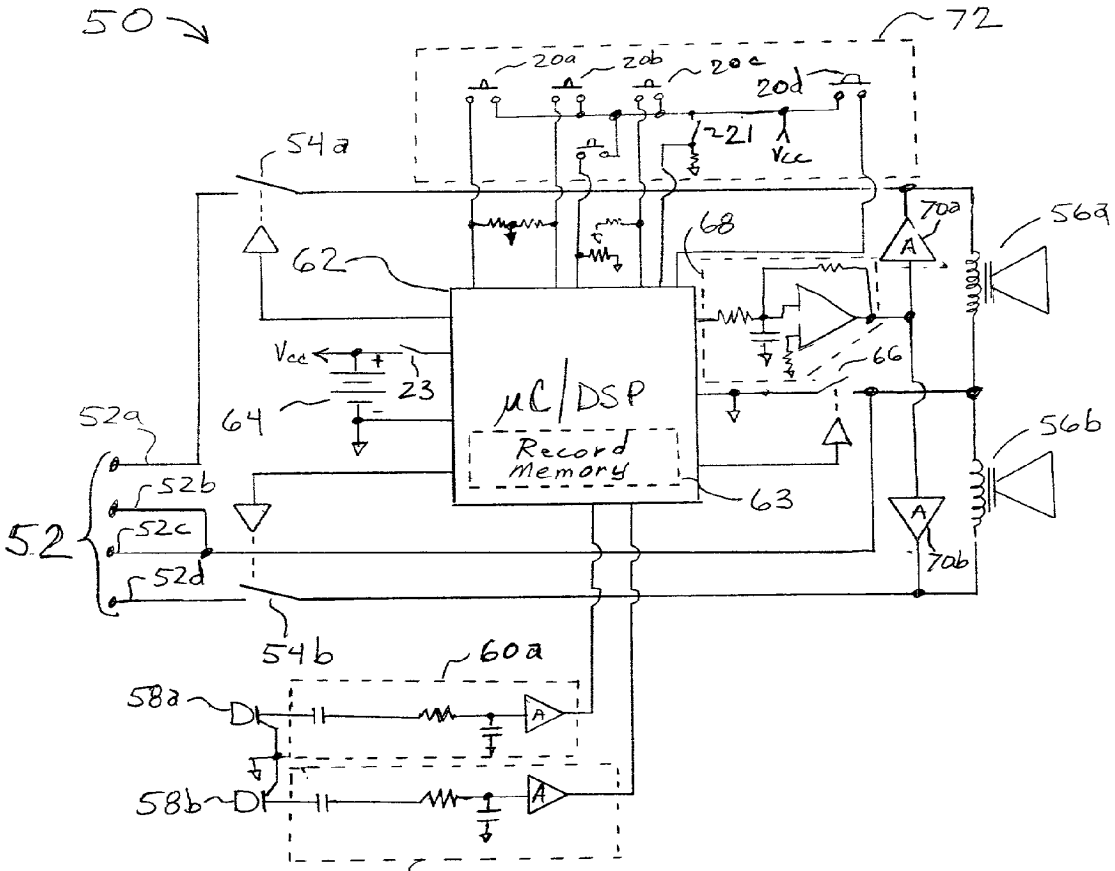

50
72
20a
20b
20c
20d
21
Vcc
54a
62
68
70a
56a
66
Vcc
23
64
μC/DSP
Record Memory
63
52a
52b
52c
52d
52
56b
70b
54b
60a
58a
58b
60b

FIG. 3

Exhibit PA-B

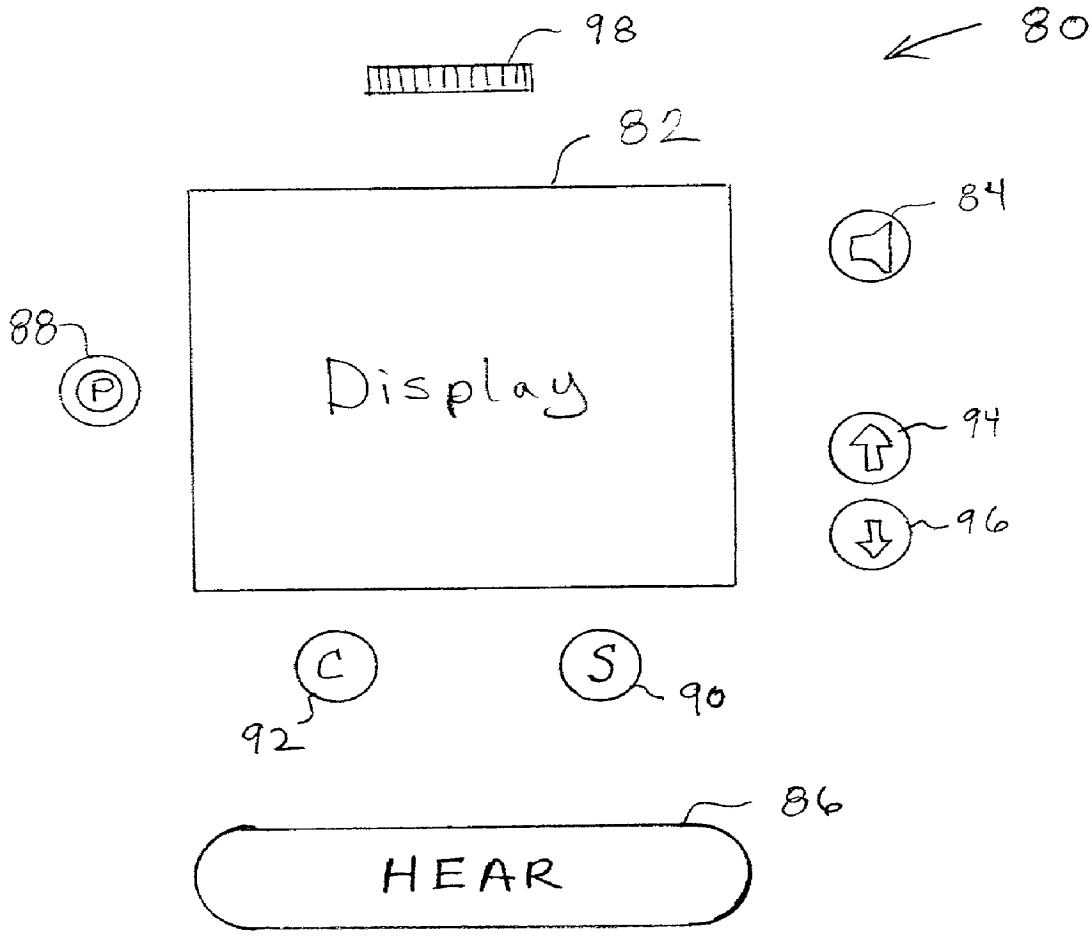


FIG. 4

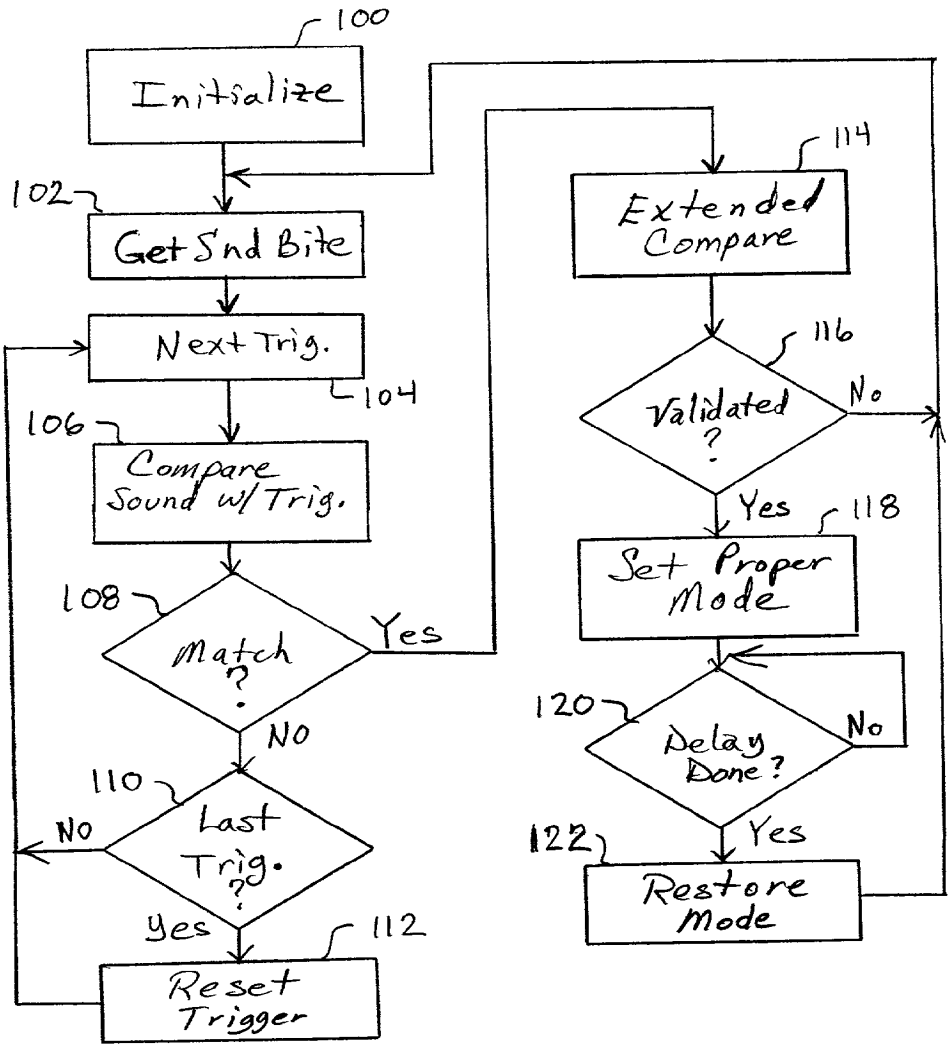

100
Initialize
102
Get Snd Bite
Next Trig.
104
106
Compare Sound w/ Trig.
108
Match ?
Yes
No
110
Last Trig. ?
No
yes
112
Reset Trigger
114
Extended Compare
116
Validated ?
No
Yes
118
Set Proper Mode
120
Delay Done?
No
Yes
122
Restore Mode

FIG. 5

SYSTEM AND METHOD FOR SELECTIVE CONTROL OF ACOUSTIC ISOLATION IN HEADSETS

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application claims priority from U.S. provisional application Ser. No. 60/199,283 filed on Apr. 24, 2000.

STATEMENT OF FEDERALLY SPONSORED RESEARCH OR DEVELOPMENT

[0002] Not Applicable

REFERENCE TO A MICROFICHE APPENDIX

[0003] Not Applicable

BACKGROUND OF THE INVENTION

[0004] 1. Field of the Invention

[0005] This invention pertains generally to headsets and more particularly to an apparatus and method of providing headsets with sound selective acoustical isolation from the external environment.

[0006] 2. Description of the Background Art

[0007] A number of situations exist in which an individual is acoustically isolated from an external environment. The isolation often results in inefficiency or inconvenience, but in other cases it may place the individual, or others nearby, at risk of bodily harm. One primary isolation situation arises from the use of headsets. A wide variety of traditional headsets exist for noise abatement or for listening to an audio source, or communication. Headsets utilized for noise abatement are often constructed as simple passive non-electronic "ear-muff" style devices that block noises within a high ambient acoustic environment. Audio headsets are ubiquitous today for the reproduction of audio sources, in particular, music. Approximately twenty six million (26×10^6) audio headsets were sold in the year 2000. In addition, headsets are utilized for maintaining communication in a high ambient noise environment, such as in an aircraft cockpit, stadium, and so forth. Generally these headsets have earpieces which either fit over each ear, or are positioned upon each ear and retained in position by a headstrap. Hearing protection headsets are used in high noise environments, such as factory floors, flight decks of aircraft carriers, airport crews, rifle ranges, and in any environment in which the ambient acoustics could prove damaging over time to the hearing of a party within that environment. Audio source headsets are often referred to as "headphones", or "stereo headphones" and they allow a party to listen to an audio source while being acoustically isolated from the external environment. Audio source headsets are most often used for listening to music, such that the individual wearing the set of headphones may listen to music without being disturbed by nearby ambient sounds (sounds, talking), or the wearer may listen to audio at high decibel volume levels without unduly disturbing others. The numerous lightweight open-air headsets on the market provide reduced acoustic isolation between the listener and the environment, and provide the aforementioned benefits to a more limited extent. Often headphones are utilized in applications for communicating between parties while blocking out sounds of the external environment, such as headphones used by pilots and other flight crew members, and coaches. Headsets are often further equipped with noise cancelation circuitry which introduces a negative environmental noise feedback signal (opposing phase) into the audio source being received, so that the ambient noise is further attenuated and thus acoustic isolation of the wearer from the environment increases without a concomitant increase in the pressure directed between the earpieces on the skull of person wearing the headphones (often commonly referred to as "head-clamping" pressure). It will also be appreciated that some personnel wear headphones strictly for noise abatement to provide hearing protecting within a high ambient noise environment. The intent within each of these headphones is to place an acoustical barrier between the ears of the individual and the sounds within the external acoustic environment. When wearing headphones, especially of the closed cup variety, the wearer is tuned out from the external environment; which indeed is largely the intent.

[0008] 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. Furthermore, the nonselective nature of the majority of noise cancelation devices provides insufficient attenuation of bothersome noises, such as the roar of an aircraft APU, while it strives to attenuate all external sounds.

[0009] Therefore, a need exists for the development of methods for controlling acoustic isolation, so that important events are not missed and non-important noises can be properly identified and ignored. The sound selective acoustic isolation provided in accordance with the present invention satisfies that need, as well as others, and overcomes deficiencies in previously known techniques.

BRIEF SUMMARY OF THE INVENTION

[0010] The present invention of an apparatus for sound selective acoustic isolation provides a number of benefits for users of audio related devices. By way of example the apparatus is exemplified as an environmentally responsive apparatus which is acoustically coupled to the ear(s) of a wearer to provide sound selective acoustical isolation from the environment. The apparatus is typically worn by and acoustically coupled to a user, referred to as the wearer. The acoustical isolation being characterized by the amount of sound attenuation in decibels which occurs as a result of the acoustical coupling of the device to the ear(s) of a wearer. An apparatus and method are described 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 characteristics.

[0011] Manual changes in acoustic isolation are provided in response to activation of a selection device attached to the headset, wherein the isolation of external sounds is decreased to improve the ability of the wearer to hear the external sounds. The changes in acoustic isolation may be

controlled by a signal conditioning circuit, or signal processor, capable of selectively conditioning and routing external sound signals from an associated microphone to the audio conversion device within the earpiece. The manual change in acoustic isolation may be coupled with attenuation of incoming audio signals, such as music programming, or communications, so that the ability of the user to hear the external environment is improved. Attenuation of incoming audio signal preferably exceeds twenty decibels (20 dB). Preferably the selection device comprises a button on the exterior of the earpiece, or headset, that changes the acoustic isolation for only a short period of time, sufficient to allow the user to be alerted to conditions within the external environment.

[0012] In automatic mode the apparatus monitors acoustics within the environment against a set of selection criterion and upon finding a match, changes the mode of the headset wherein the amount of acoustical isolation provided is altered temporarily, or until a "reset condition" is received. Automatic changes in acoustic isolation may be similarly coupled with attenuation of the incoming audio signals, as well as the generation of alerts, matching sound "echos", and so forth to further enhance user cognition of the external environment.

[0013] The apparatus is readily embodied within various forms of headsets, such as audio, communication, noise abatement, noise cancelation, and so forth. The term headset is utilized herein as this is the most common configuration for an apparatus to be acoustically coupled to a single wearer. A headset generally comprises a pair of earpieces that are capable of being retained in a position proximal the ear of a wearer. Typically, headsets (or headphones) are utilized for modifying the personal acoustic environment for a single wearer, such as in the case of listening to an audio program source, maintaining communication (such as between pilot and copilot), blocking external sounds, enhancing external audio, and so forth. It will be appreciated, therefore, that the apparatus referred to as a headset may be configured in various mechanical configurations, such as mounted as earpieces within a helmet, single sided headset, earbuds, and so forth.

[0014] The environmentally responsive headsets exemplified in the present invention are capable of changing isolation mode in response to a manual trigger, or automatically in response to matching a sound from the external environment with a stored element of sound characterization information which is programmed within the apparatus. Sounds from the external environment are preferably registered by a microphones, with one microphone attached, or integrated, into the exterior of each earpiece. Each microphone converts the received external sound energy to an external sound signal. The mode change of the apparatus generally comprises altering the level of acoustical isolation accorded the wearer by the headset. The mode of the headset may be changed by any of a number of mechanisms, for example, by temporarily routing external sound to earpieces (a "hearthrough mode"), replaying the detected sound (temporally displaced sound—an "echo back"), generating an alert signal ("user alert tone"), 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), combinations of the preceding modes, and so forth. Furthermore, if the headset is configured for listening to a programmed sound, such as that being received over a wired connection, or wireless receiver, then 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.

[0015] The selective acoustical isolation provided by the present invention preferably augments the operation of active noise cancelation circuitry within the headset to provide a number of additional benefits. It will be appreciated that noise canceling headphones also require the incorporation of a signal processing circuit coupled to one or more microphones so that the signal registered by the microphone(s) is modified into a negative feedback signal which modifies the program audio signal being received to thereby reduce the apparent receipt of noise acoustically coupled through the housing of the headset. The present invention increases headset safety and convenience by providing a hearthrough mode for the headset wearer, while allowing the wearer to select which sounds are to be heard and which are to be blocked.

[0016] The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external acoustic events (external sounds) may be correlated. 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.

[0017] One aspect of the present invention is a "hearthrough" mode which routes the acoustics of the environment, preferably at a user selected amplification level, for some period of time to the headset wearer when a hearthrough trigger occurs. The hearthrough trigger may comprise the manual activation of a hearthrough button by the wearer, or by automatic sound detection within the apparatus. In automatic sound detection a signal processing circuit receives signals from one or more microphones, or similar acoustic conversion devices that are preferably incorporated on the exterior portions of the earpieces, or otherwise attached to the headset. The signal processing unit compares the microphone signals against a set of stored selection criterion and upon finding a match alters the amount of acoustical isolation provided by the headset, and/or provides an alerting audio signal to the wearer. 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. The wearer is thereby directly provided with audio from the external environment.

[0018] The sounds to be correlated may be loaded into the apparatus as elements within a set of programmed sound selection criterion by way utilizing the apparatus to record sounds, or characteristics sounds, which are to be correlated. Many of the sounds for which the user desires reduced acoustic isolation from the environment will be unique to their particular environment, for example their spoken name, the sound of the doorbell, and so forth. Alternatively, pre-

programmed sounds may be selected for the selection criterion, loaded from a storage media such as a CD-ROM, or downloaded into the unit, either directly, as a sound, as a set of sound characterizations, as a program entity (algorithm), or as a combination thereof. For example, a single complex selection criterion of "siren" may comprise an algorithm that is capable of discerning a wide variety of common sirens in use. Utilizing this as a preprogrammed algorithm, or a downloaded algorithm would provide a broad discernment of sirens without the need to record one or more specific siren sounds.

[0019] Preferably, the unit is configured so that it can be programmed with an audio stream from a source, such as the telephone, or audio from the Internet. When set in a program download mode, the audio source is started and the unit detects, interprets the incoming data, and converts the data to a programmatic entity which provides the desired sound discernment. A simple, albeit bandwidth inefficient, method of tonal code communication may be found in the use of DTMF, wherein sets of dual tones comprise the digits 0-9. It will be recognized that the speed at which digital data may be communicated as audio for an acoustically coupled unit is generally limited to approximately 2400-9600 baud. Aside from actual sound snippets, however, the data being loaded can comprise scripts and other sound related descriptions that can be encoded in less space than individual instructions for a DSP chip. To provide convenience to the user, a library of sounds may be provided, such as over the Internet, or from a CD, wherein the user can select prerecorded sound patterns, sound characterizations, and algorithms to suit their needs. It should, however, be appreciated that the simpler preprogrammed functions of the device may be adequate for a large percentage of intended applications. Alternatively, the apparatus of the present invention may be configured with a conventional jack or similar mechanism through which programming may be received.

[0020] To increase the accuracy of sound discernment it is preferable that the user select a sound "type" prior to recording a sound, options may include such types as general sound, speech, loud sounds, repetitive sounds, alarms, telephone rings. The apparatus is thereby capable of extracting proper sound characterization from the sound being "recorded" from the environment. The improved characterization allows for greater discernment accuracy relative to wearer purpose. As an example, consider that correlation performed for the spoken sound "Robert" can be discerned from the external environment in a speaker independent manner utilizing speech recognition characterization of the sound. In contrast, if the sound were to be stored only as a general sound it would not be characterized as speech, wherein speaker independence is lost and discernment would be restricted to a very narrow interpretation.

[0021] A wide range of selection criterion may be used upon which to correlate audio signals received from the external environment. The environmental audio may be analyzed by numerous current methods, including parametric analysis, waveform comparisons, Fourier analysis, and in a larger context voice recognition. 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.

[0022] According to one aspect of the invention, the user can program the headset to recognize specific sounds within their specific environment. 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. The preferred method of changing the mode comprises changing the level of acoustical isolation provided by the headset to the wearer. For example, 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. Preferably, 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. The specific sounds to be recognized (correlated) with the external environment may be contained within the headset as a prerecorded sound, or as a sound that is recorded by the user from the target environment. The repertoire of triggering sounds can therefore be modified according to the conditions under which the wearer wants to be alerted, or interrupted.

[0023] In the case of hearing protection headsets (providing noise abatement) implemented according to the present invention; the external audio is attenuated by the earpieces, and optionally a noise canceling circuit within the headset, until the sound being received by the signal processing unit correlates with one or more of the stored selection criterion, at which time the mode of the headphone changes so that the wearer is made aware of the event associated with the stored selection criterion. One preferred method of changing the mode of the headphone is to temporarily couple the sound received from the microphone to the audio transducers, or speakers, within the headset earpieces. The wearer, therefore, is alerted and can clearly discern the danger themselves. The sound can be augmented, preferably depending on the duration, and nature of the sound being recognized, by playing back a recorded version of the precise received segment of received sound (an "echo") which was correlated with the stored selection criterion. A number of alternatives may be similarly produced, such as playing the recorded sound associated with the stored selection criterion, or the playing of audio alert signals by themselves or in combination with the aforementioned sounds.

[0024] Once a correlation occurs of a sound selected for hearthrough, the wearer is preferably provided with the environmental sounds for a period of time, such as three to five seconds (3S -5S), so that they may properly analyze the external environment to assess the situation. The wearer may at this time press a "hearthrough" button on the headset to continue listening to the environment. It will be appreciated that the "hearthrough" button would provide a similar result as removing the headset, however, it may be performed far more quickly and easily than taking off, holding the headsets, and then subsequently restoring them upon the ears.

[0025] The present invention may also be utilized within headsets which receive an audio channel, such as music, or communications, which is converted to sound within the earpieces to which the wearer is listening. The signal processing unit operates in a similar manner to the aforesaid

hearing protection headset, however, when a correlation is detected between the environmental sounds and the stored sound parameters, the audio channel is switched off or attenuated so that the external environmental sounds may be heard. The attenuation of the audio channel sound source is preferably in excess of twenty decibels (20 dB).

[0026] An object of the invention is to decrease both inconvenience and attendant safety risks for persons exposed to uncontrolled acoustic isolation.

[0027] Another object of the invention is to increase the safety and convenience of headsets utilized for noise abatement, communication, and sound reproduction.

[0028] Another object of the invention is to provide an individual with the ability to selectively control the acoustical isolation between themselves and their acoustic environment.

[0029] Another object of the invention is to provide a wearer with reduced acoustical isolation from selected external sounds that sufficiently correlate with one or more stored sounds for which the user should not be isolated.

[0030] Another object of the invention is to provide a wearer with increased acoustical isolation from selected external sounds that sufficiently correlate with a one or more stored sounds for which the user is to be isolated (blocked).

[0031] Another object of the invention is to provide an apparatus having sound selective acoustic isolation which augments the benefits derived from various forms of active noise cancelation circuitry.

[0032] Another object of the invention is to allow user programming of environmental acoustics against which external sounds are to be compared to trigger changes in the isolation provided by the acoustic barrier.

[0033] Another object of the invention is to provide alerts to the wearer in response to the detection of selected sounds within the external environment.

[0034] Another object of the invention is to provide a manually operated activation mechanism that changes the mode to one of decreased acoustic isolation, wherein the wearer can listen to the external environment without the necessity of removing headsets.

[0035] Another object of the invention is to provide the capability to record sounds and patterns of sound recognition for which a headset mode change is to take place upon detection.

[0036] Another object of the invention is provide a mechanism wherein the user is capable of indicating a sound "type" prior to recording of a correlation sound, so that sound discrimination may be improved.

[0037] Another object of the invention is to allow the wearer to listen to the external environment for a period of time after a stored hearthrough sound is correlated with a sound occurring in the environment.

[0038] Another object of the invention is to provide a display upon which programming of the headset may be indicated.

[0039] Another object of the invention is to provide audio feedback as to the recorded correlation sound such that the user can easily make recording selections.

[0040] Another object of the invention is to provide a headset having externally directed microphones and surrounding structures which are capable of providing directional sound, similar to that of unaided ears, while in hearthrough mode.

[0041] Further objects and advantages of the invention will be brought out in the following portions of the specification, wherein the detailed description is for the purpose of fully disclosing preferred embodiments of the invention without placing limitations thereon.

BRIEF DESCRIPTION OF THE DRAWINGS

[0042] The invention will be more fully understood by reference to the following drawings which are for illustrative purposes only:

[0043] **FIG. 1** is a perspective view of a headset according to an embodiment of the present invention having external microphones and programming buttons, and shown configured for listening to audio programming.

[0044] **FIG. 2** is a flowchart of sound discrimination according to the present invention.

[0045] **FIG. 3** is a schematic of selective sound discrimination circuitry according to an embodiment of the present invention.

[0046] **FIG. 4** is a facing view of a user interface according to an aspect of the present invention, shown with a display, manual selectivity mechanism, manual hearthrough button, and interface buttons.

[0047] **FIG. 5** is a flowchart of main loop functions according to an aspect of the present invention, showing segmented sound triggering.

DETAILED DESCRIPTION OF EMBODIMENT(S)

[0048] Referring more specifically to the drawings, for illustrative purposes the present invention is embodied in the apparatus generally shown in **FIG. 1** through **FIG. 5**. It will be appreciated that the apparatus may vary as to configuration and as to details of the parts without departing from the basic concepts as disclosed herein.

[0049] **FIG. 1**. is a headset **10** according to an embodiment of the present invention whose audio attenuation of external audio sounds can be selectively changed in response to the correlation of those external sounds with sounds and sound characterizations stored within the headset apparatus. The headsets are exemplified for listening to an audio source over a wired connection, however, it will be appreciated that the present invention is applicable to wireless headphones, communication headsets, noise abatement headsets, headsets incorporated into headgear such as helmets, single ear headsets, earbud style devices, and so forth. The headset **10** is shown comprising an earpiece positioning member **12**, earpieces **14***a*, **14***b*, with cushioned inner faces **16**, an inner cup portion **18** which retains an audio annunciator directed toward, and acoustically coupled to, the ear canal. An exterior surface 19 comprises control buttons **20***a*-**20***d*, programming switch **21**, audio collection grooves **22***a*, **22***b*, power switch **23**, microphone **24**, and hidden enablement switch **25**.

[0050] The audio collection grooves **22***a*, **22***b* and/or ridges which surround the microphone preferably mimic the acoustical function of the structures surrounding the exterior of the human ear, wherein sound is redirected toward the ear canal. In this way the response of the microphones may be optimized and configured to provide a peripheral field of response similar to the ear of the wearer such that the user can discern directional characteristics of sound when the headsets are configured to listen to the external environment. It will be appreciated that conventional active noise cancelation headsets have no need of these ear-fold structures, since they do not provide the wearer with the ability to listen to the external environment—that is unless the headsets are removed from the head of the wearer. The earpieces of the headphones are shown having wires **26***a*, **26***b* leading to a joined coiled wired **28** upon which a stereo phono-plug **30** is attached. It will be appreciated that the configuration of the headset may vary using different earpieces and that the positioning member **12** is optional and depends on the structure of the earpieces.

[0051] **FIG. 2** is a flowchart of basic sound discrimination within the headsets of the present invention. The flowchart assumes that a series of sounds and sound patterns to be discriminated has previously been programmed. Upon power-up of the sound discriminator, such as by use of a power-on switch or the detection of head compression by hidden detection switch **25** of **FIG. 1**, the circuit is initialized at block **40** wherein the hardware and sounds are prepared for discerning the external events. A check is made to detect command entry at block **42**, because the user at any time may elect to program a new sound or alter the mode characteristics of the headset. If a command entry is detected, such as the pressing of a button, then a command processing routine is entered, block **44**, which interfaces with the user. Segments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit of the headset. To speed discernment, a small segment of audio is compared with all the stored trigger conditions, and upon meeting a first phase of correlation, the discernment process continues to check further portions of the sound against the stored sounds and characteristics. If the external sound correlates with the programmed sound at block **46**, then the mode of operation of the headphone is changed at block **48** to that which has been programmed by the user. It will be appreciated that mode changes should generally comprise changing the amount of acoustical isolation from the environment. Such as by entering a hearthrough mode wherein sound registered by the microphones is conditioned and directed to the audio annunciators within the earcups. 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. 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.

[0052] Alternatively, when the correlated sound is one selected as a sound to be blocked by the headset, the mode change can provide enhanced attenuation of the pattern of sound in a number of ways. The sound pattern matching may be utilized to modulate an active noise cancelation circuit on headsets so equipped. Upon detecting the sound to be blocked, the headset may increase the selected level of noise cancelation from a low, or intermediate value, to a higher level of noise cancelation. The noise cancelation algorithms themselves can also accept the sound pattern as input, wherein the noise cancelation may be tuned to the sound to be blocked. In this way the noise cancelation becomes "selective sound noise cancelation" and is capable of providing high degrees of isolation to selected acoustic signature patterns without the necessity of attempting to cancel all external noise. It will be appreciated that the user can, therefore, configure an active noise cancelation circuit to attenuate all sounds by a given amount, while using the selective attenuation provided by the present invention to specifically attenuate (block) sounds which correlate to the stored sound.

[0053] **FIG. 3** is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention. Audio source connections **52***a*-**52***d* provide an analog audio channel which is routed through a pair of switches **54***a*, **54***b* to conventional audio conversion devices **56***a*, **56***b*, such as speakers. When listening to the audio source, the switches **54***a*, **54***b* are in the closed position. Microphones **58***a*, **58***b*, or similar audio conversion devices, convert sounds within the external acoustical environment to electrical signals which pass through the conditioning circuitry **60***a*, **60***b* into analog-to-digital converter (ADC) inputs of a combined digital signal processor and microcontroller **62** which contains an internal program store (ROM), random access memory (RAM), and has the ability to store sound waveforms, sound characteristics, and sound profiles **63**, to which the external sounds are to be correlated. Power for the DSP/microcontroller is exemplified as battery power **64**, although the power may be derived from a variety of sources, such as the bias voltage or signal voltage as provided on the input audio source connections **52***a*-**52***d*. The DSP/microcontroller **62** is capable of processing the audio signals from the microphones **58***a*, **58***b* and correlating these signals with a set of audio selection criterion which are stored in the sound characterization memory **63**. When the microphone input correlates with the stored parameters, such as detecting the word "Hello!", then the DSP/microcontroller **62** within the present embodiment preferably switches off the input sources by opening switches **54***a*, **54***b*, engaging switch **66** to tie into the ground of the DSP with ground for the speakers **56***a*, **56***b*, and then outputting via a digital-to-analog converter the sound collected from the microphones to an amplifier circuit **68** which is then buffered by amplifiers **70***a*, **70***b* to the separate channels of the earpiece speakers **56***a*, **56***b*. It will be appreciated that the correlation of sound and speech patterns is well known in the art. It will be recognized that the sound from the microphones is thereby selectively passed through to the listener when it correlates to a sufficiently high degree with the stored audio parameters. Once correlation occurs, the environmental sounds collected from the microphones are preferably passed through to the headset wearer for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the "hearthrough" button **20***d* as shown in **FIG. 1**, to keep listening to the environmental audio, or allow the duration to time out and return to listening to their original programming. The use of the "hearthrough" button to control hearthrough may be configured in a variety of ways. One such way is to interpret a

quick button press as a change of state; from non-hearthrough mode to a short period of hearthough, or from a hearthrough mode back to normal non-hearthrough mode. A button press of an intermediate duration, can be used to indicate that hearthrough mode should be entered for a slightly longer than conventional interval, regardless of original state. A button press of long duration may then be interpreted as a signal to lock the headset into hearthrough mode until a quick press of the "hearthrough" button occur.

[0054] To aid in understanding the correlation of external sounds with sounds stored within the apparatus, an example follows for a stored spoken sound of "hello". Speech recognition characteristics relating to the spoken sound of the word "Hello!" are stored within the headset apparatus and the characteristics are compared with the sounds registered by the microphones and fed to the signal processing unit. Thereafter, if the spoken word "Hello" is registered by the microphone from the external environment, the headphone preferably switches to a hearthrough mode to communicate environmental sounds to the wearer. In addition, as it will be appreciated that the spoken word "hello" was blocked during correlation, the sound "hello" received from the environment and just correlated may be played, in essence as an echo upon which an additional alert tone may be superimposed to alert the wearer that a hearthrough correlation has occurred and to indicate that the replayed sound has been reproduced (thereby suffering from a temporal shift) so that the user will not be confused if the echo of "hello" is heard in concert with continued verbiage from the person who uttered the "hello".

[0055] The headset contains a series of buttons including a power switch **23** which allows switching off all power to the headset to conserve power. In addition, an enable switch **25** as shown in **FIG. 1**, may be built into one or more of the earpieces so that when the headsets are placed on the wearers head the headset correlation functions will activate, providing the power switch is set to on, at which time preferably a beep, tone sequence, or the voiced phrase is played over the earpieces to alert the wearer that the environmentally responsive functionality is active. The wearer may toggle from listening to the audio channels and listening to the environmental acoustics by alternately pressing button **20***d* on the lower portion of the headsets. 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 which has been programmed or recorded for discrimination. While listening to an audio source, such as music, the wearer may press button **20***d* and elect to listen to the environment; pressing of button **20***d* again would return the headset to listening to the musical audio source. In this manner the wearer may elect to hear their environment when they so elect, as when responding to a visual cue. The wearer may also press the mode button after the three to five seconds (3 S-5 S) of environmental acoustics have been played following a sound correlation, such as "Hello!", so as to continue listening to the environment. By way of example, the user may program a sound into the headsets by assuring power switch **23** is ON, setting programming switch **21** from normal mode into program mode, and then holding down one of the program channel buttons **20***a*-**20***c* while the programmed sound is occurring, for example: they have their coworker call out "Hello!", or ring their phone. 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. In addition pressing of a program button when the program switch is in normal mode preferably causes the headset to play the stored sound over the headset so that the user may verify which sounds have been stored. It will be recognized that numerous interfaces can be implemented for programming and control of the headsets without departing from the inventive teachings.

[0056] It will be appreciated that audio headphones are often equipped with a volume control to allow the wearer to set the audio level played over the earpieces. The headset may additionally contain a sensitivity control that allows the user to set the sound level required in order to trigger the correlated response. In the example of the word "Hello!" the sensitivity adjustment would select how loud, or alternatively how accurate, the sound would need to be in order to trigger the correlation response.

[0057] The headset may also be equipped with a feedthrough control wherein the user can select the amount of external environment acoustics which are feedthrough for their hearing. Example: sounds are amplified differentially as determined from sound parameters and sound levels and so forth. The user thereby can set the dB amount of isolation they desire from the surrounding environment.

[0058] Referring now to **FIG. 4 a** user interface **80** is shown for an environmentally responsive headset according to an aspect of the present invention. The interface allows the user to program the desired functionality of the headsets, such as operation, modes changes, as well as the selection of preprogrammed sounds, and both the recording and downloading of sounds and sound characterization information for correlation with the external environment. A display **82** provides for the generation of non-audio feedback to the user. Display **82** may be configured as a small graphical LCD display, such as found on a cellular phone. A listen button **84** allows the user to listen to any selected programmed sound or set of programmed characteristics, wherein determinations of their use can be assessed. 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. It will be appreciated that headsets may be configured with only the manual selection of hearthrough, such as provided by button **86**, without departing from the teachings of the present invention. A program button **88** allows for the activation of sound recording, or characterization. It should be recognized that the duration of sound recorded is selected by the user in response to the retention of the program button, such that the user can select the amount of sound which is to be correlated prior to the occurrence of a mode change by the headset. Program button **88** is shown recessed to prevent inadvertent programming. A select button **90** allows selection of an item within a scrolling list shown on display **82**. A clear button **92** allows for clearing a selected response, and for traversing in reverse order the prior screens. A set of up **94** and down **96** scroll buttons allow moving a selection up and down within the display for making user choices. A scrolling wheel **98**, provides for altering the sensitivity of the sound discrimination circuit to the programmed sounds. The sensitivity preferably is a combination of both the amplitude of the

sounds as well as the accuracy of the sounds in relation to the sounds and sound characteristics stored, and recorded within the environmentally responsive headset.

[0059] Referring now to Table 1, by way of example and not of limitation, lists a series of commands which are preferably implemented within the headset. The user is preferably capable of selecting all operating parameters of the selective acoustical isolation apparatus through the user interface. The method of powering the device may be selectable depending on the specific form of headset being utilized, for example, power may in some cases be drawn from a wired connection that has a bias voltage, or from a rectified portion of the signal. In addition, the mode of activation and shutdown, for conserving battery power should be selectable. The mode of selective acoustical isolation to be provided by the headset apparatus as well as the sounds can preferably be programmed. A set of default characteristics which are to apply to the sounds which are to be heard in the environment may be selected. These may be overriden for any particular sound, however, the specification of a default reduces unnecessary selections by the user. Similarly, default characteristics for how to treat sounds that are to be blocked. It will be appreciated that sounds to be blocked may be ongoing "events", wherein the selective acoustical isolation device continues to apply the attenuation to the given sound. The headset preferably is received by the user with a wide variety of pre-programmed sound characterizations, such as a telephone ring, smoke alarm siren, the spoken word "Hello", and so forth. These sound characterizations are configured to provide matching with a variety of devices that need not produce the exact same sound. For example, it will be appreciated that telephone ringing cadence in the united states is fixed, wherein the discernment of ringing may be generally provided by a preprogrammed sound selection criterion. The user preferably can select which of these they wish to trigger a reduction in acoustical isolation. Similarly the user can select from preprogrammed sounds they wish to block, such as a car burglar alarm. The user may also record sounds that are specific to their environment which are to be used to modify the isolation provided by the headset. Prior to programming a sound that the user wants to be alerted to, the user may enter information about how to process the sound. For example, the user can select how the headset is to respond to the given sound, for instance should hearthrough be activated so that they may listen to the external environment, should the detected sound be repeated (echo mode), should an alert be generated. To improve correlation, the type of sound to be correlated can be selected by the user prior to recording. By selecting the type, the apparatus can store the proper sound characteristics to provide adequate correlation. For example sound characterization data stored for a spoken word is different than characterization data stored for a sound, or an alarm sound. The user can preferably select the required accuracy necessary for the unit to consider the external sound correlated with the stored sound. The user can also preferably select what is to be done with the audio source being received. The headset may be utilized for listening to music or for communication, wherein the headset is receiving signals that are being converted to audio by the audio conversion devices within the earpieces. The user can select if these signals are to be attenuated in response to the detection of a hearthrough match, for example, none (no attenuation), block (completely block), −10 dB (attenuate by 10 dB), −20 dB (attenuate by 20 dB), −40 dB (attenuate by 40 dB). The duration of hearthrough can preferably be selected by the user, such as a duration in seconds. The relative volume to which hearthrough is to be configured is preferable set by the user. The user may then select to record a given sound, or to perform an audio "download" of sounds from an audio player device. User selection with regard to blocking selected sounds are similar in nature to the above although more limited in scope. It should be readily appreciated that the aforementioned user options are described by way of example, wherein a particular environmentally responsive headset that provides selective acoustic isolation may provide a different set of options without departing from the teachings of the present invention.

[0060] **FIG. 5** is a flowchart of the method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics and changing the mode of the headset accordingly. The flowchart exemplifies a processor main loop, wherein command processing is considered to be in a separate module whose execution is in response to a processor interrupt. The electronics for the environmentally responsive headset are initialized for proper operation at block **100**. This may take place after power up, or in response to the headsets being in a state of operation, such as positioned on the head of a user. The sound correlation method described is similar to that utilized in performing text string searches, in that the first character is compared for a match, and upon matching the first character, each subsequent character is checked until either a mismatch is found, or the search string is fully matched, at which time a match is returned. In the flowchart, the sound or characteristics thereof are matched a piece at a time. Preferably, distinctive sound elements, or characteristics are searched for thereby reducing the amount of processor overhead. The 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. A sound bite is retrieved from the buffer at block **102**. It will be appreciated that the sound bite may comprise waveforms, waveform characteristics, or characteristics of sound, depending on the algorithms that are being utilized for providing correlation. A pointer to the correlation sound, or trigger condition, is advanced at block **104** and the sound bite is compared at block **106**. 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. The pointer to the correlation sound is reset at block **112** when all correlation sounds have been checked against the sound bite. 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**. A determination is made at block **116** as to whether the sound detected in the environment matches the correlation sound, or characteristics. If the sound is found not to match the correlation sound, then execution returns to processing the sound bite with other correlation sounds, as shown at block **102**. 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. For example, if the correlation sound was set for a sound that the user wished to be alerted to then 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. The audio volume of programmed sounds being output from the audio transducers is preferably attenuated to a given decibel level in relation to the sound detected, so that the wearer can easily hear the external sounds. In addition, the wearer may be alerted with additional warning tones, or a playback of the sound taken from the environment or the correlation sound. If the mode change is a temporary one, such as to allow the wearer to hear the external environment for a period of time, then a delay is checked for completion at block **120**. At the conclusion of the delay the mode of the headset may be restored to the original mode at block **122**. It will be appreciated that the preceding flowchart is a simplified description of the sound discrimination program that will allow one of ordinary skill in programming sound discrimination systems to code the program.

[0061] It will be appreciated that the invention can be implemented in a variety of ways for a number of applications without departing from the inventive teachings. One somewhat unique example involves the configuration of the headphones for use by infantrymen on the battlefield. The unit can be integrated into a helmet, or worn as a headset. The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while sounds such as voices, troop movements and so forth are amplified to provide a hearing advantage to the solder. The headset itself would preferably be configured with a radio so that each infantryman may be retained in communication contact, such as with the squad leader. The 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. Upon detection of selected sounds in the external environment the incoming communication is blocked or attenuated and the external sounds are amplified to improve the safety of the infantryman. In addition, at the time of sound correlation, a radio signal may be forwarded to the squad leader, or otherwise utilized to signal the external event. To enhance the device for use within this application, the mechanical housing which surrounds the microphones should be preferably configured to direct sound in a similar pattern to the human ear, although with perhaps a larger surface area, wherein the wearer could still discern directional characteristics of ambient sounds in a conventional manner. A similar utilization exist for a combat pilot, wherein the headset, typically with the earpieces incorporated into the flight helmet, provides for remote communication, the accentuation of critical audio, and the blocking of select noises, such as the noise of the particular engine. A second example would provide coaches, such as football, with a communication headset that allows them to still hear and communicate with players, while blocking crowd noise. Examples of use abound for those in high ambient noise environments, and those wearing communication headsets, such as pilots. In addition, it will be appreciated that disc-jockeys may find that headsets according to the invention provide the benefits of a studio quality sound reproduction headset while not preventing them from hearing necessary studio sounds, such as telephones, alerts, co-workers, and so forth. Perhaps the most common use may be for those persons wearing headphones to listen to music at work or at home, wherein they will be provided with a studio quality sound experience without the need of worrying about missing telephone calls, alarms, doorbells, or the utterances of another individual such as a spouse. Furthermore, the wearer can listen to the external environment, as triggered by sounds and speech, or as triggered by activating a hearthrough option, such as by pressing a button. The wearer is able to listen to the environment in these instances without the necessity of removing the headset from their head, or turning down the music.

[0062] It will be appreciated that the sound selective acoustical isolation system and method may be less preferably adapted for use within electronic and mechanical equipment that produces a high decibel sound output. It will be appreciated that workers in a high noise environment are essentially isolated from various forms of audio communication, such as spoken words, alarm sounds, and sounds which indicate a dangerous condition due to the high intensity of the noise produced by the equipment. The isolation danger is further exacerbated by the wearing of conventional noise protection headsets, the limitations of which have already been described, because the workers are further isolated from sounds associated with a dangerous or safety related issue. The safety of these workers is therefore compromised due to their isolation from the external environment. The present invention may be substantially incorporated within various forms of controllable sound producing equipment to reduce the risk to those persons working nearby. The microphone correlates a stored sound, or set of sound characteristics, such as screams, a siren, a fire alarm, and so forth, with sounds in the environment. After detecting a match the system changes the "acoustic isolation" afforded nearby individuals by causing the noise-producing device to enter a reduced-noise mode, such as by turning off the device, slowing down the speed of the device, or muting audio sound production. It will be appreciated, therefore, that use of a modified version of the present invention within these noisy environments can increase safety and convenience.

[0063] Accordingly, it will be seen that this invention can be employed within various equipment, particularly headsets, for a variety of applications in acoustic environments. An analog headset embodiment has been shown and described, however, it will be recognized that digital headsets, which may provide class D output to the audio conversion elements, such as speakers, are anticipated within the invention. Specifically utilizing the invention with a digital input source simplifies the system as audio input is routed through the DSP/microcontroller on the way to the speakers, and power may be derived easily from the digital signal or a bias level embedded within that signal, as well as from battery sources. It will be appreciated that the circuits described for implementing the environmentally responsive headsets are but a single design embodiment to carry out the inventive teachings described, and that one of ordinary skill in the art can render innumerable variations without creative effort.

[0064] Although the description above contains many specificities, these should not be construed as limiting the scope of the invention but as merely providing illustrations

of some of the presently preferred embodiments of this invention. Thus the scope of this invention should be determined by the appended claims and their legal equivalents. Therefore, it will be appreciated that the scope of the present invention fully encompasses other embodiments which may become obvious to those skilled in the art, and that the scope of the present invention is accordingly to be limited by nothing other than the appended claims, in which reference to an element in the singular is not intended to mean "one and only one" unless explicitly so stated, but rather "one or more." All structural, chemical, and functional equivalents to the elements of the above-described preferred embodiment that are known to those of ordinary skill in the art are expressly incorporated herein by reference and are intended to be encompassed by the present claims. Moreover, it is not necessary for a device or method to address each and every problem sought to be solved by the present invention, for it to be encompassed by the present claims. Furthermore, no element, component, or method step in the present disclosure is intended to be dedicated to the public regardless of whether the element, component, or method step is explicitly recited in the claims. No claim element herein is to be construed under the provisions of 35 U.S.C. 112, sixth paragraph, unless the element is expressly recited using the phrase "means for."

TABLE 1

Preferred Command List

Setup-configure use
- Options for: Power-off, Display use, Hear button duration, etc.
- Alert settings-tone, volume, duration, etc.

Default, Hear-Default settings for sounds to be heard
Default, Block-Default settings for sounds to attenuate
Sounds Prerecorded, Hear: select any from an audio list
- (i.e. phone, siren, scream, doorbell, Hello, danger sounds . . .)

Sounds Prerecorded, Block: select any from an audio list
- (i.e. car alarm . . .)

Sounds 1 – n:
- Mode: Through/Block
  - IF Through:
    - Hear by: Through, Repeat, Alert (check box)
    - Sound type: Spoken/Sound/Sound repetitive/Alarm (select)
    - Accuracy: required fit of sound (1 – n)
    - Amplification: (1 – n)
    - Audio source: NC, Block, –10 dB, –20 dB, –40 dB
    - Duration: time for pass through (1 – n)
    - Volume: (1 – n)
    - Record (start recording)
    - Download sound/algorithm (start recording)
  - IF Block:
    - Accuracy: required fit of sound (1 – n)
    - Attenuation: (1 – n)
    - Record (start sound recording)
    - Download sound/algorithm (start recording)

What is claimed is:

1. A headset for providing selective acoustical isolation to the wearer, comprising:

(a) an earpiece capable of being positioned proximal the ear of a person wearing said apparatus;

(b) an audio conversion device within the earpiece that is positioned so as to be acoustically coupled to the ear canal of said wearer, said audio conversion device configured to receive electrical energy and convert it to sound;

(c) a microphone attached to said earpiece and configured to register the energy of sounds which occur external to the earpiece and to convert the sound energy to an external sound signal;

(d) a selection device attached to said headset capable of being activated by the headset wearer; and

(e) a signal conditioning circuit configured to sufficiently amplify and couple the external sound signal to the audio conversion device in response to activation of the selection device such that the headset enters a hearthrough mode which improves the ability of the wearer to hear external sounds.

2. A headset as recited in claim 1, further comprising a timing mechanism within the headset that automatically deactivates the selection device a predetermined period of time after activation.

3. A headset as recited in claim 1, wherein the selection device comprises a switch activated by contact by the wearer.

4. A headset as recited in claim 1, wherein the headset is configured to receive input signals which are coupled to the audio conversion device, such that the wearer may listen to programming such as music and audio communications that are received within the input signals.

5. A headset as recited in claim 4, wherein the signal conditioning circuit is further configured to attenuate the input signals upon the activation of the selection device so as to improve the ability of the wearer to hear the external environment.

6. A headset as recited in claim 5, wherein the level of attenuation comprises a muting of the input source by at least 20 dB.

7. A headset as recited in claim 1, wherein the signal conditioning circuit comprises a signal processing unit which is further configured to compare the sounds registered by the microphone against a set of stored sound selection criterion, whereupon while the selection device remains inactive and in response to the signal processing circuit discerning a sufficient level of sound matching it conditions and couples the sound registered by the microphone to the audio conversion device within the earpiece such that the ability of the wearer to hear external sounds is responsive to the detection of selected sounds.

8. An apparatus for providing selective acoustical isolation to the wearer, comprising:

(a) an earpiece configured for positioning proximal the ear of a person wearing said apparatus and as a result providing a given number of decibels of acoustical isolation from the external environment;

(b) an audio conversion device within the earpiece that is positioned so as to be acoustically coupled to an ear canal of said wearer, said audio conversion device configured to receive an electrical signal which is converted to sound;

(c) a microphone attached to said earpiece that is configured to register the acoustical energy of external environmental sounds which are converted to an external sound signal; and

(d) a signal processing circuit configured to receive the external sound signal from said microphone which is compared against a set of programmed sound selection

criterion, whereupon the amount of acoustic isolation provided by the apparatus is changed in response to a sufficient level of sound matching thereof.

9. An apparatus as recited in claim 8, wherein the change in the amount of acoustic isolation occurs for a sufficient period of time to alert the wearer of conditions in the external environment.

10. A headset as recited in claim 8, further comprising a manually operated selection device mounted on the apparatus which upon activation triggers a reduction in the amount of acoustical isolation provided by the apparatus.

11. An apparatus as recited in claim 8, wherein said audio conversion device is configured for receipt and conversion of a received program signal, such as music or communications, to audio for listening by the wearer.

12. An apparatus as recited in claim 8, wherein the signal processing circuit is configured to record sound characterizations as programmed sound selection criterion upon user activation of characterization recording in approximate temporal synchronization with sounds from the external environment whose sound characteristics are to be programmed.

13. An apparatus as recited in claim 8, wherein the signal processing circuit is configured to allow the user to select programmed sound selection criterion from a preprogrammed set of sound selection criterion.

14. An apparatus as recited in claim 8, wherein the set of programmed sounds comprise characterization information about sounds, referred to as hearthrough sounds, which the listener should be alerted to in response to their occurrence within the external environment, wherein in response to a sufficient level of sound matching between the sounds from the external environment and the hearthrough sounds the acoustical isolation of the apparatus is decreased by applying the external sound signal registered by the microphone to the audio conversion device to reproduce external sounds within the earpiece.

15. An apparatus as recited in claim 14, wherein said audio conversion device is configured for receipt and conversion of a received program signal, such as music or communications, with said program signal being attenuated in response to sufficient matching of the hearthrough sound with sounds registered from the external environment such that the ability of the wearer to listen to the external environment is improved.

16. An apparatus as recited in claim 8, wherein the signal processing circuit is further configured to generate an alerting signal to the audio conversion device to signal the wearer that a sufficient sound match has occurred.

17. An apparatus as recited in claim 8, wherein the signal processing circuit is configured with noise cancelation circuitry to enhance the earpiece acoustical isolation by modifying the output from the audio conversion device with a negative feedback, opposing phase, signal based on sounds registered by the microphone.

18. An apparatus as recited in claim 17, wherein the set of programmed sounds comprise characterization information about sounds, referred to as hearthrough sounds, which the listener should be alerted to in response to their occurrence within the external environment, and wherein the signal processing circuit is configured to reduce the acoustic isolation provided by the noise cancelation circuitry after a hearthrough sound is sufficient matched with a sound from the external environment so as to improve the ability of the wearer to hear the external environment.

19. An apparatus as recited in claim 17, wherein the noise cancelation circuitry is configured to selectively increase acoustical isolation provided by the headset of sound characterization elements contained within the set of programmed sound selection criterion that specify sounds which are to be blocked from the external environment, referred to as a blocked sound, wherein the noise cancelation circuitry specifically enhances its active attenuation of blocked sounds as received from the external environment.

20. A method of providing selective acoustic isolation of a headset wearer from the external acoustic environment, comprising the steps of:

(a) acoustic monitoring of the external environment so that the acoustics are registered by a microphone and converted to electrical signals received by a signal processing circuit;

(b) comparison of the monitored acoustics from the microphones by the signal processing circuit with a stored set of acoustic sound parameters; and

(c) converting the monitored acoustics registered by the microphones into acoustic energy within the headset in response to matches of the monitored acoustics with elements in the stored set of acoustic sound parameters such that the wearer of the headset is made aware of selected external acoustics which are associated to the set of stored acoustic parameters.

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Exhibit PA-I

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(54) **EAR TERMINAL WITH MICROPHONE FOR VOICE PICKUP**

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(58) **Field of Search** 381/328, 313–315, 381/370, 317, 324, 71.6, 312, 316, 318, 320, 319, 322, 323, 71.1, 71.2; 600/25, 539, 23, 24; 340/573.1

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,985,925 A 1/1991 Langberg et al.
5,426,719 A 6/1995 Franks et al.
6,456,199 B1 * 9/2002 Michael 340/573.1

OTHER PUBLICATIONS

Chang et al. ; Hearing Protector; Jan. 6, 1994; WO 94/00089.*

* cited by examiner

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(74) *Attorney, Agent, or Firm*—Young & Thompson

(57) **ABSTRACT**

An ear terminal includes a sealing section arranged for use in the ear meatus of a human, with an inner microphone having a sound inlet for being directed into the meatus and an electronic unit including filtering elements coupled to the inner microphone for filtering the signal from the inner microphone, the filtering elements being programmable to transform the signals based on the sounds received in the ear by the inner microphone into sounds having essentially the characteristics of spoken sounds of the wearer of the ear terminal.

20 Claims, 9 Drawing Sheets

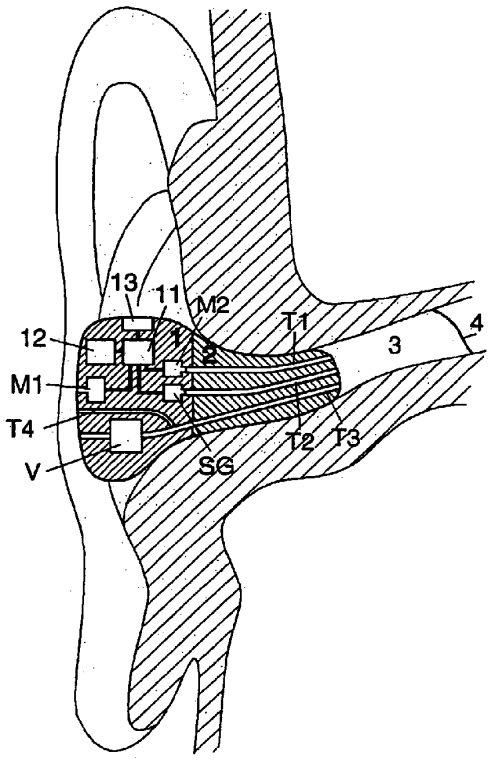


Exhibit PA-I

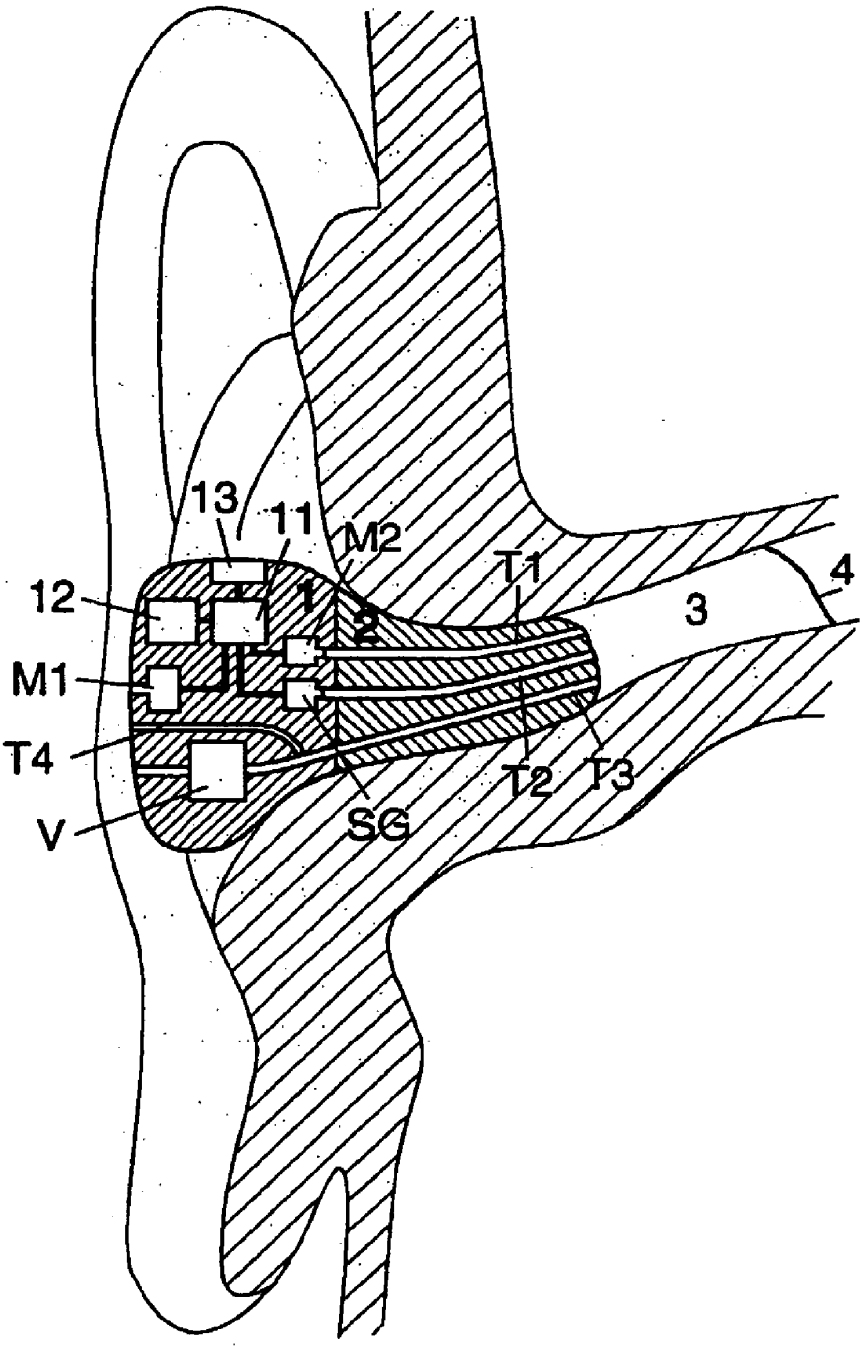


Fig. 1

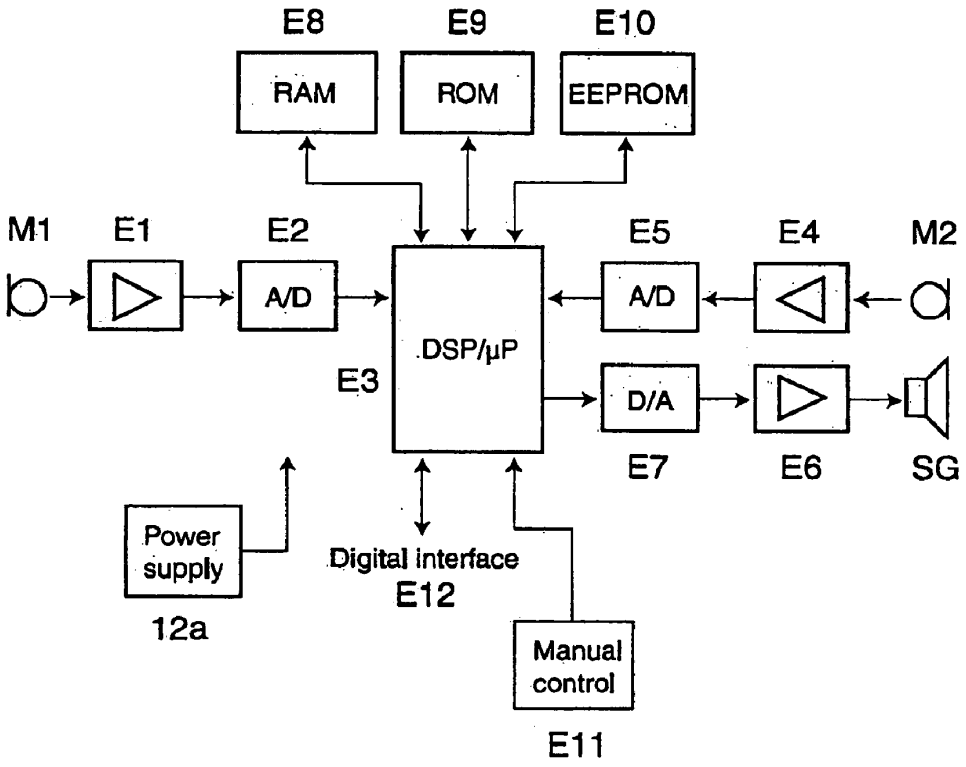

E8
E9
E10
RAM
ROM
EEPROM
M1
E1
E2
E5
E4
M2
A/D
A/D
DSP/µP
E3
D/A
E7
E6
SG
Power supply
12a
Digital interface
E12
Manual control
E11

Fig. 2

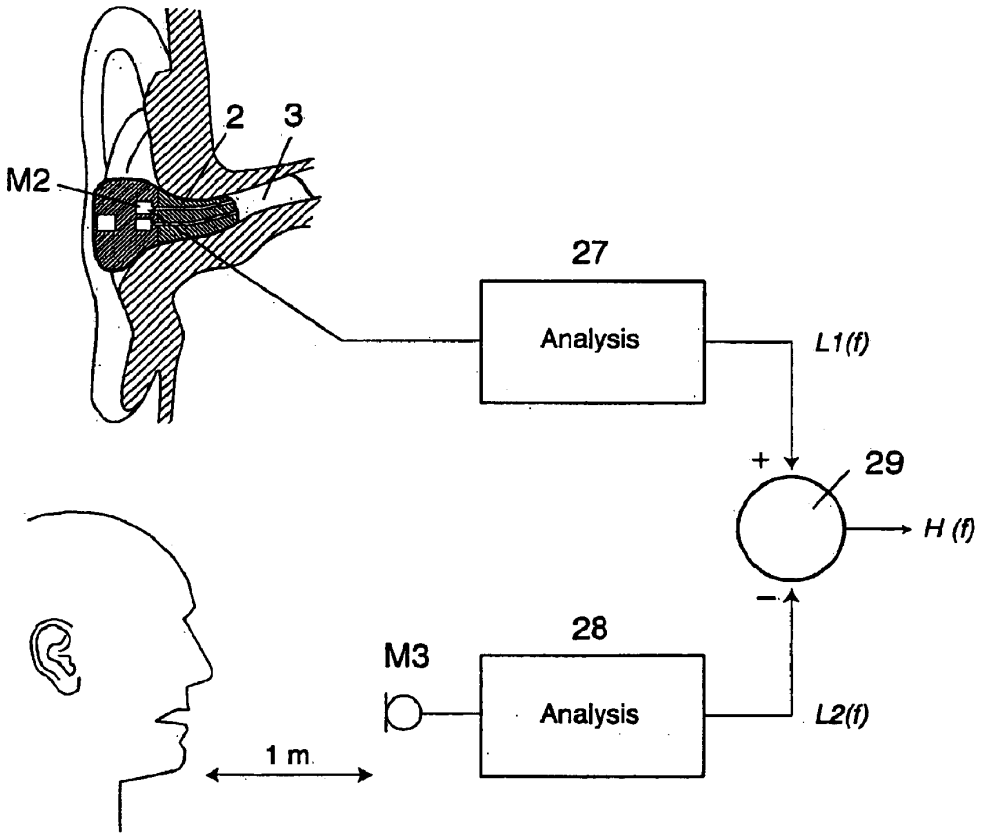


Fig. 3

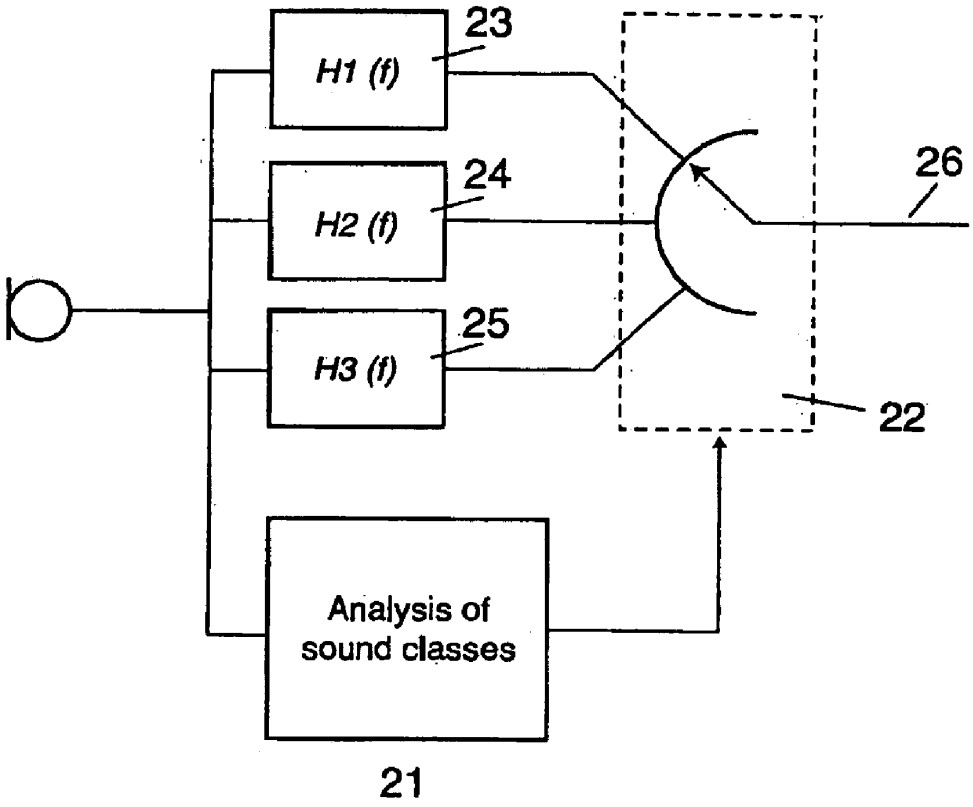


Fig. 4

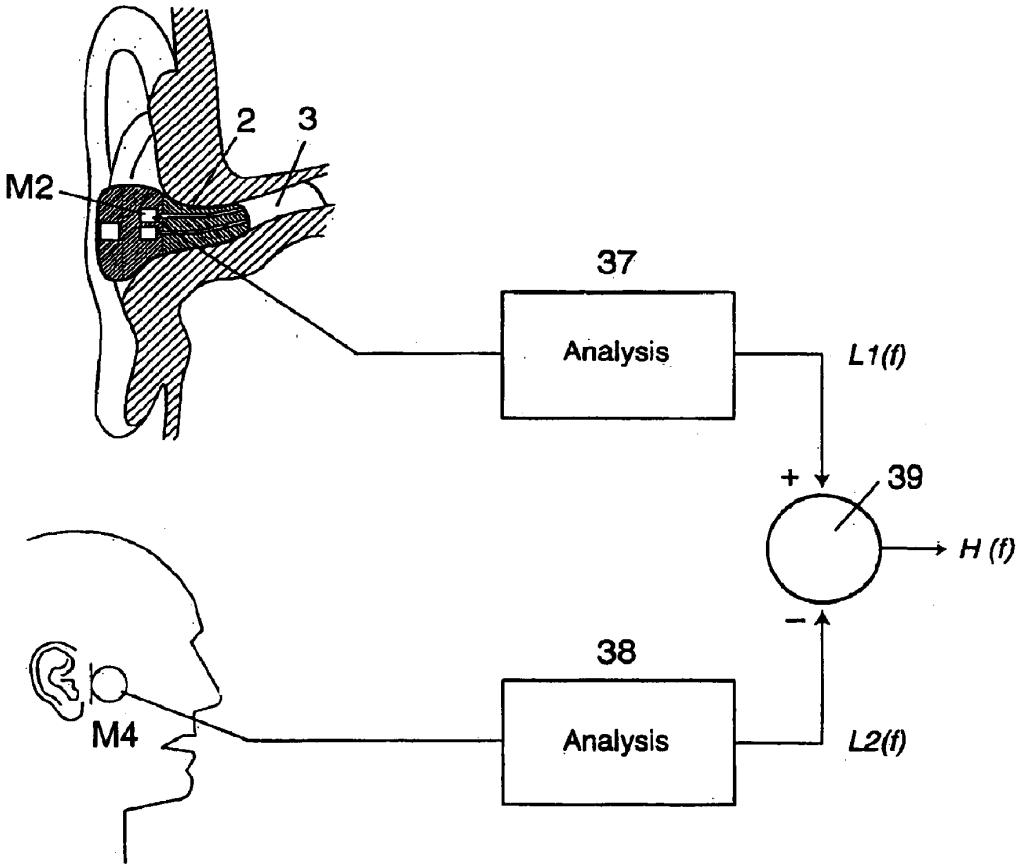


Fig. 5

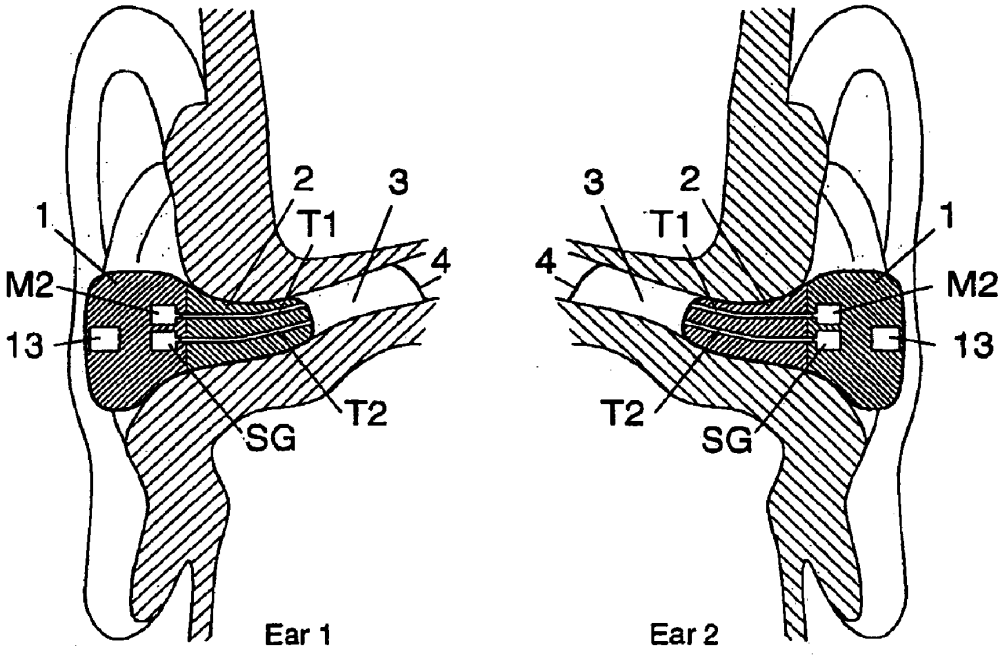


Fig. 6

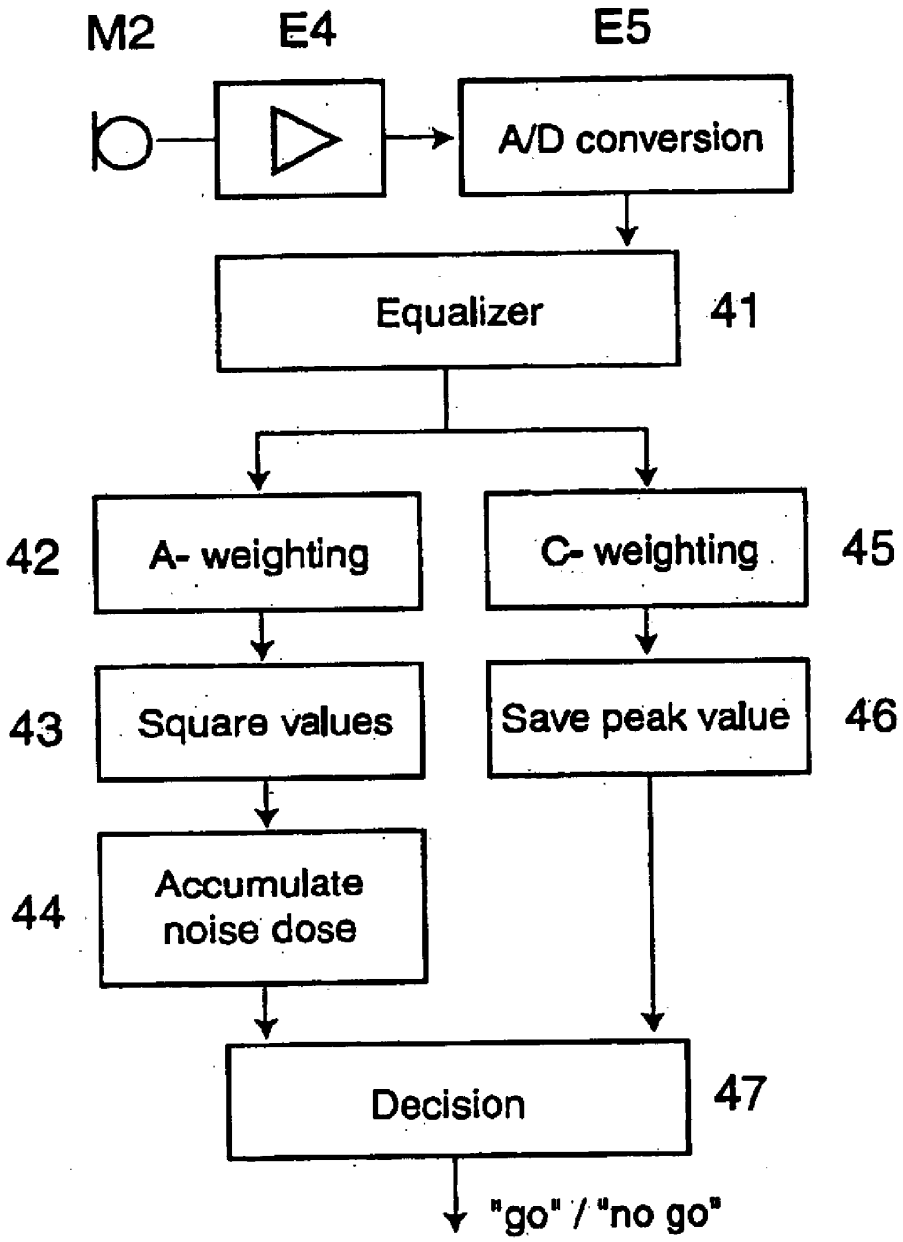

M2
E4
E5
A/D conversion
Equalizer
41
42
A- weighting
C- weighting
45
43
Square values
Save peak value
46
44
Accumulate noise dose
Decision
47
"go" / "no go"

Fig. 7

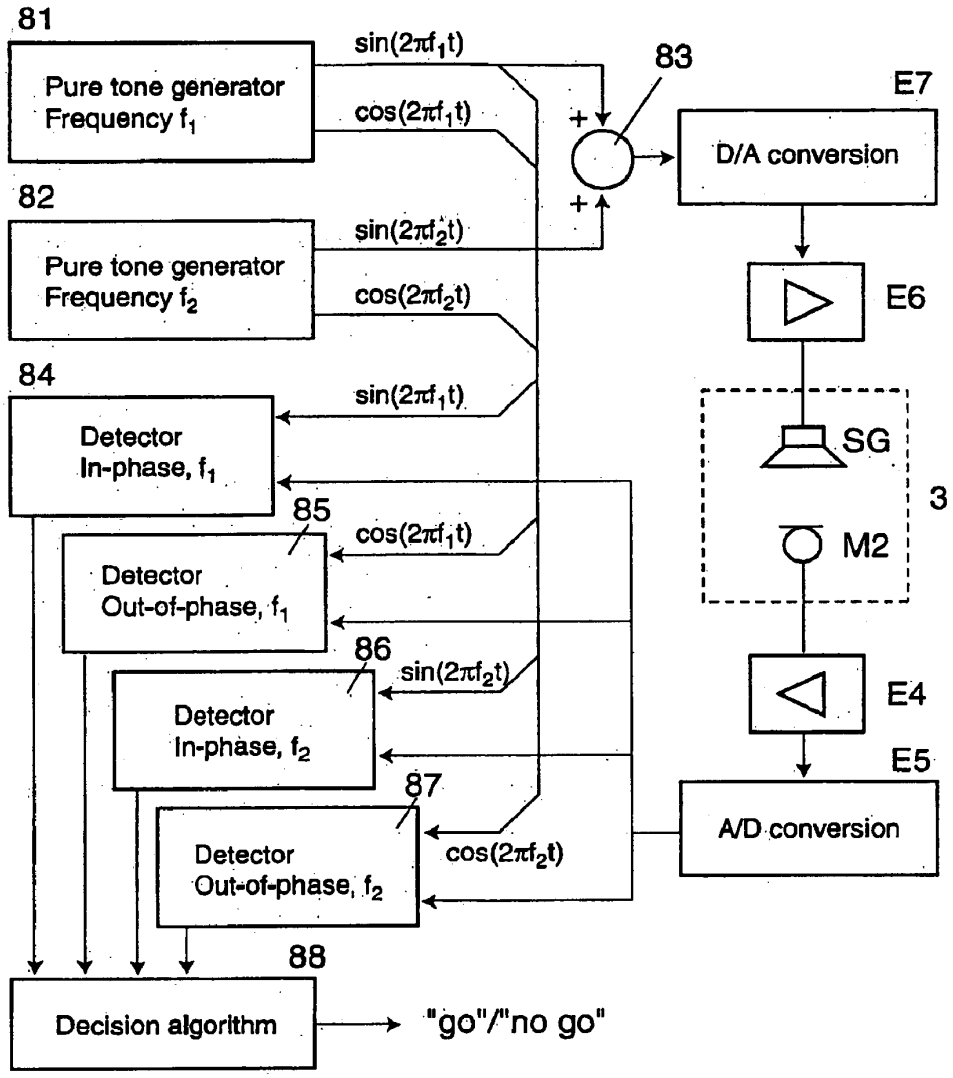

81
Pure tone generator
Frequency f_1
$\sin(2\pi f_1 t)$
$\cos(2\pi f_1 t)$
83
+
+
E7
D/A conversion
82
Pure tone generator
Frequency f_2
$\sin(2\pi f_2 t)$
$\cos(2\pi f_2 t)$
E6
84
Detector
In-phase, f_1
$\sin(2\pi f_1 t)$
SG
3
M2
85
Detector
Out-of-phase, f_1
$\cos(2\pi f_1 t)$
86
Detector
In-phase, f_2
$\sin(2\pi f_2 t)$
E4
E5
87
Detector
Out-of-phase, f_2
$\cos(2\pi f_2 t)$
A/D conversion
88
Decision algorithm
"go"/"no go"

Fig. 8

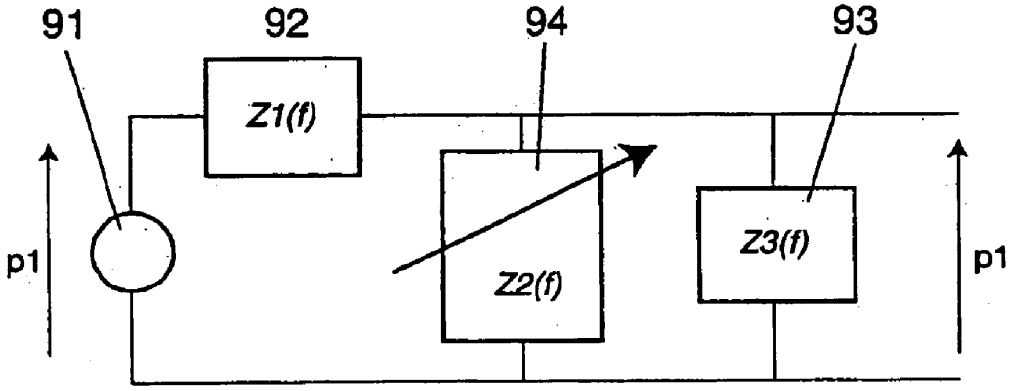


Fig. 9

EAR TERMINAL WITH MICROPHONE FOR VOICE PICKUP

BACKGROUND OF THE INVENTION

The invention concerns the physical design of an adaptive hearing protective earplug combined with an audio communications terminal.

DESCRIPTION OF THE RELATED ART

There exist a lot of solutions for hearing protection and audio communication in noisy environments based on earplugs and ear-muffs with earphones (loudspeakers), boom microphones, cheek-bone microphones, or throat microphones. All these solutions have one of more of the following undesirable properties:

heavy and clumsy.

uncomfortable.

inferior quality of sound pick-up and restoration.

poor noise attenuation.

attenuate both desired and unwanted sounds.

SUMMARY OF THE INVENTION

It is an object of this invention to provide an ear terminal having none of these shortcomings, being a lightweight, all-in-the-ear intelligent hearing protector with wireless communication. The noise attenuation is automatically adapted to the noise conditions and communication modes. The present invention therefore simultaneously protects the hearing and provides improved communication abilities in different noise environments. It is intended for continuous use during the working day or other periods when hearing protection and/or voice communication is needed.

The invention also concerns a device for utilising the speech sound produced in the ear of a person carrying hearing protective communications ear plugs according to the invention.

Present day devices intended to pick up speech from a person in a very noisy environment represent a technological challenge and take several forms. Common types include

- A microphone in close proximity to the mouth, carried on a microphone boom. The microphone is made with a characteristic emphasising the near field from the mouth. This type is sometimes referred to as "noise cancelling".
- A vibration pickup in contact with the throat, picking up the vibrations of the vocal cord.
- A vibration pickup in contact with the wall of the meatus, the outer ear canal, picking up the vibrations of the tissue in the head.
- A similar pickup in contact with the cheek-bone.

These device types are either fairly sensitive to acoustic noise masking the speech, or certain speech sounds are poorly transmitted, especially the high frequency consonant sounds necessary for good intelligibility.

Persons exposed to high noise levels are required by health and safety regulations to wear hearing protectors. The protectors take the form either of sealing cups which enclose the ear, or ear plugs which block the ear canal. The latter type of protector is often preferred because of its small size and relatively good comfort.

Thus it is an additional object of this invention to provide an ear plug with two desirable properties:

- The cavity sealed off in the inner portion of the meatus by the ear plug is relatively free of external noise, this is the purpose of the ear plug in protecting the hearing.

- The sound field in the cavity generated by the persons own voice contains all the frequency components necessary to reconstruct the speech with good intelligibility.

The solution according to invention takes advantage of these facts. By using a microphone to pick up the acoustic sound field in the inner portion of the meatus and processing the microphone signal according to the invention, a speech signal of high quality and low noise masking is produced.

It is an additional object of this invention to provide a system for increasing the user's feeling of naturalness of the user's own voice when using a hearing protective communications terminal according to the invention.

Using ordinary earplugs or earmuffs, the user usually feels his own voice being distorted, a feature reducing the comfort of wearing hearing protectors. Ordinary hearing defenders changes the normal sound transmission path from the mouth to the eardrums. Thus the auditory feedback from the users own voice is affected resulting in an unintended change the speech output. A normal response is to raise one's own voice level when using headsets or earplugs.

The invention solves this problem by filtering and mixing in the user's own voice picked up by either the outer or the inner microphone at one ear and reproduce the signal at the loudspeaker in the other ear. It is also possible to reproduce the signal by the loudspeaker in the same ear, in which case feedback cancellation has to be applied. Thus the user's voice is felt more natural both with respect to frequency response and speech level. This feature will increase the level of acceptance for continuous use of hearing protectors during the whole working day. The own voice signal is added and reproduced in such a way that the noise reduction property of the hearing protector is maintained.

An additional object of this invention is to provide a programmable personal noise exposure dose meter that measures the true exposure in the user's ear and calculates the hearing damage risk.

Present day noise exposure dose meters, also called dosimeters, usually consist of a microphone and a small electronics unit that may be attached to the body or worn in a pocket. The microphone may be mounted on the electronics unit or it may be fastened to the collar or on the shoulder. ANSI S1.25 specifies dosimeters.

Present day dosimeters have several shortcomings:

- Dosimeters do not measure the noise that actually affects the hearing organ (e.g. when the user wears a hearing protector, helmet, etc.). Even when the ear is not covered, measurements may be influenced by body shielding.
- Dosimeters are susceptible to non-intentional or intentional errors, which may influence readings, such as wearers tapping or singing into dosimeter microphones or by wind-generated noise.
- Dosimeters are inaccurate if impulse or impact noise is present.

The invention solves these problems by using a microphone that measures the sound at the eardrum and employs analysis procedures that take into account both stationary and impulsive sound. When the dose meter is part of a communications terminal this includes external noises, incoming communication signal, as well as possible malfunctioning of the equipment.

It is also an object of this invention to provide a device for verifying in situ that a hearing protector is properly used.

Present day hearing protectors take the form either of sealing cups which enclose the ear, or ear plugs which blocks the ear canal. For both types, it is critically important

to avoid leakage of the noise sound through or around the sealing and blocking parts of the hearing protectors.

Experience shows that several factors may compromise the sealing of a hearing protector and thereby increase the risk of hearing damage. These factors include

- Irregular surfaces which the sealing material is not able to follow properly. Examples are spctacles used with ear cups, and ear plugs used by persons with irregularly formed ear canals.
- Improper placement of the hearing protector. Experience and patience is required by the user to get a hearing protector mounted correctly. In cases where the user is wearing a helmet or cap, the hearing protector may be accidentally pushed out of position during use.
- Ageing of the materials in the sealing may reduce the resilience of the sealing and thereby allow leakage around the sealing.

The result of leakage is reduced damping of potentially harmful noise. Ideally, the leakage should be detected and remedied prior to noise exposure. The leakage may not be clearly audible. Accordingly, noise situations may comprise of intermittent or impulsive components which may damage the hearing almost instantaneously if a hearing protector should be malfunctioning or imperfect without the user's knowledge.

The invention solves these problems by an in situ acoustical measurement, which is analysed and reported to the user in audible form, or to external equipment by means of communication signals. The devices necessary for the measurement are an integral part of the hearing protector. Verification may be activated by the user at any time, or be continuously running when the application is critical. Optionally, verification may be activated by other persons (or devices) than the user, e.g. to verify hearing protector function before admittance to a noisy area is allowed.

The above mentioned problems are solved by the invention according to the attached claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be described below with reference to the accompanying drawings, the drawings illustrating the invention by way of examples.

FIG. **1** is a simplified vertical section along the central axis of the meatus of the outer ear of an erect human, with an inserted ear terminal according to one embodiment of the invention also shown in vertical section along the axis, locally coincident with the meatus axis.

FIG. **2** is an electrical wiring diagram showing the functional components and connections between electronic components in a preferred embodiment according to the invention.

FIG. **3** is an illustration of one method according to the invention, showing that spectral analysis of sound picked up in the ear is compared with spectral analysis of sound picked up by a microphone at a standard distance, e.g. of 1 meter, under otherwise quiet conditions.

FIG. **4** is an illustration of speech sound analysis and following sound source classification with filtering conducted according to the sound source classification, according to one embodiment of the invention.

FIG. **5** is an illustration of another method according to the invention, illustrating an analysis of sound picked up close to the ear being compared with an analysis of sound picked up by a microphone arranged in the meatus.

FIG. **6** illustrates a simplified section through a human's right and left ears with ear terminals according to the invention illustrated for improved natural sound purposes.

FIG. **7** illustrates a process diagramme for one embodiment of the invention concerning noise dose metering, here illustrating an A-weighting with accumulated noise dose measurements, and also with C-weighting for peak noise value registration.

FIG. **8** illustrates another embodiment of the invention illustrating a processing scheme for online verification of hearing protector performance.

FIG. **9** illustrates an electric analogy diagram of the acoustic phenomenon on which an embodiment for online verification of hearing protector performance is based.

DESCRIPTION OF PREFERRED EMBODIMENTS OF THE INVENTION

The physical design of an embodiment of the present invention enables the construction of a complete all-in-the-ear hearing protector and communications terminal with strong passive sound attenuation, strong active sound attenuation, high quality sound restoration, high quality sound pick-up, small size, low weight, and comfortable fit.

One embodiment of this invention is illustrated in FIG. **1**, and provides the general physical design of a complete all-in-the-ear hearing protector and communications terminal, regarded as a combination of passive sealing, characteristics and placement of electro-acoustic transducers as well as acoustic filters, electric circuitry, and a ventilation system for pressure equalisation.

The ear terminal comprises an outer section **1** arranged for sitting adjacent to the outward facing portion of the sealing section **2** and a part of the inward facing portion of the outer section **1** is formed to fit the concha around the outer portion of the meatus **3**.

The physical design represented by an embodiment of the invention enables some or all of the following functionalities:

- External sounds are attenuated by a combination of passive and active noise control. The passive attenuation is obtained by means of an earplug **1**,**2** with a sealing system **2** inserted in the outer part of the ear canal or meatus **3**. The active noise control is achieved by using one or two microphones **M1**, **M2** and a loudspeaker SG together with electronic circuits in an electronics unit **11** mounted in the earplug system. The algoritmes for noise control are per se known and will not be described in any detail here, but may include active noise cancelling by feedback of acoustic signals converted by at least one of said microphones (**M1**,**M2**) throught the sound generator (SG).
- Restoration of desired sounds (external sounds and signals from the communication system) at the eardrum or tympanum **4** is achieved by using the same microphones **M1**, **M2**, and loudspeaker SG and the electronics unit **11**. Again, the algoritmes for obtaining this are per se known and will not be described in any detail here, but may include amplification of chosen frequencies converted by said the microphone (**M1**) and generating a corresponding acoustic signal through said sound generator (SG). The frequencies may for example be within the normal range of the human voice.
- Ear terminal according to claim **1**, comprising a sound generator (SG) arranged for being directed toward the meatus and being coupled to said electronics unit (**11**), wherein the electronics unit (**11**) comprises filtering means for active sound transmission e.g. by amplifica-

tion of chosen frequencies converted by said outer microphone (M1) and generating a corresponding acoustic signal through said sound generator (SG).

Pick-up of the user's voice is performed by a microphone M2 with access to the closed space in the meatus 3. This signal is processed by means of analogue or digital electronics in the electronics unit 11 to make it highly natural and intelligible, either for the user himself or his communication partners or both parts. This signal is of high quality and well suited for voice control and speech recognition.

Online control and verification of the hearing protector performance is obtained by injecting an acoustic measurement signal, preferrably by the sound generator or loudspeaker SG in the meatus, and analysing the signal picked up by the microphone M2 that has access to the acoustic signal in the meatus 3.

Measurement of noise exposure dose at the tympanum 4 and online calculation performed by electronic circuits, and warning of hearing damage risk either by audible or other warning signals, either to the wearer of the hearing protection or other relevant personnel.

Equalisation of pressure between the two sides of the earplug system is obtained by using a very thin duct T3,T4 or a valve that equalises static pressure differences, while retaining strong low frequency sound attenuation. A safety valve V to take care of rapid decompression may be incorporated in the pressure equalisation system T3,T4.

FIG. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones M1 and M2 and a sound generator SG. The main section is designed in a way that provides comfortable and secure placement in the concha (the bowl-shaped cavity at the entrance of the ear canal). This may be obtained by using individually moulded ear-pieces that are held in position by the outer ear or by using a flexible surrounding pressing against the structure of the outer ear. A sealing section 2 is attached to the main section. The sealing section may be an integral part of the earplug, or it may be interchangeable. The sound inlet of microphone M1 is connected to the outside of the earplug, picking up the external sounds. The microphone M2 is connected to the inner portion of the meatus 3 by means of an acoustic transmission channel T1. The acoustic transmission channel may contain optional additional acoustic filtering elements. An outlet S_{SG} of sound generator SG is open into the inner portion of the meatus 3 by means of an acoustic transmission channel T2 between the sound generator SG and the inward facing portion of the sealing section 2. The acoustic transmission channel T2 may contain optional additional acoustic filtering elements.

When smaller microphones M2, and sound generators SG are available, it will be possible to mount the microphone M2 and the sound generator SG at the innermost part of the sealing section. Then there is no need for the transmission channels T1 and T2.

The two microphones and the sound generator are connected to an electronics unit 11, which may be connected to other equipment by a connection interface 13 that may transmit digital or analogue signals, or both, and optionally power.

Electronics and a power supply 12, e.g. a battery, may be included in main section 1 or in a separate section.

The microphones M1,M2 may in a preferred embodiment be standard miniature electret microphones like the ones used in hearing aids. Recently developed silicon microphones may also be used.

The sound generator SG may in a preferred embodiment be based on the electromagnetic or electrodynamic principle, like sound generators applied in hearing aids.

According to a preferred embodiment of the invention, a safety valve V is incorporated in the ventilation duct comprising the channels T3 and T4. The valve V is arranged to open if the static pressure in the inner part of the meatus 3 exceeds the outside pressure by a predetermined amount, allowing for pressure equalisation during rapid decompression. Such decompression may occur for military or civilian air personnel experiencing rapid loss of external air pressure. Such a decompression may also occur for parachuters, divers, and the like. Pressure equalisation for slowly varying pressure changes is obtained by using a narrow vent T4 which may bypass the valve V. A proper design of this vent T4 allows for static pressure equalisation without sacrificing low-frequency noise attenuation.

The main section of the earplug may be made of standard polymer materials that are used for ordinary hearing aids. The sealing part may be made of a resilient, slowly re-expanding shape retaining polymer foam like PVC, PUR or other materials suitable for earplugs.

For some applications (less extreme noise levels) the earplug may be moulded in one piece 1,2 combining the main section 1 and the sealing section 2. The material for this design may be a typical material used for passive earplugs (Elacin, acryl).

It is also possible to make the earplug in one piece comprising the main section 1 and the sealing section 2, all made of a polymer foam mentioned above, but then the channels T1,T2,T3,T4 have to be made of a wall material preventing the channels T1,T2,T3,T4 to collapse when the sealing section 2 is inserted in the meatus 3.

All the features mentioned above may be obtained by an electric circuitry represented by the block diagram in FIG. 2.

The microphone M1 picks up the ambient sound. A signal from the microphone M1 is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and fed to a processing unit E3 that may be a digital signal processor (DSP), a microprocessor (μP) or a combination of both. A signal 51 from microphone M2, which picks up the sound in the meatus 3 between the isolating section 2 and the tympanum 4, is amplified in the amplifier E4 and sampled and digitised in the analogue to digital converter E5 and fed to the processing unit E3.

A desired digital signal DS is generated in the processing unit E3. This signal DS is converted to analogue form in the digital to analogue converter E7 and fed to the analogue output amplifier E6 that drives the loudspeaker SG. The sound signal produced by the loudspeaker SG is fed to the tympanum 4 via the channel T2 into the meatus 3 as described above.

The 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. The memories E8,E9, and E10 are in a preferred embodiment of the invention used for storing computer programs, filter coefficients, analysis data and other relevant data.

The electronic circuitry 11 may be connected to other electrical units by a bi-directional digital interface E12. The communication with other electrical units may be performed via a cable or wireless through a digital radio link. The Bluetooth standard for digital short-range radio (Specification of the Bluetooth System, Version 1.0 B, 1 Dec. 1999, Telefonaktiebolaget LM Ericsson) is one possible candidate for wireless communication for this digital interface E12.

In a preferred embodiment of the invention, signals that may be transmitted through this interface are:

- program code for the processing unit E3
- analysis data from the processing unit E3
- synchronisation data when two ear terminals 1,2 are used in a binaural mode
- digitised audio signals in both directions to and from an ear terminal 1,2.
- control signals for controlling the operation of the ear terminal.
- digital measurement signals for diagnosis of the ear terminal performance.

A manual control signal may be generated in E11 and fed to the processing unit E3. The control signal may be generated by operating buttons, switches, etc, and may be used to turn the unit on and off, to change operation mode, etc. In an alternative embodiment, a predetermined voice signal may constitute control signals to the processing unit E3.

The electric circuitry is powered by the power supply 12*a* that may be a primary or rechargeable battery arranged in the earplug or in a separate unit, or it may be powered via a connection to another equipment, e.g. a communication radio.

One embodiment of the invention concerns the use of the ear terminal as an "in-the-ear voice pick-up". The sound of a person's own voice as heard in the meatus is not identical to the sound of the same person's voice as heard by an external listener. The present embodiment of the invention remedies this problem. The microphone M2 illustrated in FIG. 3 picks up the sound in the inner portion of the meatus 3 sealed off by a sealing section 2 in an ear protecting communications device of the earplug type. The signal is amplified by the amplifier E4 illustrated in FIG. 2, A/D converted by the A/D converter E5, and processed in the digital signal processing (DSP) or microcomputer unit E3. The processing may be viewed as a signal dependent filtering taking into account the speech signal properties as well as computed estimates of the location of sound generation for the different speech sounds. Thereby the speech intelligibility and naturalness may be improved.

FIGS. 1 and 3 show examples of embodiments of the invention, with the microphone M2 being integrated in a hearing protective communications earplug. The acoustic transmission channel T1 connects microphone M2 to the inner portion of the meatus 3. Microphone M2 picks up the sound field produced by the person's own voice. The signal may be amplified in amplifier E4, A/D converted in A/D converter E5 and processed in the digital signal processing (DSP) or microcomputer unit E3. A processed signal from E3 may be transmitted in digital form through a digital interface E12 to other electrical units. In an alternative embodiment, the processed signal from E3 may be D/A converted and transmitted in analogue form to other electrical units.

FIG. 4 illustrates one possible signal processing arrangement according to the invention. It illustrates an example of the type of signal dependent filtering which may be applied to the signal from microphone M2 in order to obtain a good reconstruction of the speech signal, making it highly intelligible, even in extremely noisy environments.

After amplification in E4 and A/D-conversion E5, the microphone M2 signal is analysed in the DSP/uP processing unit E3. The analysis represented by block 21 in FIG. 4 may comprise a short term estimate of the spectral power in the microphone signal, a short term auto-correlation estimate of the microphone signal, or a combination of both. Based on these estimates, a running classification with corresponding decision represented by block 22 may be made in the processing unit E3 for the selection of the most suitable conditioning filter for the signal from microphone M2. In the example shown in FIG. 4, the selection may made between e.g. three filters H1(f), H2(f) and H3(f) represented by blocks 23, 24 and 25, appropriate for vowel sounds, nasal sounds and fricative sounds respectively. The processed signal is present at output 26 of block 22. Other sound classifications using more sophisticated subdivisions between sound classifications and corresponding sound filters and analysis algorithms may be applied. The selection algorithm may comprise gradual transitions between the filter outputs in order to avoid audible artefacts. Filtering and selection is carried out in the processing unit E3 concurrently with the sound analysis and classification.

The basis for the filter characteristics and the corresponding analysis and classification in the processing unit E3 may be derived from an experiment of the form shown in FIG. 3. An ear plug with a microphone M2 with the same properties as the one used for the voice pickup is used to pick up the voice of a test subject from the meatus 3 illustrated in the upper part of FIG. 3. Concurrently, the voice is recorded by a high quality microphone M3 in front of the subject, at a nominal distance of 1 meter, under an-echoic conditions. Estimates of the power spectral densities may be computed for the two signals by the analyses represented by blocks 27 and 28 respectively, and the corresponding levels L1(f) and L2(f) are compared in comparator 29. The output from the comparator is represented by the transfer function H(f). The analyses may be short time spectral estimates, e.g. ⅓ octave spectra in the frequency range 100 Hz to 14000 Hz. The test sequences which the subject utters may comprise speech sounds held constant for approximately 1 second. For voiced sounds, the subject person may make the pitch vary during the analysis period. The transfer functions of the filters described in connection with FIG. 4 may be based on diagrams of H(f), the spectral density levels of the free field microphone M3 subtracted from the corresponding levels of the in-the-ear microphone M2.

A simplest embodiment of the invention may reduce the system in FIG. 4 to one single time invariant filter. The analysis and selection processing may then be omitted. The transfer function of the single filter is still based on diagrams of the spectral density levels of the free field microphone subtracted from the corresponding levels of the in-the-ear microphone, described in connection with FIG. 3. The transfer function may be a combination of the results for the various speech sounds, weighted in accordance with their importance for the intelligibility and naturalness of the processed speech.

Another embodiment of the invention is best understood under the term "Natural Own Voice", indicating that a person wearing an ear terminal shall perceive his own voice as being natural while having the meatus blocked by an earplug.

The inner microphone M2 or the outer microphone M1, or a combination of both, picks up the sound signal representing the users voice signal. The signal is amplified, A/D converted, and analysed in the digital signal processor E3. Based on previously measured transfer functions from the user's speech to the microphone M2 (and/or M1), the microphone signal may be filtered to regain the naturalness of the user's speech. The signal is then D/A-converted, amplified and reproduced at an internal loudspeaker SG. The internal loudspeaker SG may be arranged in a similar ear

terminal **1,2** in the wearer's other ear to prevent local feedback in the earplug. In a more acoustically demanding arrangement the loudspeaker SG, arranged in the same meatus **3** as the inner pickup microphone **M2** is situated, may be used, thus demanding feedback cancellation. The desired signal to the loudspeaker SG in the other ear may be transmitted via electric conductors outside of the wearer's head, or via radio signals.

FIG. **6** shows one preferred embodiment of the invention with the natural own voice feature being integrated in two active hearing protective communications earplugs. Each earplug may comprise a main section **1** containing two microphones, an outer microphone **M1** and an inner microphone **M2**, and a sound generator SG. The right and left earplugs are generally symmetrical, otherwise identical for both ears. Section **2** is the acoustic sealing of the hearing protector. An acoustic transmission channel **T1** connects microphone **M2** to the inner portion of meatus **3**. Microphone **M2** picks up the sound from the meatus **3**. When the user is speaking and the ear canal is sealed, this signal is mainly the user's own voice signal. This signal is filtered and reproduced at the loudspeaker SG at the other ear. An acoustic transmission channel **T2** connects sound generator SG to the inner portion of meatus **3**. A block diagram of the electronic system is shown in FIG. **2**.

FIG. **4** shows an example of the type of signal dependent filtering which may be applied to the microphone signal in order to obtain a good reconstruction of the voice.

After amplification in **E4** and A/D-conversion **E5**, the microphone **M2** signal is analysed in the DSP/uP processing unit **E3**. The analysis represented by block **21** in FIG. **4** may comprise a short term estimate of the spectral power in the microphone signal, a short term auto-correlation estimate of the microphone signal, or a combination of both. Based on these estimates, a running classification with corresponding decision represented by block **22** may be made in the processing unit **E3** for the selection of the most suitable conditioning filter for the signal from microphone **M2**. In the example shown in FIG. **4**, the selection may made between e.g. three filters **H1**(f), **H2**(f) and **H3**(f) represented by blocks **23**, **24** and **25**, appropriate for vowel sounds, nasal sounds and fricative sounds respectively. The processed signal is present at output **26** of block **22**. Other sound classifications using more sophisticated subdivisions between sound classifications and corresponding sound filters and analysis algorithms may be applied. The selection algorithm may comprise gradual transitions between the filter outputs in order to avoid audible artefacts. Filtering and selection is carried out in the processing unit **E3** concurrently with the sound analysis and classification.

The basis for the filter characteristics and the corresponding analysis and classification in the processing unit **E3** may be derived from an experiment of the form shown in FIG. **5**. An ear plug with a microphone **M2** with generally the same properties as the one used for the voice pickup is used to pick up the voice of a test subject from the meatus **3** illustrated in the upper part of FIG. **5**. Concurrently, the voice is recorded by a high quality microphone **M4** close to the subject's ear, under an-echoic conditions. Estimates of the power spectral densities may be computed for the two signals by the analyses represented by blocks **37** and **38** respectively, and the corresponding levels **L1**(f) and **L2**(f) are compared in comparator **39**. The output from the comparator is represented by the transfer function H(f). The analyses may be short time spectral estimates, e.g. ⅓ octave spectra in the frequency range 100 Hz to 14000 Hz. The test sequences which the subject utters may comprise speech sounds held constant for approximately 1 second. For voiced sounds, the subject may make the pitch vary during the analysis period. The transfer functions of the filters described in connection with FIG. **4** may be based on diagrams of H(f), the spectral density levels of the free field microphone **M4** subtracted from the corresponding levels of the in-the-ear microphone **M2**.

A simplest embodiment of the invention may reduce the system in FIG. **4** to one single time invariant filter. The analysis and selection processing may then be omitted. The transfer function of the single filter is still based on diagrams of the spectral density levels of the free field microphone subtracted from the corresponding levels of the in-the-ear microphone, described in connection with FIG. **5**. The transfer function may be a combination of the results for the various speech sounds, weighted in accordance with their importance for the naturalness of the processed speech.

Another embodiment of the invention is called a "Personal Noise Exposure Dose Meter". Similar to the above embodiments, a microphone **M2** picks up the sound in the meatus **3**. One of the novel features is that this noise exposure is measured in the meatus, even while the ear is already noise protected. The signal from the microphone **M2** is amplified, A/D converted, and analysed in a digital signal processing (DSP) or microcomputer unit **E3** in the same way as described above. According to a preferred embodiment of the invention, the analysis covers both stationary or semistationary noise, and impulsive noise. The result of the analysis is compared is to damage risk criteria and the user gets an audible or other form of warning signal when certain limits are about to be exceeded and actions have to be made. The warning signal may also be transmitted to other parties, e.g. industrial health care monitoring devices. The time record of the analysis may according to a preferred embodiment be stored in a memory, e.g. in the RAM **E8** for later read-out and processing.

FIG. **1** shows a preferred embodiment of the invention with the personal noise exposure dose meter integrated in an active hearing protective communications earplug, comprising a main section **1** containing two microphones, an outer microphone **M1** and an inner microphone **M2**, and a sound generator SG. Since this embodiment is part of a hearing-protecting earplug, a sealing section **2** is attached to the main section. An acoustic transmission channel **T1** connects microphone **M2** to the inner portion of the meatus **3**. Microphone **M2** therefore picks up the sound present in the meatus **3**, just outside the eardrum (tympanum) **4**. An acoustic transmission channel **T2** connects sound generator SG to the inner portion of the meatus **3**. The sound generator SG may provide audible information to the user, in form of warning signals or synthetic speech.

All the electronics as well as the battery are provided in the main section **1**.

A block diagram of one possible implementation of this embodiment is shown in FIG. **2**. The sound is picked up by the microphone **M2**, amplified, and AD-converted before it is fed to the processing unit **E3** with DSP or uP (or both) as central processing units. The memory units **E8** with RAM, **E9** with ROM, and **E10** with EEPROM may store programs, configuration data, and analysis results. Information to the user is generated in the central processing unit **E3**, DA-converted, amplified, and may be presented as audible information via the loudspeaker SG. The digital interface is used for programming, control, and readout of results.

The signal processing for the computation of noise exposure is shown in the flow diagram in FIG. **7**. The signal from microphone **M2** is amplified, converted to digital form and

analysed by algorithms in processing unit E3. First, sample-by-sample equalization represented by block **41** is applied to compensate for irregularities in the microphone response, the transmission channel T1 and the missing ear canal response due to the blocking of the canal by the earplug. The processed samples may according to the invention be evaluated in at least two ways. To obtain the stationary or semistationary noise dose, an A-weighting represented by block **42** is applied. Standards for this A-weighting exists: IEC **179**, and the samples values are squared and accumulated in blocks **43** and **44** respectively. To obtain the peak value for assessing impulsive noise, a C-weighting represented by block **45** is applied according to internationally accepted standards, also IEC **179**, and the peak value (regardless of sign) is saved in block **46**. The noise dose and peak values are finally compared to predetermined limits in a decision algorithm represented by block **47** so that a warning may be given. The audible information to the user may be provided in form of warning signals or synthetic speech. The warning signal may also be transmitted to other parties, e.g. industrial health care monitoring devices. The time record of the two may also be stored in the memory of the processing unit E3 for later readout and further evaluation.

In addition to the use in passive ear protection devices this embodiment of the invention may be used as ear protection when the terminal is used as a headphone coupled to CD players for similar, monitoring the noise dose submitted from the headphones to the ear over time, or in peaks.

Another embodiment of the invention, called "Online verification/control of hearing protector performance", utilises the fact that a sound field locally generated in the cavity near the ear drum is influenced by leakage in the hearing protector. A small electro-acoustic transducer (sound source) SG and a microphone M2 are arranged in a sealing section **2** arranged for attenuating sounds entering the meatus cavity **3**. A digital signal processing (DSP) or microcomputer unit E3 in the main section **1** or in the sealing section **2** is used to generate a predetermined signal which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the sound source SG, which generates a sound field in the closed part of the meatus **3**. The microphone M2 picks up the sound in the meatus cavity **3**. This signal is amplified by amplifier E4, A/D converted by A/D converter ES, and analysed in the digital signal processor or microprocessor E3. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. One example may be that an ear terminal according to the invention is used for checking for leakage in the hearing protection while the wearer is at a gate controlling admittance to a noise exposed area. If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

FIG. **1** illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section **1** containing a microphone M2 and a sound generator SG. An inner sealing section **2** is attached to the outer section, but may be made in one integrated outer section/sealing section **1**,**2**. An acoustic transmission channel T2 connects sound generator SG to the inner portion of the meatus **3**. The sound generator SG produces a predetermined acoustic signal, which generates a sound field in the meatus **3**. An acoustic transmission channel T1 connects microphone M2 to the inner portion of the meatus **3**. Microphone M2 picks up the sound field being set up by the sound generator SG. The signal generation and analysis is carried out in a digital signal processing (DSP) or microcomputer unit E3 with appropriate amplifiers and converters as described in the previous paragraphs. All the electronics **11** as well as the power supply **12** are provided in the outer section **1**.

FIG. **8** illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in the cavity, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented simultaneously, or in sequence. Alternatively, a pseudorandom sequence may be employed. In both cases, prefer rably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm.

An example of the signal processing is shown in the flow diagram in FIG. **8**. In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the processing unit E3. The generators are represented by blocks **81** and **82** respectively. The generators generate both the in-phase (sin) and out-of-phase (cos) components. The in-phase components are added together in block **83**, converted to analogue form, amplified and applied to the sound generator SG. The resulting sound field is picked up by the microphone M2, amplified, converted to digital form and analysed by algorithms in the processing unit E3 for a series of detectors represented by blocks **84**, **85**, **86** and **87**. The in-phase and out-of-phase components of the microphone M2 signal are analysed for each of the two frequencies. The detector algorithm performs a sample-by-sample multiplication of the two input signals and smoothes the result with a low-pass filter. The four detector outputs are applied to a decision algorithm represented by block **88** where they are compared to stored values. The decision result may be a digital "go"/"no go" real time signal indicating acceptable noise protection attenuation or unacceptable protection conditions. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions.

The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

The number and values of frequencies and the smoothing characteristics of the detectors are chosen as a compromise between audibility and response time. If a continuously running verification should be necessary, low frequencies, e.g. in the range of 10–20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance. The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.

The acoustic phenomenon on which the embodiment of the invention is based is illustrated by the electric analogy diagram in FIG. **9**. In the diagram, the sound generator SG is modelled by its acoustic Thevenin equivalent represented by blocks **91** and **92**. The sound pressure p1 is generated by the Thevenin generator **91**, resulting in a volume velocity through the Thevenin impedance Z1(f) **92**. The microphone

M2 is modelled by its acoustic impedance Z3(f) represented by block 93. The sound pressure p2 at the microphone entrance is converted to an electric signal by the microphone. For the purpose of the present illustration, all acoustic elements exposed to the sound pressure generated by the sound generator SG, except the microphone, are lumped together in the acoustic impedance Z2(f) represented by block 95. A leakage in the hearing protector may be modelled by a change in the variable acoustic impedance Z2(f). The change will usually affect both the frequency dependent modulus and the frequency dependent phase of Z2(f). This change leads to a change in the relationship between the sound pressures p2 and p1, which is analysed as described in connection with FIG. 8.

What is claimed is:

1. Ear terminal comprising a sealing section (2) arranged for use in the ear meatus (3) of a human, comprising

an inner microphone (M2) having a sound inlet (S2) for being directed into the meatus (3);

an electronic unit (11) including filtering means coupled to said inner microphone for filtering the signal from said inner microphone (M2), said filtering means being programmable to transform the signals based on the sounds received in the ear by said inner microphone (M2) into sounds having essentially the characteristics of spoken sounds of the wearer of the ear terminal.

2. Ear terminal according to claim 1, comprising a connection interface (13, E12) for transmitting the filtered signal from the ear terminal.

3. Ear terminal according to claim 1, comprising a canal (T1) comprised by the sound inlet (S2), the canal (T1) being arranged through the sealing section (2) between the inward facing surface of the sealing section (2) arranged for sealing off a space portion of the meatus (3), and the sound inlet (S2) of the inner microphone (M2).

4. Ear terminal according to claim 1, comprising a pressure alignment channel (T3) for slow air throughput to and from the meatus (3) through the sealing section (2).

5. Ear terminal according to claim 1, characterized in an outer section (1) arranged for sitting adjacent to the outward facing portion of the sealing section (2) and that part of the inward facing portion of the outer section (1) is formed to fit the concha around the outer portion of the meatus (3).

6. Ear terminal according to claim 4, wherein the pressure alignment channel (T3) includes a pressure release valve (V).

7. Ear terminal according to claim 6, comprising a bypass channel (T4) in the pressure alignment channel (T3).

8. Ear terminal according to claim 1 comprising a filtering means coupled to said inner microphone for filtering the signal from said inner microphone, and sound generation means for generating sounds based on the filtered signal, said filtering means being preprogrammed to transform the signals based on the sounds received in the ear into a sound being recognized by the user as his own voice.

9. Ear terminal according to claim 7, wherein the sound generator is positioned in a separate unit being placed in the user's other ear.

10. Ear terminal according to claim 1, comprising analyzing means coupled to said inner microphone (M2) and indicator means coupled to said analyzing means, said analyzing means analyzing the received sounds and comparing this with predetermined values for acceptable noise levels, and activating said indicating means when said acceptable noise levels are exceeded.

11. Ear terminal according to claim 10, wherein said indicator means is a sound generator directed toward the user's meatus.

12. Ear terminal according to claim 1, comprising a sound generator for being directed toward the user meatus and analyzing means coupled to said inner microphone for measuring the resulting sound field in the meatus, comparing the measured predetermined characteristics characterizing a properly functioning ear protection and indicating means coupled to said analyzing means for being activated when the received signals differ significantly from said predetermined values.

13. An ear terminal, comprising:

a sealing section shaped to insert in the meatus of a human user's ear;

an inner microphone with a sound inlet directed towards the meatus;

an electronic unit including filtering elements coupled to said inner microphone for filtering a signal from said inner microphone, said filtering elements being programmable to transform the signal based on sounds received in the ear by said inner microphone into sounds having essentially the characteristics of spoken sounds of the user of the ear terminal;

a pressure alignment channel for slow air throughput to and from the meatus through the sealing section; and

a pressure release valve included in the pressure alignment channel.

14. An ear terminal, comprising:

a sealing section shaped to insert in the meatus of a human user's ear;

an inner microphone with a sound inlet directed towards the meatus; and

an electronic unit with programmable filtering elements coupled to said inner microphone to filter a signal from said inner microphone and transform the signal based on sounds received from the meatus by said inner microphone into sounds having essentially the characteristics of spoken sounds of the wearer of the ear terminal.

15. The ear terminal of claim 14, further comprising:

an outer microphone with an inlet directed towards an exterior of the human user's ear; and

a canal connecting the sound inlet of the inner microphone through the sealing section to the meatus.

16. The ear terminal of claim 14, wherein the inner microphone is positioned to pick up the user's voice from within a closed space in the meatus, the sealing section creating the closed space when inserted in the user's ear.

17. The ear terminal of claim 16, further comprising:

an air pressure alignment channel extending from the exterior, through the sealing section, to the meatus; and

a pressure release valve included in the pressure alignment channel.

18. The ear terminal of claim 14, wherein, the inner microphone is arranged adjacent the sealing section and with the sound inlet directed towards the meatus.

19. The ear terminal of claim 18, further comprising:

a canal connecting the sound inlet of the inner microphone through the sealing section to the meatus.

20. The ear terminal of claim 16, further comprising:

a pressure release valve to equalize pressure between the exterior and the meatus.

* * * * *

Exhibit PA-K



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

(10) **Patent No.: US 7,502,484 B2**
(45) **Date of Patent: Mar. 10, 2009**

(54) **EAR SENSOR ASSEMBLY FOR SPEECH PROCESSING**

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H04R 25/00 (2006.01)

(52) **U.S. Cl.** **381/320**; 381/312

(58) **Field of Classification Search** 381/23.1, 381/71.1, 71.2, 71.6, 72, 122, 312–315, 322, 381/324, 328; 600/23–25, 539; 340/573.1; 704/270

See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

3,633,705	A	1/1972	Teder
3,995,113	A	11/1976	Tani
4,017,797	A	4/1977	Laessig
4,025,734	A	5/1977	Aloupis
4,150,262	A	4/1979	Ono
4,429,702	A	2/1984	von Recklinghausen
4,533,795	A	8/1985	Baumhauer, Jr. et al.
4,652,702	A	3/1987	Yoshii
4,654,883	A	3/1987	Iwata
4,922,471	A	5/1990	Kuehnel
4,930,156	A	5/1990	Norris
5,003,606	A	3/1991	Bordewijk
5,138,663	A	8/1992	Moseley

(Continued)

FOREIGN PATENT DOCUMENTS

CH	678692	10/1991

(Continued)

OTHER PUBLICATIONS

Sound Radio Products, "Better Living Through Wireless Technology", 2001, http://www.soundradio.com/en-921.html.

(Continued)

Primary Examiner—Brian Ensey
(74) *Attorney, Agent, or Firm*—Renner, Otto, Boisselle & Sklar, LLP

(57) **ABSTRACT**

An ear sensor apparatus includes a microphone positionable with respect to an ear canal to detect sound signals from the ear canal and to generate a corresponding output signal. The apparatus may include a seal for substantially blocking ambient sound signals from reaching the microphone and an interface to convey the output signal to a remote speech recognition apparatus. The output signal may be normalized so that the normalized output signal is more representative of speech from a user's mouth than speech detected from the user's ear.

16 Claims, 5 Drawing Sheets

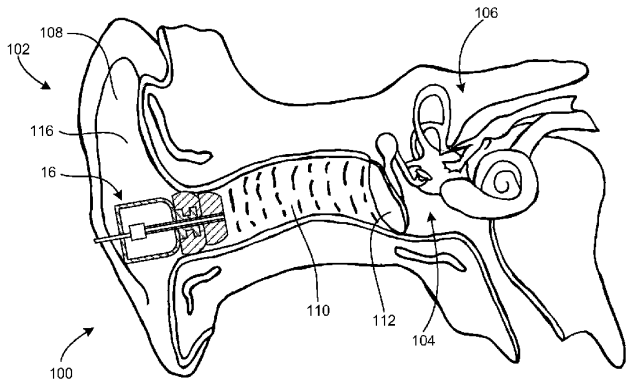


Exhibit PA-K

U.S. PATENT DOCUMENTS

5,149,104	A		9/1992	Edelstein
5,164,984	A		11/1992	Suhami et al.
5,251,263	A		10/1993	Andrea et al.
5,280,524	A		1/1994	Norris
5,298,692	A		3/1994	Ikeda et al.
5,327,506	A		7/1994	Stites, III
5,363,444	A		11/1994	Norris
5,373,555	A		12/1994	Norris et al.
5,426,719	A		6/1995	Franks et al.
5,448,637	A		9/1995	Yamaguchi et al.
5,606,607	A		2/1997	Yamaguchi et al.
5,659,156	A		8/1997	Mauney et al.
5,659,620	A		8/1997	Kuhlman
5,664,014	A		9/1997	Yamaguchi et al.
5,721,783	A	*	2/1998	Anderson 381/328
5,740,258	A		4/1998	Goodwin-Johansson
5,774,565	A		6/1998	Benning et al.
5,790,684	A		8/1998	Niino et al.
5,812,659	A		9/1998	Mauney et al.
5,812,978	A		9/1998	Nolan
5,844,824	A		12/1998	Newman et al.
5,844,984	A		12/1998	Yamaguchi et al.
5,878,396	A		3/1999	Henton
5,881,159	A		3/1999	Aceti et al.
5,896,451	A		4/1999	Deas
5,933,506	A		8/1999	Aoki et al.
6,004,274	A		12/1999	Nolan et al.
6,021,207	A		2/2000	Puthuff et al.
6,022,311	A		2/2000	Juneau et al.
6,024,700	A		2/2000	Nemirovski et al.
6,072,884	A		6/2000	Kates
6,094,492	A		7/2000	Boesen
6,156,585	A		12/2000	Gogoi et al.
6,175,633	B1		1/2001	Morrill et al.
6,181,801	B1		1/2001	Puthuff et al.
6,185,309	B1		2/2001	Attias
6,253,175	B1		6/2001	Basu et al.
6,283,915	B1		9/2001	Aceti et al.
6,408,081	B1		6/2002	Boesen
6,456,721	B1		9/2002	Fukuda
6,503,197	B1		1/2003	Nemirovski
6,560,468	B1		5/2003	Boesen
6,567,524	B1		5/2003	Svean et al.
6,574,345	B1		6/2003	Huang
6,647,368	B2	*	11/2003	Nemirovski 704/270
6,661,901	B1		12/2003	Svean et al.
6,671,379	B2		12/2003	Nemirovski
6,683,965	B1		1/2004	Sapiejewski
6,691,073	B1		2/2004	Erten et al.
6,718,043	B1		4/2004	Boesen
6,728,385	B2		4/2004	Kvaloy et al.
6,741,718	B1		5/2004	Brumitt et al.
6,754,358	B1		6/2004	Boesen et al.
6,754,359	B1	*	6/2004	Svean et al. 381/328
6,819,762	B2		11/2004	Jones et al.
6,917,688	B2	*	7/2005	Yu et al. 381/122
6,952,483	B2		10/2005	Boesen et al.
7,039,195	B1		5/2006	Svean et al.
2001/0017926	A1		8/2001	Vicamini
2002/0118852	A1		8/2002	Boesen
2002/0186858	A1		12/2002	Masuda et al.
2003/0147544	A1		8/2003	Lichtblau
2003/0185403	A1		10/2003	Sibbald
2004/0125979	A1		7/2004	Elidan et al.
2004/0197002	A1		10/2004	Atsumi et al.
2005/0013456	A1		1/2005	Chalupper et al.
2005/0058313	A1		3/2005	Victorian et al.
2005/0117771	A1		6/2005	Vosburgh et al.
2005/0147266	A1		7/2005	Eggers et al.
2005/0157895	A1		7/2005	Lichtblau

FOREIGN PATENT DOCUMENTS

GB	2197158	5/1988
GB	2234882	2/1991
JP	10023578	1/1998
SE	526085	6/2005
WO	9410818	5/1994

OTHER PUBLICATIONS

Ngia, Lester S. H., "Separable Nonlinear Least-Squares Methods for Efficient Off-Line and On-line Modeling of Systems Using Kautz and Laguerre Filters," IEEE Transactions on Circuits and Systems—II: Analog and Digital Signal Processing, vol. 48, No. 6, pp. 562-579, Jun. 2001.

Rupp, Markus, "A Family of Adaptive Filter Algorithms with Decorrelating Properties," IEEE Transactions on Signal Processing, vol. 46, No. 3, pp. 771-775, Mar. 1998.

Yasukawa, Hiroshi and Shimada, Shoji, "An Acoustic Echo Canceller Using Subband Sampling and Decorrelation Methods," IEEE Transactions on Signal Processing, Vo. 41, No. 2, pp. 926-930, Feb. 1993.

Sjoberg, Jonas and Ngia, Lester S.H., "Nonlinear Modeling, Advanced Black-Box Techniques," Chapter 1, "Neural Nets and Related Model Structures for Nonlinear System Identification," pp. 1-28, Kluwer Academic Publishers, 1998.

Westerlund et al., "In-ear Microphone Equalization Exploiting an Active Noise Control", Proceedings of Internoise 2001, Aug. 2001, pp. 1-6.

International Search Report and Written Opinion from corresponding International Application No. PCT/US07/71209.

* cited by examiner

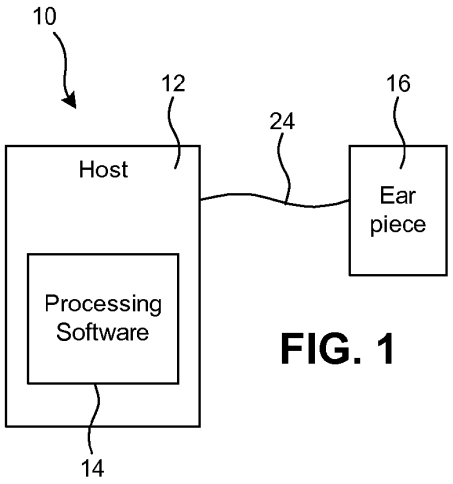


FIG. 1

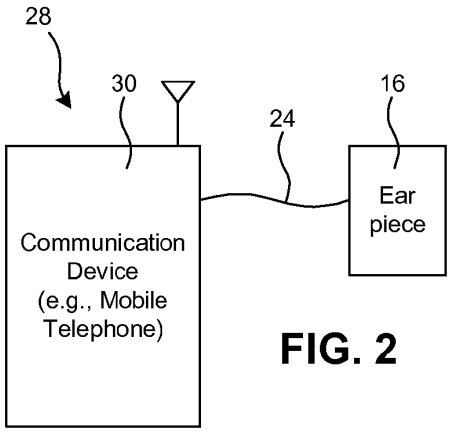


FIG. 2

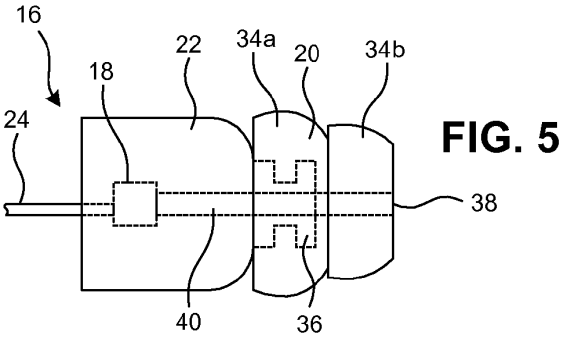


FIG. 5

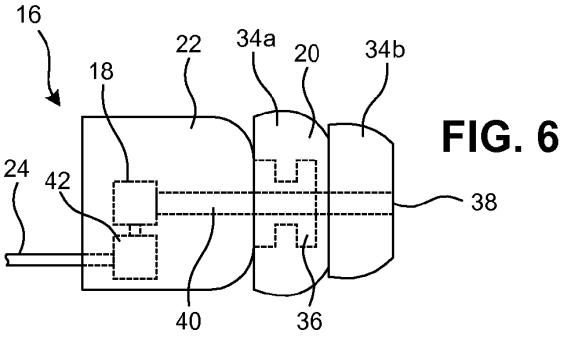


FIG. 6

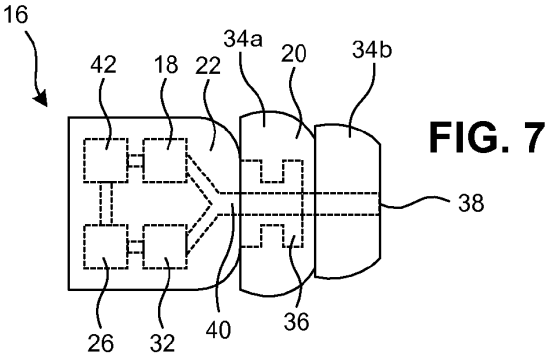


FIG. 7

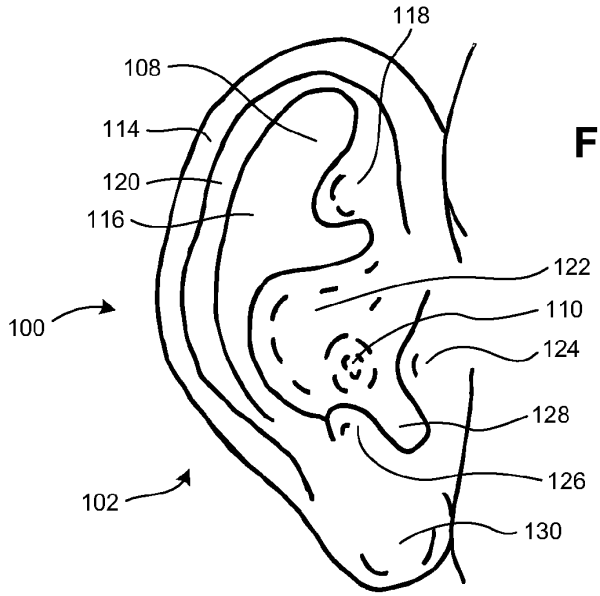

118
108
114
120
116
122
110
100
124
128
126
102
130

FIG. 3

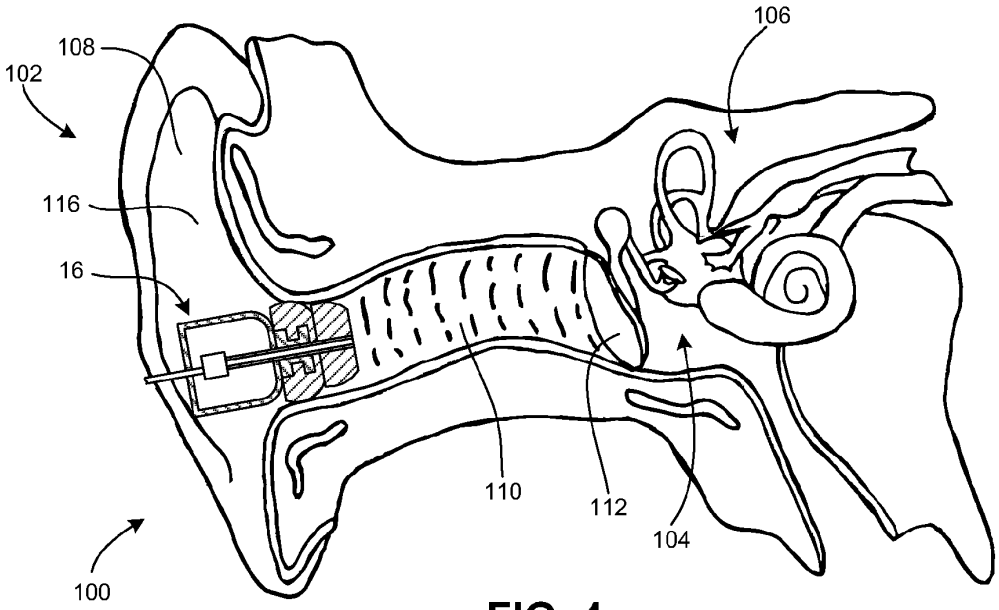

106
108
102
116
16
110
112
104
100

FIG. 4

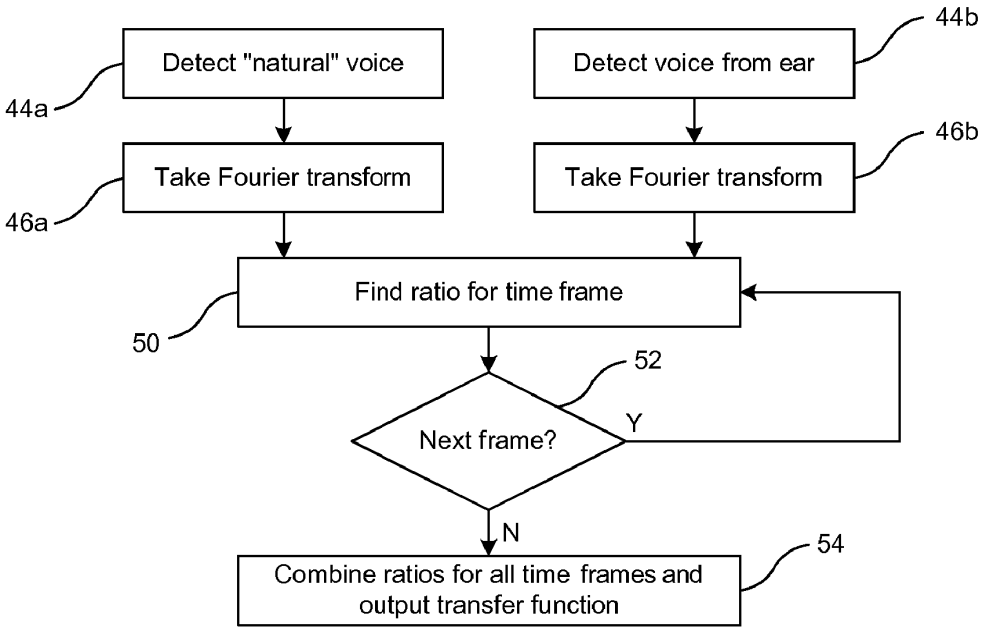

Detect "natural" voice
Detect voice from ear
44a
44b
Take Fourier transform
Take Fourier transform
46a
46b
Find ratio for time frame
50
52
Next frame?
Y
N
Combine ratios for all time frames and output transfer function
54

FIG. 8

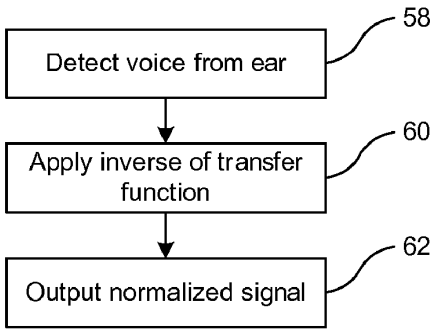

Detect voice from ear
58
Apply inverse of transfer function
60
Output normalized signal
62

FIG. 11