IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

STATON TECHIYA, LLC, Plaintiff, v. HARMAN INTERNATIONAL INDUSTRIES, INCORPORATED and HARMAN-KARDON, INC. Defendants.	Civil Action No. _____ JURY TRIAL DEMANDED

COMPLAINT FOR PATENT INFRINGEMENT

1. Staton Techiya, LLC brings this action for patent infringement under 35 U.S.C. § 271 against Defendants HARMAN International Industries, Incorporated, and Harman-Kardon, Inc. (collectively, "Harman" or "Defendants") and alleges as follows:

THE PARTIES

2. Plaintiff Staton Techiya, LLC ("Techiya" or "Plaintiff") is a Delaware limited liability company having a place of business at 9501 Jagged Creek Ct., Delray Beach, FL, 33446. It was founded in June 2017.

3. Defendant HARMAN International Industries, Incorporated ("HARMAN") is a Delaware corporation having a principal place of business at 400 Atlantic Street, Stamford, CT 06901.

4. Defendant Harman-Kardon, Inc. ("Harman Kardon") is a Delaware corporation having a principal place of business at 250 Crossways Park Drive, Woodbury, NY 11797.

5. On information and belief, Harman Kardon is a wholly owned subsidiary of HARMAN.

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6. On information and belief, JBL, JBL Professional, and AKG are brand names of product lines offered for sale by Harman.

THE ASSERTED PATENTS

7. United States Patent No. 8,081,780 ("the '780 Patent"), entitled "Method and Device for Acoustic Management Control of Multiple Microphones," issued on December 20, 2011, to inventors Steven Wayne Goldstein, John Usher, Marc Andre Boillot, and Jason McIntosh. The '780 Patent issued from U.S. Patent App. Ser. No. 12/115,349, filed on May 5, 2008, and was previously published as U.S. Patent Pub. No. 2009/0016541 on January 15, 2009. A true and correct copy of the '780 Patent is attached as Exhibit A.

8. United States Patent No. 8,805,692 ("the '692 Patent"), entitled "Personal Audio Assistant Device and Method," issued on August 12, 2014, to inventor Steven Wayne Goldstein. The '692 Patent issued from U.S. Patent App. Ser. No. 14/148,752, filed on January 7, 2014, and was previously published as U.S. Patent Pub. No. 2014/0129229 on May 8, 2014. A true and correct copy of the '692 Patent is attached as Exhibit B.

9. United States Patent No. 8,150,044 ("the '044 Patent"), entitled "Method and Device Configured for Sound Signature Detection," issued on April 3, 2012, to inventors Steven W. Goldstein, Mark A. Clements, and Marc A. Boillot. The '044 Patent issued from U.S. Patent App. Ser. No. 11/966,457, filed on December 28, 2007, and was previously published as U.S. Patent Pub. No. 2008/0240458 on October 2, 2008. A true and correct copy of the '044 Patent is attached as Exhibit C.

10. United States Patent No. 11,489,966 ("the '966 Patent"), entitled "Method and Apparatus for In-Ear Canal Sound Suppression," issued on November 1, 2022, to inventors Jason McIntosh, Marc Boillot, John Usher, and Steven Goldstein. The '966 Patent issued from U.S. Patent App. Ser. No. 16,992,861, filed on August 13, 2020, and was previously published as U.S.

Patent Pub. No. 2020/0374404 on November 26, 2020. A true and correct copy of the '632 Patent is attached as Exhibit D.

11. United States Patent No. 11,659,315 ("the '315 Patent"), entitled "Methods and Mechanisms for Inflation," issued on May 23, 2023, to inventors Moises Perez and Steven W. Goldstein. The '315 Patent issued from U.S. Patent App. Ser. No. 17/211,814, filed on March 24, 2021, and was previously published as U.S. Patent Pub. No. 2021/0211801 on July 8, 2021. A true and correct copy of the '315 Patent is attached as Exhibit E.

12. United States Patent No. 11,683,643 ("the '643 Patent"), entitled "Method and Device for In Ear Canal Echo Suppression," issued on June 20, 2023, to inventors Steven Wayne Goldstein, Marc Andre Boillot, John Usher, Jason McIntosh. The '643 Patent issued from U.S. Patent App. Ser. No. 17/215,804, filed on March 29, 2021, and was previously published as U.S. Patent Pub. No. 2021/0219051 on July 15, 2021. A true and correct copy of the '643 Patent is attached as Exhibit F.

13. United States Patent No. 11,665,493 ("the '493 Patent"), entitled "Acoustic Sealing Analysis System," issued on May 30, 2023, to inventors John Usher and John P. Keady. The '493 Patent issued from U.S. Patent App. Ser. No. 17/182,570, filed on February 23, 2021, and was previously published as U.S. Patent Pub. No. 2021/0176576 on June 10, 2021. A true and correct copy of the '493 Patent is attached as Exhibit G.

14. United States Patent No. 11,610,587 ("the '587 Patent"), entitled "Personalized Sound Management and Method," issued on March 21, 2023, to inventors Steven Goldstein and John P. Keady. The '587 Patent issued from U.S. Patent App. Ser. No. 17/736,180, filed on May 4, 2022, and was previously published as U.S. Patent Pub. No. 2022/0262361 on August 18, 2022. A true and correct copy of the '587 Patent is attached as Exhibit H.

15. United States Patent No. 11,521,632 ("the '632 Patent"), entitled "Personal Audio Assistant Device and Method," issued on December 6, 2022, to inventor Steven Wayne Goldstein. The '632 Patent issued from U.S. Patent App. Ser. No. 16/785,822, filed on February 10, 2020, and was previously published as U.S. Patent Pub. No. 2020/0227062 on July 16, 2020. A true and correct copy of the '632 Patent is attached as Exhibit I.

16. The '780, '692, '044, '966, '315, '643, '493, '587, and '632 Patents are collectively referred to herein as the "Asserted Patents."

JURISDICTION AND VENUE

17. This action arises under the Patent Act, 35 U.S.C. § 1 et seq.

18. Subject matter jurisdiction is proper in this Court under 28 U.S.C. §§ 1331 and 1338(a) because this is a civil action for patent infringement arising under the federal patent laws of the United States.

19. Defendants are subject to personal jurisdiction in this District by virtue of the fact that they are incorporated under the laws of the State of Delaware.

20. Furthermore, on information and belief, Defendants have substantial and continuous contacts within the State of Delaware because they have placed the AKG N400NC TWS Headphones ("AKG Accused Product"), Harman Kardon FLY TWS Headphones ("Harman Kardon Accused Product"), the JBL Live 300TWS, Tour Pro+ TWS, Tour Pro 2, Tune Buds, Live Free 2 TWS, Live Pro + TWS, and Quantum TWS Air Headphones ("JBL Accused Products") (collectively, the "Accused Products") into the stream of commerce by shipping the Accused Products into this District.

21. Venue is proper in this Judicial District pursuant to 28 U.S.C. § 1400(b) because Defendants reside within this District as they are incorporated within this District.

JOINDER

22. Joinder of Defendants is proper under 35 U.S.C. § 299. The allegations of patent infringement contained herein arise out of the same series of transactions or occurrences relating to the making (or having made), using (or inducing the use of), selling, or offering for sale within the United States, the same Accused Products. Nonlimiting examples of these products include, but are not limited to the AKG Accused Product, the Harman Kardon Accused Product, and the JBL Accused Products imported, sold, offered for sale, and/or used in this District.

ALLEGATIONS OF PATENT INFRINGEMENT

23. Plaintiff incorporates the allegations of the foregoing paragraphs as if fully restated herein.

24. This case involves patented technologies that helped to revolutionize and have become widely adopted in the field of ear-worn technology. The field of ear-worn technology is a rapidly developing field that encompasses: sound processing, for personal assistance technologies or hearing aids; intelligent systems packaging, including microelectronics and micro-mechanical systems that enable comfort, fit, and performance; and data processing, including artificial intelligence and big data processing.

25. The result of comprehensive research and development, Plaintiff's patented technologies combine many disciplines of ear-worn technology to create solutions that provide exceptional functionality to users.

26. For example, Plaintiff's patented technologies allow level-dependent ambient awareness, user-controlled passthrough, dosimetry, personal voice activation in high-ambient-noise conditions, and acoustic event detection.

27. Plaintiff's patented technology represents such a vast improvement on the traditional ear-worn technology systems that it has become widely adopted by other market participants.

28. Harman provide ear-worn technology systems that, without authorization, implement Plaintiff's patented technologies.

29. No later than September 14, 2018, Harman and Plaintiff met to discuss Plaintiff's patented technology and ongoing research and development efforts in the ear-worn technology field. Plaintiff initially met with Chief Audio Fellow of HARMAN, Dr. Sean Olive.

30. Plaintiff and Harman executed a mutual non-disclosure agreement shortly thereafter. The mutual non-disclosure agreement protected disclosure of information even to parent entities of Harman.

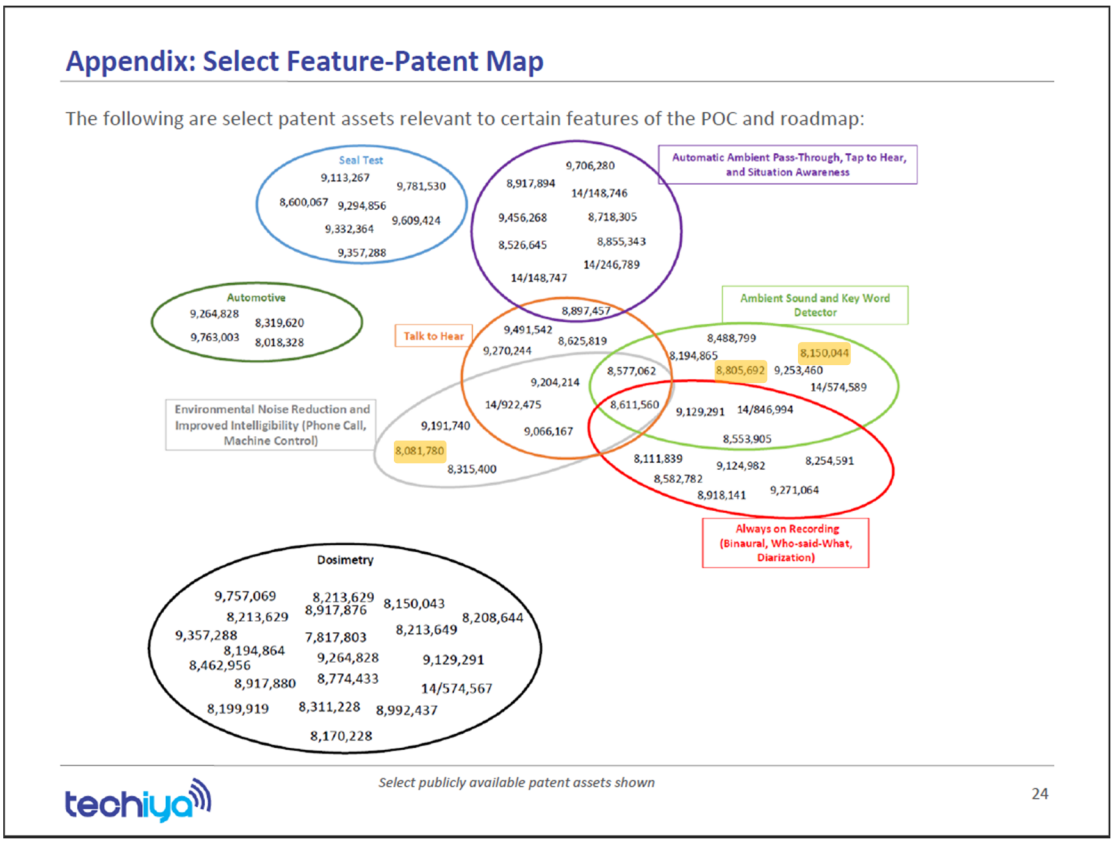
31. Beginning in October of 2018 Plaintiff sent studies, marketing materials, and prototypes to Harman. Some of the information sent to Harman was directly related to the Asserted Patents. For example, Plaintiff provided information to Harman regarding Plaintiff's "talk-to-hear" technology and ambient awareness technology, embodied in at least the '780 Patent, '632 Patent, '692 Patent, '966 Patent, '315 Patent, and '643 Patent.

32. Harman reviewed the information received from Plaintiff. Todd Welti (Senior Principal Engineer of Acoustics at HARMAN) conducted various tests on the prototypes that Plaintiff sent to Harman. Harman subsequently requested additional information and more prototypes. In response, Plaintiff provided additional prototypes and shared proprietary whitepapers.

33. In early 2019, Plaintiff met with other representatives from Harman. Plaintiff presented additional demonstrations of the prototypes and showed a promotional video advertising patented ear-worn technologies and how consumers would benefit by using them. Plaintiff subsequently sent Harman the promotional video.

34. In February 2019, in a follow-up message to a meeting, Plaintiff sent an "Opportunity Overview" slide deck to Harman. The deck contained an enumeration of several

different advantages offered by Plaintiff's ear-worn technology and a description of patents relevant to some of the related fields. Several of the Asserted Patents were identified in the deck.



Excerpted slide from deck sent to Harman in February 2019. Several Asserted Patents were included in this deck (emphasized in yellow).

35. In March of 2019, Plaintiff met with HARMAN Chief Financial Officer Evelyn Heinbach. The agenda, proposed by Ms. Heinbach, included questions about Plaintiff's intellectual property and whether Plaintiff could provide demonstrations and technical background on its patented technology.

36. Later that year, on May 15, 2019, Plaintiff presented to Harman a comprehensive valuation model of Plaintiff's intellectual property arranged in a spreadsheet. The spreadsheet contained hundreds of patents and patent applications, categorized by feature and technology. This spreadsheet contained references to the '780 and '692 Patents.

37. On information and belief, Harman released the AKG N400NC TWS Headphones (the AKG Accused Product) no earlier than March of 2020 and the Harman Kardon FLY TWS (the Harman Kardon Accused Product) no earlier than April of 2020. On information and belief, the other Accused Products were released no earlier than March of 2021.

38. On May 4, 2020, Ms. Heinbach confirmed that "Samsung has done a detailed review of [Plaintiff's] patent portfolio." On information and belief, based on this "detailed review," Harman and its parent, Samsung, had knowledge and notice of Plaintiff's patents that had issued on or before May 4, 2020, including an understanding of the scope of the claims of those patents. Despite this knowledge Harman has infringed, and continues to infringe, patents asserted in this Complaint that were issued prior to May 4, 2020.

39. On May 23, 2020 Ms. Heinbach informed Plaintiff that Harman would not be "looking to pursue further steps at this point."

40. On information and belief, despite Harman's robust knowledge of the Asserted Patents and Plaintiff's patented technology, Harman made the deliberate decision to sell products and services that they knew infringe these patents. Harman's continued infringement of the Asserted Patents with extensive knowledge about them constitutes willful infringement.

41. As set forth below, the Accused Products incorporate, without any license or permission from Plaintiff, technology protected by the Asserted Patents. Plaintiff respectfully seeks relief from this Court for Defendants' infringement.

REPRESENTATIVE ACCUSED PRODUCTS

42. Representative Accused Products include, but are not limited to, the AKG N400NC TWS Headphones, Harman Kardon FLY TWS Headphones, the JBL Live 300TWS, Tour Pro+ TWS, Tour Pro 2, Tune Buds, Live Free 2 TWS, Live Pro+ TWS, and Quantum TWS Air

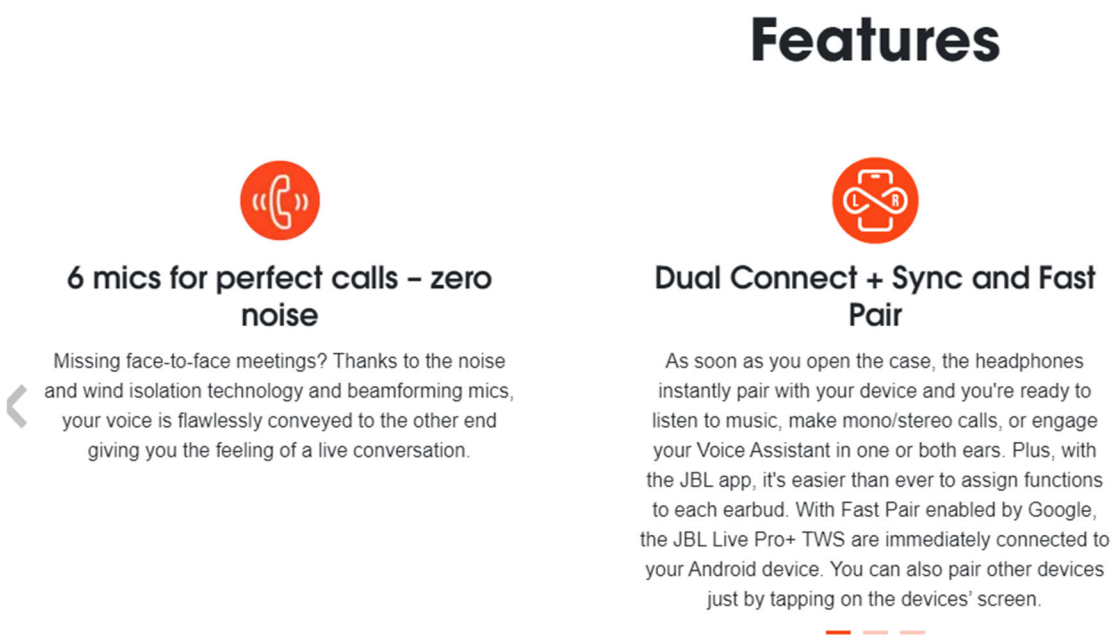
Headphones and other similar devices. Some of these products are designed and configured to be used with Samsung's Bixby voice-driven user interface.

43. The JBL Live Pro+ TWS Headphones is an exemplary infringing product.



Source: https://www.jbl.com/wireless-headphones/LIVE-PRO-PLUS-TWS-.html

44. On information and belief, the JBL Live Pro+ TWS Headphones utilize multiple microphones to perform noise and wind isolation. For example, the JBL Live Pro+ TWS Headphones utilize a dedicated microphone for "environmental noise reduction."



Source: https://www.jbl.com/wireless-headphones/LIVE-PRO-PLUS-TWS-.html

45. On information and belief, the JBL Live Pro+ TWS Headphones perform Adaptive Noise Cancelling, Ambient Aware, Talk Thru, and Voice Aware. These features require detecting and analyzing environmental sound captured by the various microphones on the product.

46. On information and belief, the AKG N400NC TWS, Harman Kardon FLY TWS Headphones, the JBL Live 300TWS, Tour Pro+ TWS, Tour Pro 2, Tune Buds, Live Free 2 TWS, Live Pro + TWS, and Quantum TWS Air Headphones also have same or similar features, components, and functionalities.

COUNT I

Defendants' Infringement of the '780 Patent

47. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

48. Plaintiff is the assignee and lawful owner of the '780 Patent, and Plaintiff holds all right, title and interest in and to the '780 Patent. The '780 Patent is valid and enforceable.

49. Defendants have directly infringed, and continue to directly and willfully infringe, the '780 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '780 Patent, such as the JBL Accused Products. Further discovery may reveal additional infringing products and/or models.

50. For example, and without limitation, the JBL Accused Products, including the JBL Live Pro+ TWS and similar earphone products, infringe at least claim 1 of the '780 Patent.

51. Defendants have had notice of their infringement of the '780 Patent since receiving the aforementioned slide deck from Plaintiff in February of 2019. In the alternative, Defendants have had notice of their infringement of the '780 Patent at least since the date of filing of the complaint in this action.

52. The JBL Accused Products are exemplary products that practice at least claim 1 of the '780 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the '780 Patent.

53. The '780 Patent claims: 1. A method for acoustic management control suitable for use in an earpiece, the method comprising the steps of: capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal; capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal; measuring a background noise signal from the electronic ambient signal or the electronic internal signal; and mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal.

54. The JBL Accused Products are earpieces that have three microphones: 2 outer microphones and 1 inner microphone. Those microphones are used, in part, to track and to eliminate outside sounds. Two of the microphones are used to generate signals based on ambient sounds and the inner microphone generates signals based on inner ear sounds. On information and belief, the three

microphones generate first, second, and third microphone signals, wherein the first and second microphone signals represent an electronic ambient signal, and the third microphone signal represents an electronic internal signal. The JBL Accused Products include a speaker, specifically a two-way dynamic speaker. On information and belief, the JBL Accused Products process the microphone signals and audio signals generated from the three microphones to measure a background noise signal. The JBL Accused Products support adaptive noise cancellation (ANC) and TalkThru. The ANC feature tracks and eliminates outside sounds by using the onboard microphones to detect ambient noise frequencies and then electronically generates opposite sound frequencies (based on "the first microphone signal or the second microphone signal or the third microphone signal or a combination of microphone signals") to create a noise reduction signal. On information and belief, when the ANC feature is turned on, the JBL Accused Products mix the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal to achieve ANC. Similarly, when performing TalkThru, the mixed signal produced allows the user of the earpiece to hear his or her own voice.

55. Defendants have indirectly infringed and continue to indirectly infringe the '780 Patent by actively inducing infringement of the '780 Patent by others, such as users of the JBL Accused Products within this District.

56. Defendants' direct and indirect infringement of the '780 Patent has damaged and will continue to damage Plaintiff.

57. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

58. After receiving actual notice of the '780 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the JBL Accused Products.

59. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '780 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '780 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '780 patent.

60. Defendants have willfully infringed and continue to willfully infringe the '780 Patent.

COUNT II

Defendants' Infringement of the '692 Patent

61. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

62. Plaintiff is the assignee and lawful owner of the '692 Patent, and Plaintiff holds all right, title and interest in and to the '692 Patent. The '692 Patent is valid and enforceable.

63. Defendants have directly infringed, and continue to directly and willfully infringe, the '692 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '692 Patent, such as the Accused Products. Further discovery may reveal additional infringing products and/or models.

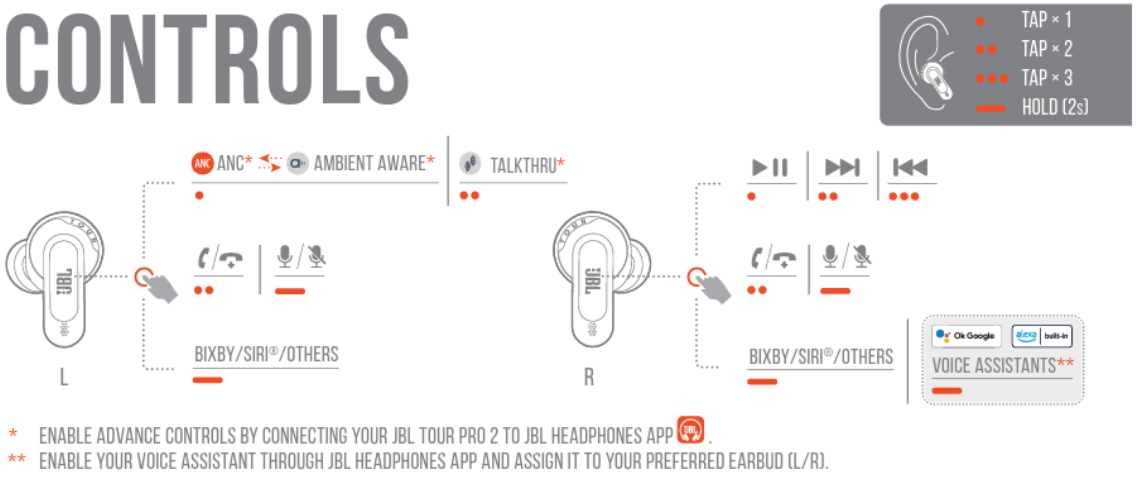
64. For example, and without limitation, the Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the '692 Patent.

65. Defendants have had notice of their infringement of the '692 Patent since receiving the aforementioned slide deck from Plaintiff in February of 2019. In the alternative, Defendants have had notice of their infringement of the '692 Patent at least since the date of filing of the complaint in this action.

66. The Accused Products are exemplary products that practice at least claim 1 of the '692 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the '692 Patent.

67. The '692 Patent claims: 1. A wearable device, comprising: a housing for the wearable device; an ambient microphone within or on the housing; a communication module within or on the housing; a logic circuit communicatively coupled to the ambient microphone; a memory storage unit communicatively coupled to the logic circuit; and an interaction element, the interaction element and logic circuit cooperatively initiate control of media content from the wearable device or initiate operations of the wearable device; wherein the logic circuit is configured to cooperatively initiate control of media content of the wearable device in response to analysis of audio captured at the wearable device at the ambient microphone and analyzed at a remote server in response to instructions received from the remote server.

68. As an example, the JBL Tour Pro 2 is an earpiece that has an ambient microphone on the housing. On information and belief, the earpiece also has a chipset that contains a communication module, a logic circuit coupled to the microphone on the housing, and a memory unit coupled to the logic circuit. A user can activate an interaction element of the earpiece to initiate operations of the earpiece, for example by tapping on the outside to activate voice assistant controls, toggle Adaptive Noise Cancelling, activate TalkThru, or answer and hang up calls. When the interactive element is activated, a user can control functionalities like Ambient Aware or Adaptive Noise Cancelling through a remote server in the JBL smartphone app. Moreover, these functionalities allow the user to control media content on the device or the level of ambient noise passed through to the device.



Source: https://www.jbl.com/on/demandware.static/-/Sites-masterCatalog_Harman/default/dw511411ed/pdfs/JBL_TOUR%20Pro2_QSG%20Multilingual.pdf

69. Defendants have indirectly infringed and continue to indirectly infringe the '692 Patent by actively inducing infringement of the '692 Patent by others, such as users of the Accused Products within this District.

70. Defendants' direct and indirect infringement of the '692 Patent has damaged and will continue to damage Plaintiff.

71. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

72. After receiving actual notice of the '692 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the Accused Products.

73. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '692 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '692 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '692 patent.

74. Defendants have willfully infringed and continue to willfully infringe the ’692 Patent.

COUNT III

Defendants’ Infringement of the ’044 Patent

75. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

76. Plaintiff is the assignee and lawful owner of the ’044 Patent, and Plaintiff holds all right, title and interest in and to the ’044 Patent. The ’044 Patent is valid and enforceable.

77. Defendants have directly infringed, and continue to directly and willfully infringe, the ’044 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the ’044 Patent, such as the Accused Products. Further discovery may reveal additional infringing products and/or models.

78. For example, and without limitation, the Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the ’044 Patent.

79. Defendants have had notice of their infringement of the ’044 Patent since receiving the aforementioned slide deck from Plaintiff in February of 2019. In the alternative, Defendants have had notice of their infringement of the ’044 Patent at least since the date of filing of the complaint in this action.

80. The Accused Products are exemplary products that practice at least claim 1 of the ’044 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the ’044 Patent.

81. The ’044 Patent claims: 1. An acoustic device, comprising: an Ambient Sound Microphone (ASM) configured to capture ambient sound; at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the ECR, where the processor monitors the ambient sound for a target sound and adjusts the audio

delivered to the ear canal based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.

82. As an example, the JBL Live Pro+ is an earpiece that includes VoiceAware and Ambient Aware features. The earpiece includes at least three microphones. Upon information and belief, at least one of the microphones operates to capture ambient sound. Said microphone may choose to pick up the ambient sound based on its sound signature. The earpiece also includes an ear canal receiver designed to broadcast audio to the user's ear. Inside the earpiece a processor generates a sound signature from the captured ambient sound. Upon information and belief, the earpiece also includes a VoiceAware feature to capture the target sound of the user's own voice and adjusts the playback to the user based on the user's voice.



Source: https://www.jbl.com/on/demandware.static/-/Sites-masterCatalog_Harman/default/dw3dfb14dd/pdfs/JBL_TOUR%20Pro2_QSG%20Multilingual.pdf

83. Defendants have indirectly infringed and continue to indirectly infringe the '044 Patent by actively inducing infringement of the '044 Patent by others, such as users of the Accused Products within this District.

84. Defendants' direct and indirect infringement of the '044 Patent has damaged and will continue to damage Plaintiff.

85. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

86. After receiving actual notice of the '044 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the Accused Products.

87. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '044 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '044 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '044 patent.

88. Defendants have willfully infringed and continue to willfully infringe the '044 Patent.

COUNT IV

Defendants' Infringement of the '966 Patent

89. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

90. Plaintiff is the assignee and lawful owner of the '966 Patent, and Plaintiff holds all right, title and interest in and to the '966 Patent. The '966 Patent is valid and enforceable.

91. Defendants have directly infringed, and continue to directly and willfully infringe, the '966 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '966 Patent, such as the JBL Accused Products. Further discovery may reveal additional infringing products and/or models.

92. For example, and without limitation, the JBL Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the ’966 Patent.

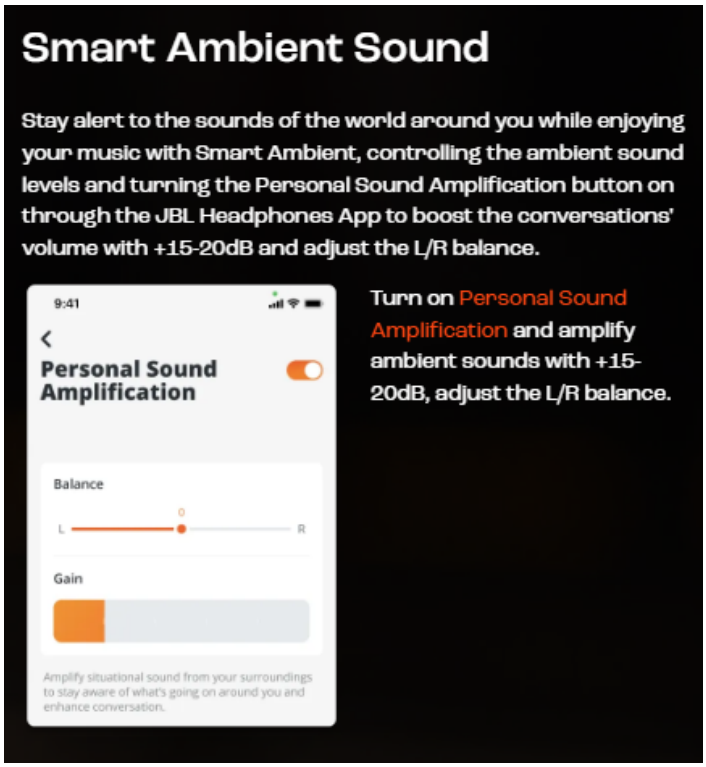
93. Defendants have had notice of their infringement of the ’966 Patent at least since the date of filing of the complaint in this action.

94. The JBL Accused Products are exemplary products that practice at least claim 1 of the ’966 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the ’966 Patent.

95. The ’966 Patent claims: 1. A method for acoustic management control suitable for use in an earpiece, the method comprising the steps of: measuring an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal; measuring an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal; generating a background noise level (BNL) from at least one of the electronic ambient signal or the electronic internal signal or a combination of both; generating a voice activity level (VAL); generating a gain, wherein the gain is a function of the background noise level and the voice activity level; generating a modified electronic ambient signal by applying the gain to the electronic ambient signal; mixing the modified electronic ambient signal with an audio signal to produce a mixed audio signal; and sending the mixed audio signal to a speaker.

96. As an example, the JBL Live Pro+ is an earpiece that includes Adaptive Noise Cancelling and Personal Sound Amplification features. The earpiece includes at least three microphones. Upon information and belief, at least one of the microphones operates to produce an electronic ambient signal and another operates as an ear canal microphone to produce an electronic internal signal. Upon information and belief, the earpiece generates a background noise level from at least one of the signals and a voice activity level, which is then used to generate a gain. For example,

the Personal Sound Amplification functionality allows the user to boost conversation volume. The boosted conversation volume is an application of the gain to the electronic ambient signal from the external microphone, which is mixed and sent to the receiver in the earpiece.



Source: https://www.jbl.com/wireless-earbuds/TOUR-PRO-2.html

97. Defendants have indirectly infringed and continue to indirectly infringe the '966 Patent by actively inducing infringement of the '966 Patent by others, such as users of the JBL Accused Products within this District.

98. Defendants' direct and indirect infringement of the '966 Patent has damaged and will continue to damage Plaintiff.

99. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

100. After receiving actual notice of the '966 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the JBL Accused Products.

101. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '966 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '966 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '966 patent.

102. Defendants have willfully infringed and continue to willfully infringe the '966 Patent.

COUNT V

Defendants' Infringement of the '315 Patent

103. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

104. Plaintiff is the assignee and lawful owner of the '315 Patent, and Plaintiff holds all right, title and interest in and to the '315 Patent. The '315 Patent is valid and enforceable.

105. Defendants have directly infringed, and continue to directly and willfully infringe, the '315 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '315 Patent, such as the JBL Accused Products. Further discovery may reveal additional infringing products and/or models.

106. For example, and without limitation, the JBL Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the '315 Patent.

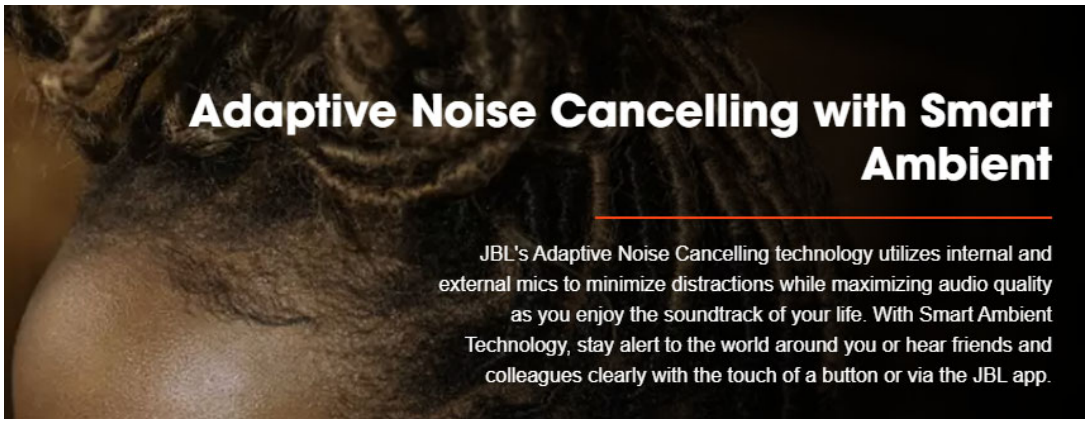
107. Defendants have had notice of their infringement of the '315 Patent at least since the date of filing of the complaint in this action.

108. The JBL Accused Products are exemplary products that practice at least claim 1 of the '315 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the '315 Patent.

109. The ’315 Patent claims: 1. An earpiece comprising: a first ambient microphone configured to generate a first microphone signal; a second ambient microphone configured to generate a second microphone signal; an ear canal microphone configured to generate a third microphone signal; a speaker; a memory that stores instructions; a logic circuit, where the logic circuit is configured to execute the instructions to perform operations comprising: generating a noise reduction signal using at least one of the first microphone signal or the second microphone signal or the third microphone signal or a combination of microphone signals; receiving an audio signal; generating a mixed audio signal by mixing the noise reduction signal with the audio signal; and sending a modified audio signal to the speaker, where the modified audio signal includes the mixed audio signal.

110. As an example, the JBL Live Pro+ is an earpiece that includes Adaptive Noise Cancelling and Personal Sound Amplification features. The earpiece includes at least three microphones. Upon information and belief, at least two of the microphones operate to produce a first microphone signal and a second microphone signal. Upon information and belief, the third microphone is an ear canal microphone configured to generate a third microphone signal. The earpiece also contains a speaker, a memory to store instructions, and a logic circuit configured to execute instructions. Upon information and belief, when the Adaptive Noise Cancelling technology is activated, the logic circuit will generate a noise reduction signal using at least one of the microphone signals or a combination of said signals. Then the circuit will receive an audio signal (from, for example, the user’s media), which it mixes with the noise reduction signal and sends to the speaker for the user to experience reduced distraction and maximized audio quality during the use of Adaptive Noise Cancelling.

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Source: https://www.jbl.com/headphones/TOUR-PRO-TWS-.html?dwvar_TOUR-PRO-TWS-_color=Black-AM-Current&cgid=headphones

111. Defendants have indirectly infringed and continue to indirectly infringe the ’315 Patent by actively inducing infringement of the ’315 Patent by others, such as users of the JBL Accused Products within this District.

112. Defendants’ direct and indirect infringement of the ’315 Patent has damaged and will continue to damage Plaintiff.

113. Plaintiff is entitled to recover damages adequate to compensate it for Defendants’ direct and indirect infringement.

114. After receiving actual notice of the ’315 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the JBL Accused Products.

115. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the ’315 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the ’315 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the ’315 patent.

116. Defendants have willfully infringed and continue to willfully infringe the ’315 Patent.

COUNT VI

Defendants' Infringement of the '643 Patent

117. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

118. Plaintiff is the assignee and lawful owner of the '643 Patent, and Plaintiff holds all right, title and interest in and to the '643 Patent. The '643 Patent is valid and enforceable.

119. Defendants have directly infringed, and continue to directly and willfully infringe, the '643 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '643 Patent, such as the JBL Accused Products. Further discovery may reveal additional infringing products and/or models.

120. For example, and without limitation, the JBL Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the '643 Patent.

121. Defendants have had notice of their infringement of the '643 Patent at least since the date of filing of the complaint in this action.

122. The JBL Accused Products are exemplary products that practice at least claim 1 of the '643 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the '643 Patent.

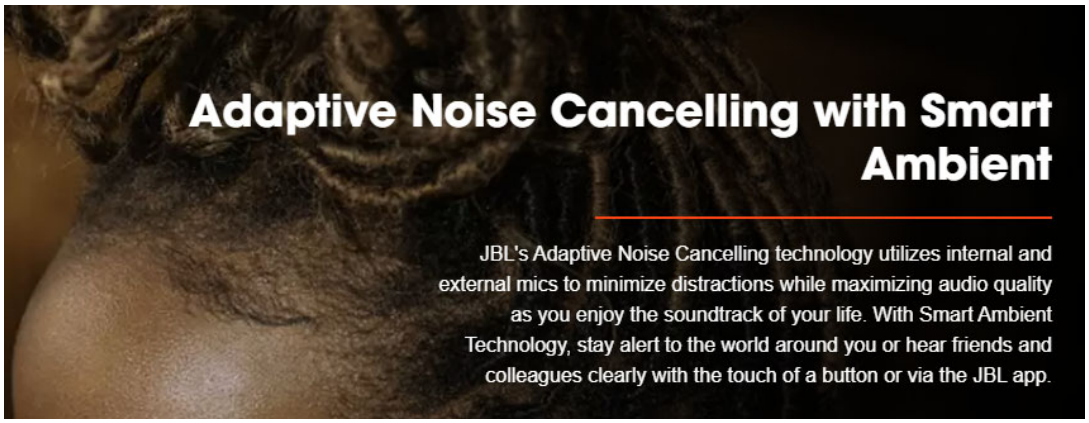
123. The '643 Patent claims: 1. A device, comprising: a first ambient microphone configured to generate a first ambient signal, wherein the first ambient microphone is part of an earpiece; a second ambient microphone configured to generate a second ambient signal, wherein the second ambient microphone is part of the earpiece; an ear canal microphone configured to generate an ear canal signal, wherein the ear canal microphone is part of the earpiece; a speaker, wherein the speaker is part of the earpiece; an eartip, configured to seal the earpiece between a

first side and a second side of the earpiece; a memory that stores instructions; and a processor, wherein the processor is part of the earpiece, wherein the processor is operatively connected to the first ambient microphone, wherein the processor is operatively connected to the second ambient microphone, wherein the processor is operatively connected to the ear canal microphone, wherein the processor is operatively connected to the speaker, wherein the processor is operatively connected to the memory, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: receiving a first microphone signal, wherein the first microphone signal is at least one of the first ambient signal or a modified first ambient signal; receiving a second microphone signal, wherein the second microphone signal is at least one of the ear canal signal or a modified ear canal signal; receiving a third microphone signal, wherein the third microphone signal is at least one of the second ambient signal or a modified second ambient signal; generating a background noise reduction signal using either the first and second microphone signals or the second and third microphone signals; mixing an audio content signal and the background noise reduction signal generating a mixed signal; and sending a modified mixed signal to the speaker.

124. As an example, the JBL Live Pro+ is an earpiece that includes Adaptive Noise Cancelling and Personal Sound Amplification features. The earpiece includes at least three microphones. Upon information and belief, at least two of the microphones operate to produce a first ambient signal and a second ambient signal. Upon information and belief, the third microphone is an ear canal microphone configured to generate an ear canal signal. The earpiece also contains a speaker, an eartip, a memory to store instructions, and a processor configured to execute instructions. Upon information and belief, when the Adaptive Noise Cancelling technology is activated, the processor will generate a background noise reduction signal using at least one of the microphone signals or a combination of said signals. Then the circuit will receive an audio signal (from, for example, the user's

media), which it mixes with the background noise reduction signal and sends to the speaker for the user to experience reduced distraction and maximized audio quality during the use of Adaptive Noise Cancelling.

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Source: https://www.jbl.com/headphones/TOUR-PRO-TWS-.html?dwvar_TOUR-PRO-TWS-_color=Black-AM-Current&cgid=headphones

125. Defendants have indirectly infringed and continue to indirectly infringe the '643 Patent by actively inducing infringement of the '643 Patent by others, such as users of the JBL Accused Products within this District.

126. Defendants' direct and indirect infringement of the '643 Patent has damaged and will continue to damage Plaintiff.

127. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

128. After receiving actual notice of the '643 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the JBL Accused Products.

129. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '643 Patent. Defendants knew and should have known that their actions would cause direct and induced

infringement of the ’643 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the ’643 patent.

130. Defendants have willfully infringed and continue to willfully infringe the ’643 Patent.

COUNT VII

Defendants’ Infringement of the ’493 Patent

131. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

132. Plaintiff is the assignee and lawful owner of the ’493 Patent, and Plaintiff holds all right, title and interest in and to the ’493 Patent. The ’493 Patent is valid and enforceable.

133. Defendants have directly infringed, and continue to directly and willfully infringe, the ’493 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the ’493 Patent, such as the JBL Accused Products. Further discovery may reveal additional infringing products and/or models.

134. For example, and without limitation, the JBL Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the ’493 Patent.

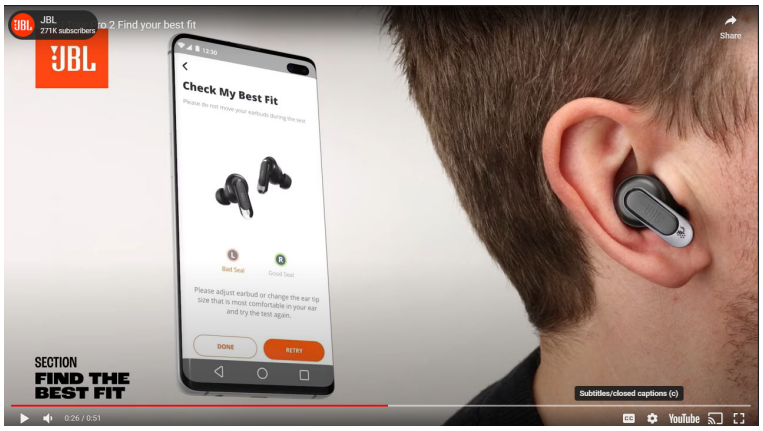
135. Defendants have had notice of their infringement of the ’493 Patent at least since the date of filing of the complaint in this action.

136. The JBL Accused Products are exemplary products that practice at least claim 1 of the ’493 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the ’493 Patent.

137. The ’493 Patent claims: An audio device configured to test a seal of an earphone comprising: a memory that stores instructions; and a processor, wherein the processor is configured to execute the instructions to perform operations, the operations comprising: generating a test signal, wherein the test signal includes multiple frequency components, wherein at least one

component lies between a range of 30 Hz and 200 Hz; sending the test signal to a speaker of an earphone, wherein the earphone includes a first microphone, the speaker, and a sealing section, and wherein the sealing section is configured to form an acoustic barrier between a first side of the earphone and a second side of the earphone when the earphone is worn by a user, receiving a first microphone signal from the first microphone; calculating a value, using a first signal and a second signal, wherein the value is the result of a non phase calculation of at least one of a coherence, a correlation, a cross-correlation, a difference, or a combination; comparing the value to a threshold to determine a seal quality between the first side and the second side; generating a message to convey the seal quality; and sending the message to the user.

138. As an example, the JBL Live Pro+ is an earpiece that includes a feature to "Check My Best Fit." This function finds the ear tip that best suits the user for maximum comfort, minimum sound leakage, and optimal support to noise cancelling. Upon information and belief the earpiece contains a memory unit that stores instructions and processor configured to execute instructions. Upon information and belief, the earpiece generates a test signal with a frequency component between a range of 30 Hz and 200Hz, which is sent to a speaker in the earpiece. The microphones on the earpiece then detect the signal and calculate a value to determine seal quality, which aids the JBL smartphone app in determining the seal quality. The quality metric is then conveyed and sent to the user.



Source: https://www.youtube.com/watch?v=9mvzzAa0T7s&t=26s



Source: https://www.jbl.com/on/demandware.static/-/Sites-masterCatalog_Harman/default/dw511411ed/pdfs/JBL_TOUR%20Pro2_QSG%20Multilingual.pdf

139. Defendants have indirectly infringed and continue to indirectly infringe the '493 Patent by actively inducing infringement of the '493 Patent by others, such as users of the JBL Accused Products within this District.

140. Defendants' direct and indirect infringement of the '493 Patent has damaged and will continue to damage Plaintiff.

141. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

142. After receiving actual notice of the '493 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the JBL Accused Products.

143. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '493 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '493 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '493 patent.

144. Defendants have willfully infringed and continue to willfully infringe the '493 Patent.

COUNT VIII

Defendants' Infringement of the '587 Patent

145. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

146. Plaintiff is the assignee and lawful owner of the '587 Patent, and Plaintiff holds all right, title and interest in and to the '587 Patent. The '587 Patent is valid and enforceable.

147. Defendants have directly infringed, and continue to directly and willfully infringe, the '587 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the '587 Patent, such as the Accused Products. Further discovery may reveal additional infringing products and/or models.

148. For example, and without limitation, the Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the '587 Patent.

149. Defendants have had notice of their infringement of the '587 Patent at least since the date of filing of the complaint in this action.

150. The Accused Products are exemplary products that practice at least claim 1 of the '587 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the '587 Patent.

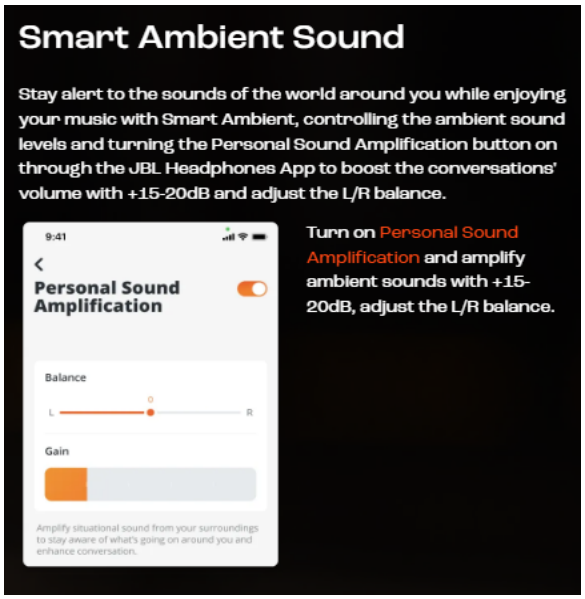
151. The '587 Patent claims: 1. A system comprising: a user interface; and an audio device comprising: a microphone configured to measure ambient sound and generate a microphone signal; a speaker configured to emit an audio signal; a memory configured to store instructions; an audio buffer configured to store at least a portion of the microphone signal; a data buffer; a processor operatively coupled to the memory, the processor operatively coupled to the audio buffer, the processor operatively coupled to the data buffer, wherein the processor is configured to execute the instructions to perform operations comprising: receiving the microphone signal;

sending a portion of the microphone signal to the audio buffer; analyzing the audio buffer for detecting a sonic signature; notifying a user when the sonic signature is detected; sending a modified microphone signal to the data buffer replacing or adding to previous data stored in the data buffer; and sending a portion of the data stored in the data buffer, spanning a time period, to the speaker in response to a user request, wherein the user request is one of the sonic signature that is a voice command or a manual input from the user interface.

152. As an example, the JBL Live Pro+ is an earpiece that connects to the JBL Headphones APP, which is a user interface for controlling functionalities on the earpiece. The earpiece includes the functionality of VoiceAware and Ambient Aware features. The earpiece includes at least three microphones. Upon information and belief, at least one of the microphones operates to capture ambient sound and generate a microphone signal. The earpiece also includes a speaker configured to emit sound, a memory unit to store instructions, and a processor to execute instructions. Upon information and belief when the VoiceAware or Ambient Aware features are activated, the processor receives the microphone signal and detects a sonic signature, such as the user's voice. When the user makes a request (such as activating the VoiceAware feature from the Headphones APP), some portion of the data is sent to the speaker.



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Source: https://www.jbl.com/wireless-earbuds/TOUR-PRO-2.html

153. Defendants have indirectly infringed and continue to indirectly infringe the ’587 Patent by actively inducing infringement of the ’587 Patent by others, such as users of the Accused Products within this District.

154. Defendants’ direct and indirect infringement of the ’587 Patent has damaged and will continue to damage Plaintiff.

155. Plaintiff is entitled to recover damages adequate to compensate it for Defendants’ direct and indirect infringement.

156. After receiving actual notice of the ’587 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the Accused Products.

157. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the ’587 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the ’587 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the ’587 patent.

158. Defendants have willfully infringed and continue to willfully infringe the ’587 Patent.

COUNT IX

Defendants’ Infringement of the ’632 Patent

159. Plaintiff incorporates the allegations of all of the foregoing paragraphs as if fully restated herein.

160. Plaintiff is the assignee and lawful owner of the ’632 Patent, and Plaintiff holds all right, title and interest in and to the ’632 Patent. The ’632 Patent is valid and enforceable.

161. Defendants have directly infringed, and continue to directly and willfully infringe, the ’632 Patent by making, using, selling, offering for sale, or importing into the United States products that infringe the ’632 Patent, such as the Accused Products. Further discovery may reveal additional infringing products and/or models.

162. For example, and without limitation, the Accused Products, including the JBL Tour Pro 2 and similar earphone products, infringe at least claim 1 of the ’632 Patent.

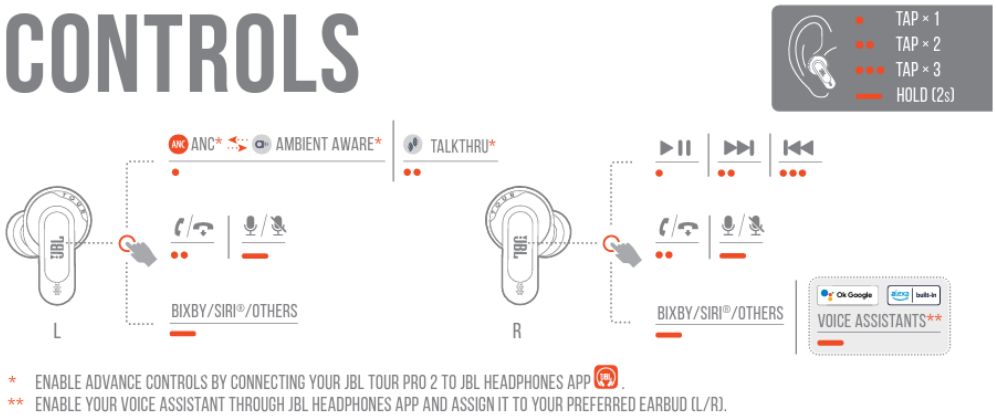
163. Defendants have had notice of their infringement of the ’632 Patent at least since the date of filing of the complaint in this action.

164. The Accused Products are exemplary products that practice at least claim 1 of the ’632 Patent. On information and belief, many other products provided by Harman also infringe claim 1 of the ’632 Patent.

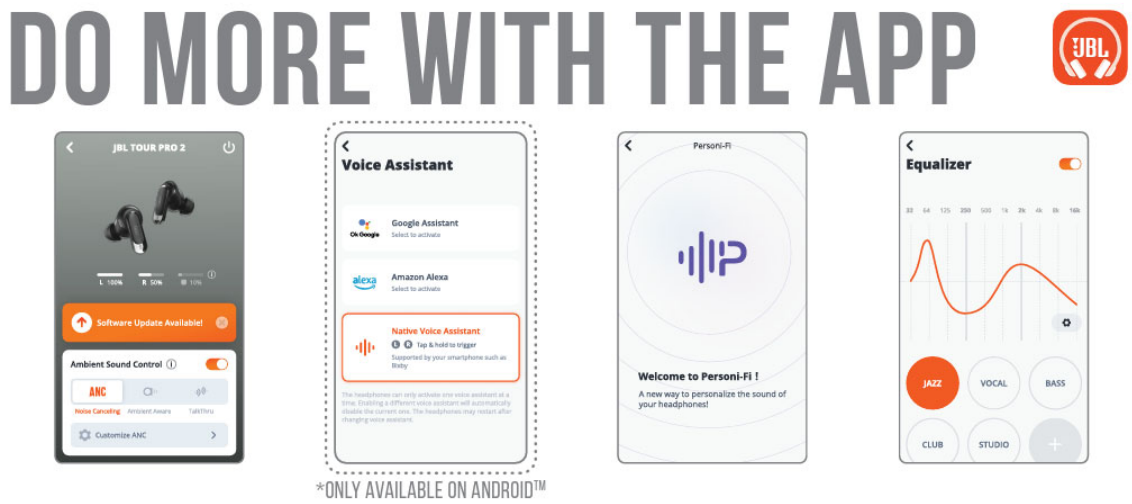
165. The ’632 Patent claims: An earpiece, comprising: a sealing section configured to be inserted into and seal an ear canal of a user; an ambient microphone configured to measure ambient audio content in the ambient environment near the user's ear; an ear canal microphone configured to measure internal audio in the ear canal of the user; a speaker; an interaction element, wherein the interaction element is a tactile interface; a location unit configured to determine the location of the earpiece; a memory that stores instructions; and a logic circuit configured to execute the

instructions to perform operations, the operations comprising: generating active noise reduction using at least one of the measured ambient audio, the internal audio or a combination of both; activating a voice controlled interface when a manual activation of the interaction element is detected; detecting a user's voice from measured ambient audio content; extracting an audio signal of the user's voice; and analyzing the audio signal to determine a voice command.

166. As an example, the JBL Live Pro+ is an earpiece that supports touch and voice control to activate Bixby virtual assistant or other functionalities like Adaptive Noise Cancelling or TalkThru mode. The earpiece contains at least one ambient microphone configured to measure ambient audio content and at least one ear canal microphone configured to measure internal audio. Additionally, the earpiece contains a speaker and a tactile interface that can be pressed to activate functionality and a BlueTooth unit configured to determine the location of the earpiece. Upon information and belief, the earpiece also contains memory that stores instructions and logic circuit configured to execute instructions. When certain functionalities like Adaptive Noise Cancelling are activated, the earpiece uses at least one of the ambient audio, internal audio, or a combination of both to provide the user with reduced distraction and maximized audio quality. Additionally, when the user interacts with the touch surface on the side of the earpiece, the earpiece can detect the user's voice from the measured ambient audio and determine if a voice command was spoken.



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Source: https://www.jbl.com/on/demandware.static/-/Sites-masterCatalog_Harman/default/dw511411ed/pdfs/JBL_TOUR Pro2_QSG Multilingual.pdf

167. Defendants have indirectly infringed and continue to indirectly infringe the '632 Patent by actively inducing infringement of the '632 Patent by others, such as users of the Accused Products within this District.

168. Defendants' direct and indirect infringement of the '632 Patent has damaged and will continue to damage Plaintiff.

169. Plaintiff is entitled to recover damages adequate to compensate it for Defendants' direct and indirect infringement.

170. After receiving actual notice of the '632 Patent, Defendants proceeded to make, use, test, sell, and/or offer to sell in this District and elsewhere in the United States, and import into this District and elsewhere in the United States, the Accused Products.

171. On information and belief, Defendants engaged in such activities despite an objectively high likelihood that their actions constituted infringement and inducement of infringement of the '632 Patent. Defendants knew and should have known that their actions would cause direct and induced infringement of the '632 Patent, and on information and belief, Defendants have monitored the substantial prior art submissions made prior to issuance of the '632 patent.

172. Defendants have willfully infringed and continue to willfully infringe the ’632 Patent.

DEMAND FOR JURY TRIAL

173. Pursuant to Fed. R. Civ. P. 38, Plaintiff hereby respectfully requests trial by jury on all claims and issues so triable.

REQUEST FOR RELIEF

174. WHEREFORE, Plaintiff respectfully requests the following relief:

a) The entry of judgment, in favor Plaintiff and against Defendants, that the Asserted Patents are valid and enforceable;

b) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’780 Patent;

c) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’780 Patent has been willful, and that the Defendants' continued infringement of the ’780 Patent is willful;

d) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’692 Patent;

e) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’692 Patent has been willful, and that the Defendants' continued infringement of the ’692 Patent is willful;

f) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’044 Patent;

g) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’044 Patent has been willful, and that the Defendants' continued infringement of the ’044 Patent is willful;

h) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’966 Patent;

i) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’966 Patent has been willful, and that the Defendants' continued infringement of the ’966 Patent is willful;

j) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’315 Patent;

k) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’315 Patent has been willful, and that the Defendants' continued infringement of the ’315 Patent is willful;

l) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’643 Patent;

m) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’643 Patent has been willful, and that the Defendants' continued infringement of the ’643 Patent is willful;

n) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’493 Patent;

o) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’493 Patent has been willful, and that the Defendants' continued infringement of the ’493 Patent is willful;

p) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’587 Patent;

q) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’587 Patent has been willful, and that the Defendants' continued infringement of the ’587 Patent is willful;

r) The entry of judgment, in favor Plaintiff and against Defendants, that Defendants have infringed, directly and indirectly, either literally or under the Doctrine of Equivalents, one or more claims of the ’632 Patent;

s) The entry of judgment, in favor Plaintiff and against Defendants, that the Defendants' infringement of the ’632 Patent has been willful, and that the Defendants' continued infringement of the ’632 Patent is willful;

t) The entry of judgment, in favor Plaintiff and against Defendants, awarding Plaintiff all appropriate damages under 35 U.S.C. § 284 for Defendants' past infringement, and any continuing or future infringement of the Asserted Patents, including pre- and post-judgment interest, costs, and disbursements as justified under 35 U.S.C. § 284 and, if necessary, an accounting:

   i. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’780 Patent;

   ii. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’692 Patent;

iii. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’044 Patent;

iv. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’966 Patent;

v. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’315 Patent;

vi. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’643 Patent;

vii. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’493 Patent;

viii. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’587 Patent;

ix. That Plaintiff be awarded enhanced damages by reason of the Defendants' willful infringement of the ’632 Patent;

x. That this case be declared exceptional within the meaning of 35 U.S.C. § 285 and that Plaintiff be awarded its reasonable attorneys' fees against the Defendants incurred in prosecuting this action; and

xi. That Plaintiff be awarded its costs and expenses incurred in prosecuting this action.

u) The entry of a permanent injunction, enjoining Defendants, their subsidiaries, and any person or entity acting in concert with Defendants, from making, using, selling, offering to sell, or importing any products that infringe the Asserted Patents, and any other injunctive relief the Court deems just and equitable; and

v) The entry of judgment, in favor of Plaintiff and against Defendants, awarding Plaintiff such further relief at law or in equity as the Court deems just and proper.

ASHBY & GEDDES

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Dated: July 25, 2023

Exhibit A

US008081780B2

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

(10) **Patent No.: US 8,081,780 B2**
(45) **Date of Patent: Dec. 20, 2011**

(54) **METHOD AND DEVICE FOR ACOUSTIC MANAGEMENT CONTROL OF MULTIPLE MICROPHONES**

(75) Inventors: **Steven Wayne Goldstein**, Delray Beach, FL (US); **John Usher**, Montreal (CA); **Marc Andre Boillot**, Plantation, FL (US); **Jason McIntosh**, Sugar Hill, GA (US)

(73) Assignee: **Personics Holdings Inc.**, Boca Raton, FL (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 850 days.

(21) Appl. No.: **12/115,349**

(22) Filed: **May 5, 2008**

(65) **Prior Publication Data**

US 2009/0016541 A1 Jan. 15, 2009

Related U.S. Application Data

(60) Provisional application No. 60/916,271, filed on May 4, 2007.

(51) **Int. Cl.**
H04B 1/00 (2006.01)

(52) **U.S. Cl.** **381/119**; 381/328; 381/56

(58) **Field of Classification Search** 381/119, 381/328, 56
See application file for complete search history.

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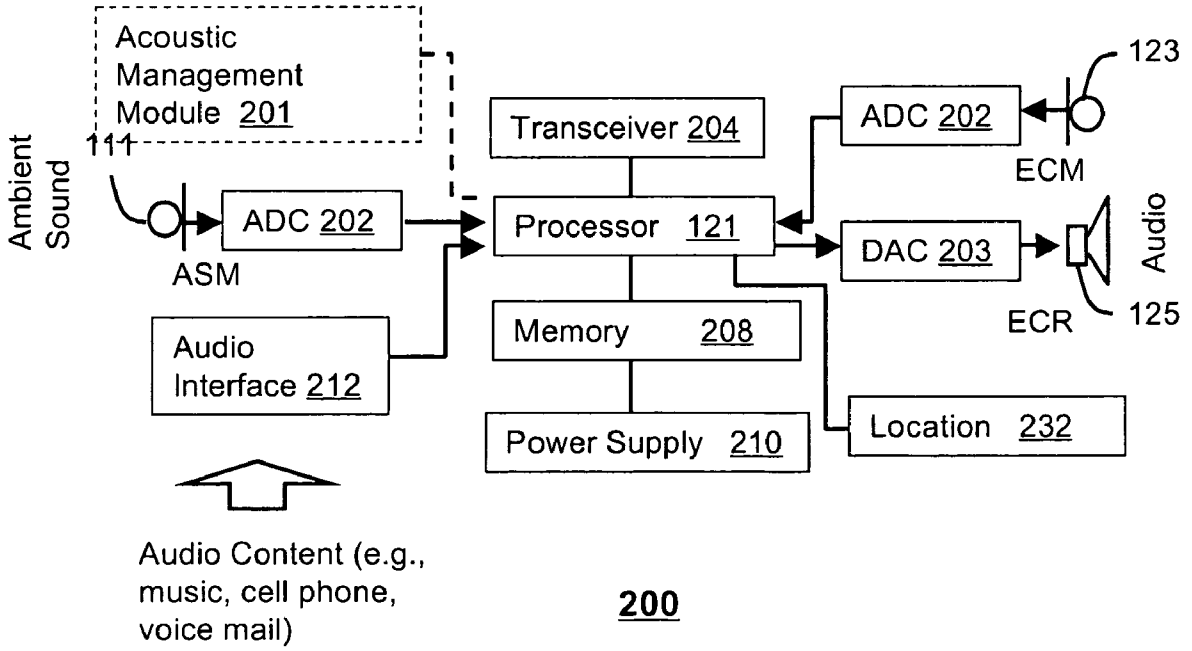
* cited by examiner

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(74) *Attorney, Agent, or Firm* — RatnerPrestia

(57) **ABSTRACT**

An earpiece (**100**) and a method (**640**) for acoustic management of multiple microphones is provided. The method can include capturing an ambient acoustic signal from an Ambient Sound Microphone (ASM) to produce an electronic ambient signal, capturing in an ear canal an internal sound from an Ear Canal Microphone (ECM) to produce an electronic internal signal, measuring a background noise signal, and mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal. The mixing can adjust an internal gain of the electronic internal signal and an external gain of the electronic ambient signal based on the background noise characteristics. The mixing can account for an acoustic attenuation level and an audio content level of the earpiece. Other embodiments are provided.

25 Claims, 8 Drawing Sheets



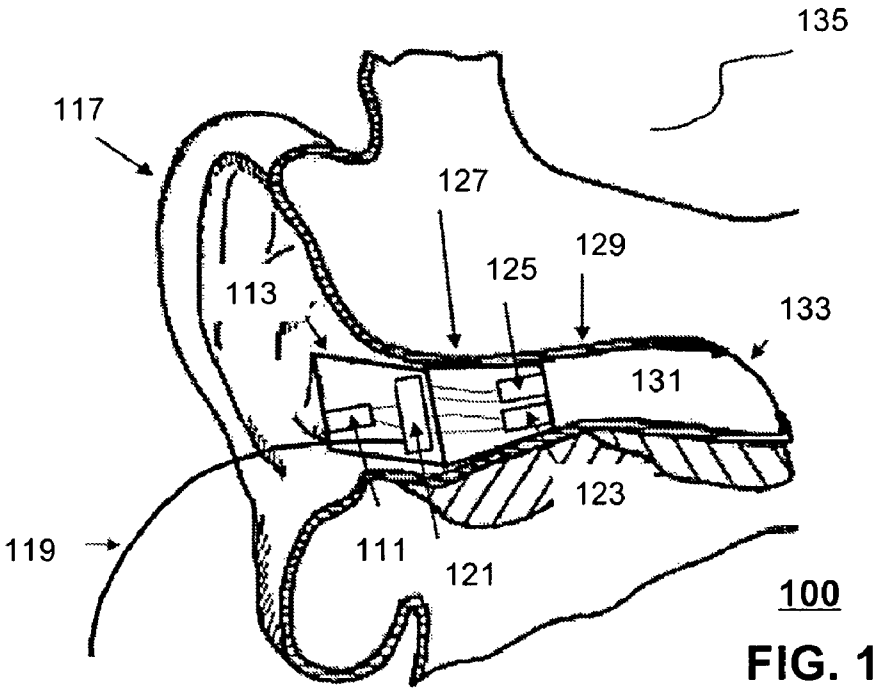

135
117
127
129
125
113
133
131
123
119
111
121
100

FIG. 1

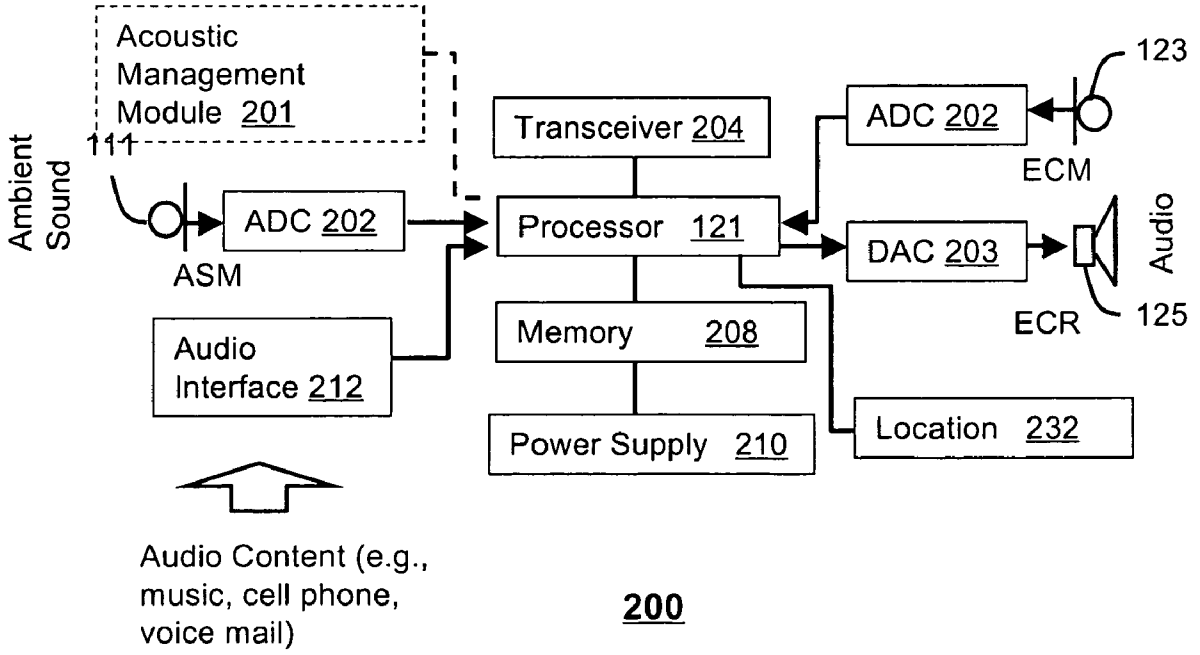
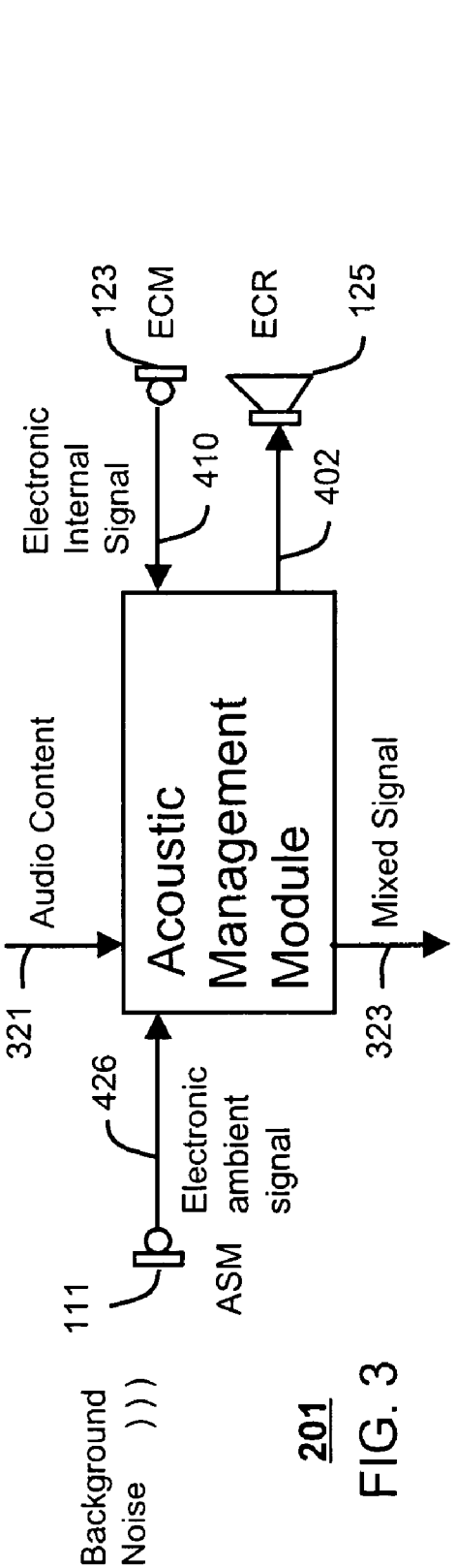

Acoustic Management Module 201
111
Ambient Sound
ADC 202
ASM
Transceiver 204
Processor 121
Memory 208
Power Supply 210
Audio Interface 212
ADC 202
123
ECM
DAC 203
Audio
ECR
125
Location 232
Audio Content (e.g., music, cell phone, voice mail)
200

FIG. 2



201

FIG. 3

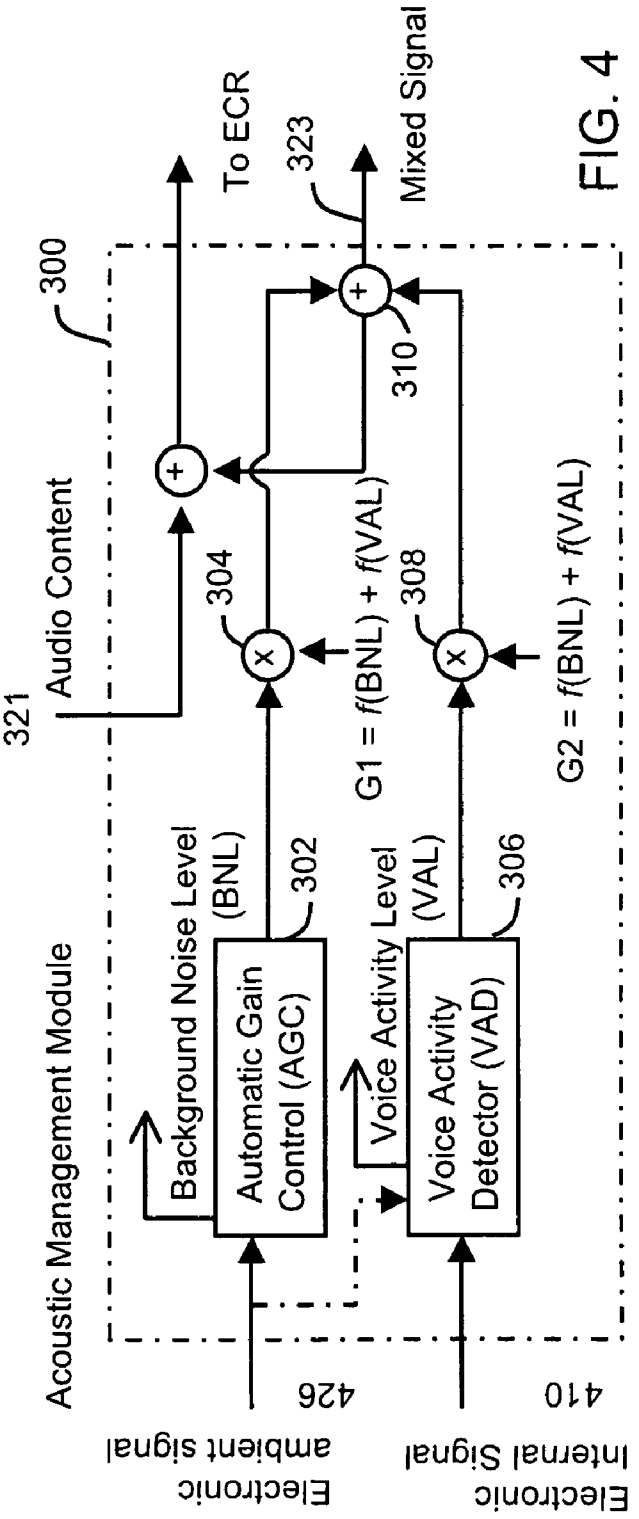


FIG. 4

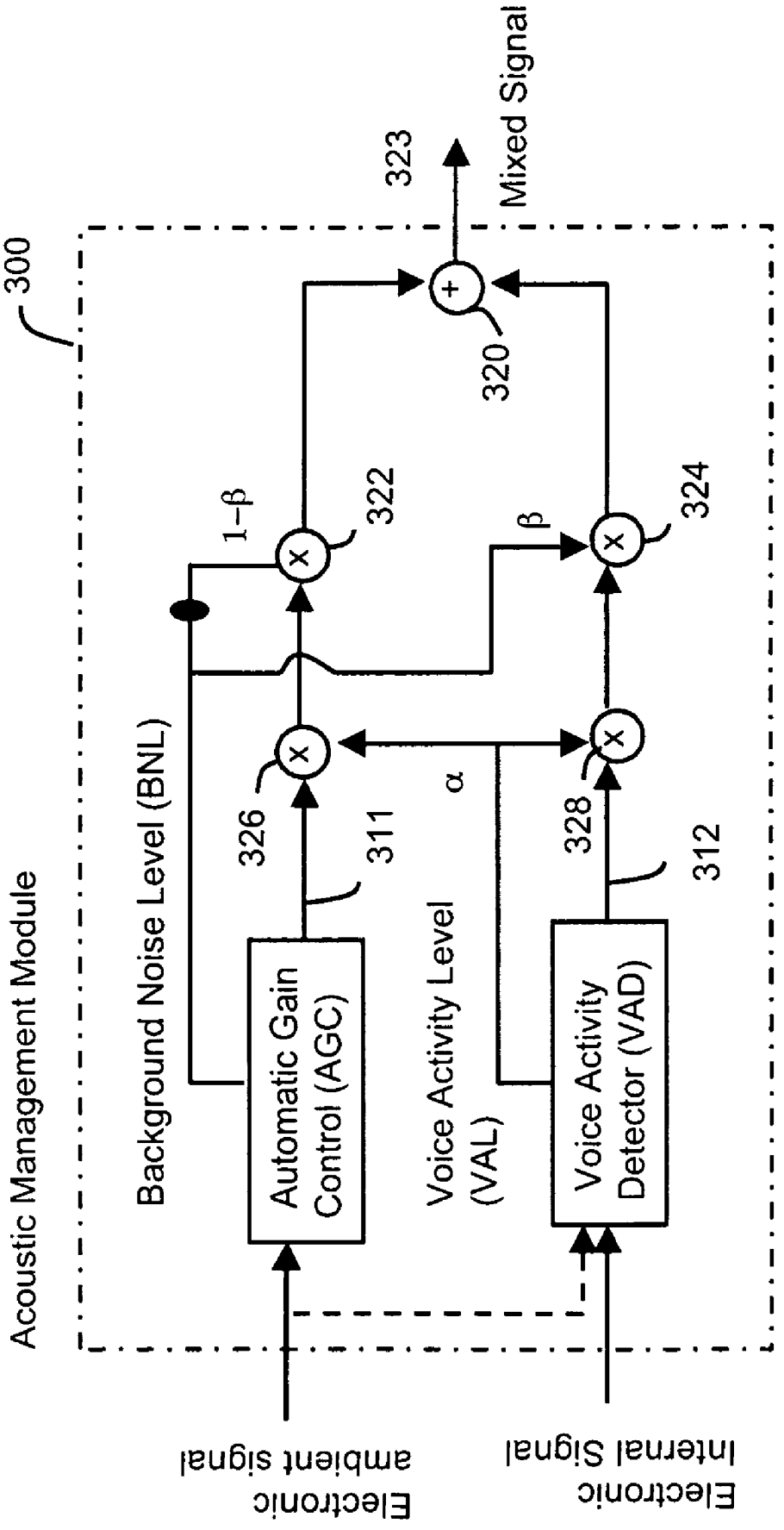

Acoustic Management Module
300
Background Noise Level (BNL)
Voice Activity Level (VAL)
Automatic Gain Control (AGC)
Voice Activity Detector (VAD)
Electronic ambient signal
Electronic Internal Signal
Mixed Signal
323
320
322
324
326
328
311
312
1−β
β
α
+
x

FIG. 5

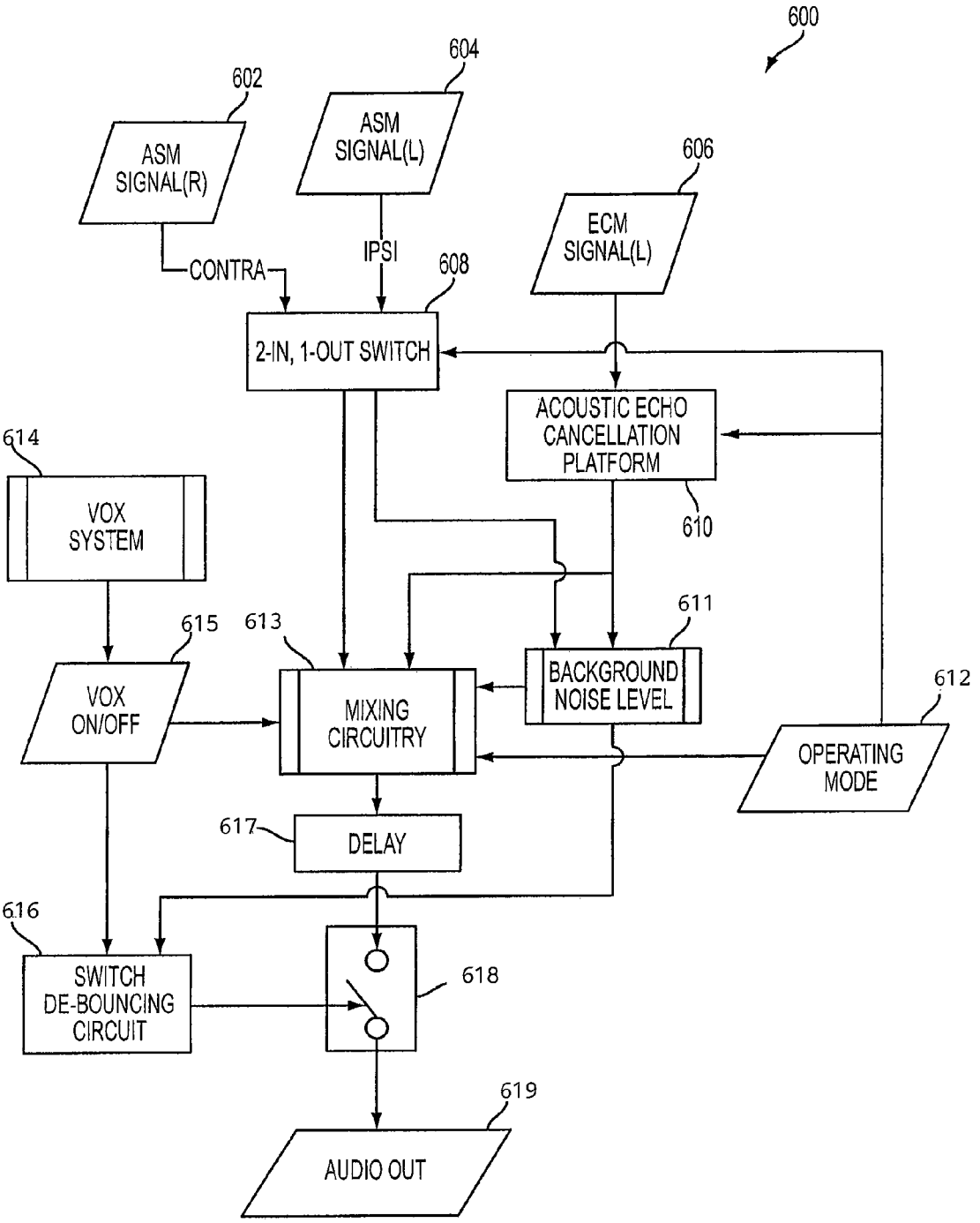

600
602
ASM SIGNAL(R)
604
ASM SIGNAL(L)
606
ECM SIGNAL(L)
CONTRA
IPSI
608
2-IN, 1-OUT SWITCH
ACOUSTIC ECHO CANCELLATION PLATFORM
610
614
VOX SYSTEM
615
VOX ON/OFF
613
MIXING CIRCUITRY
611
BACKGROUND NOISE LEVEL
612
OPERATING MODE
617
DELAY
616
SWITCH DE-BOUNCING CIRCUIT
618
619
AUDIO OUT

FIG. 6

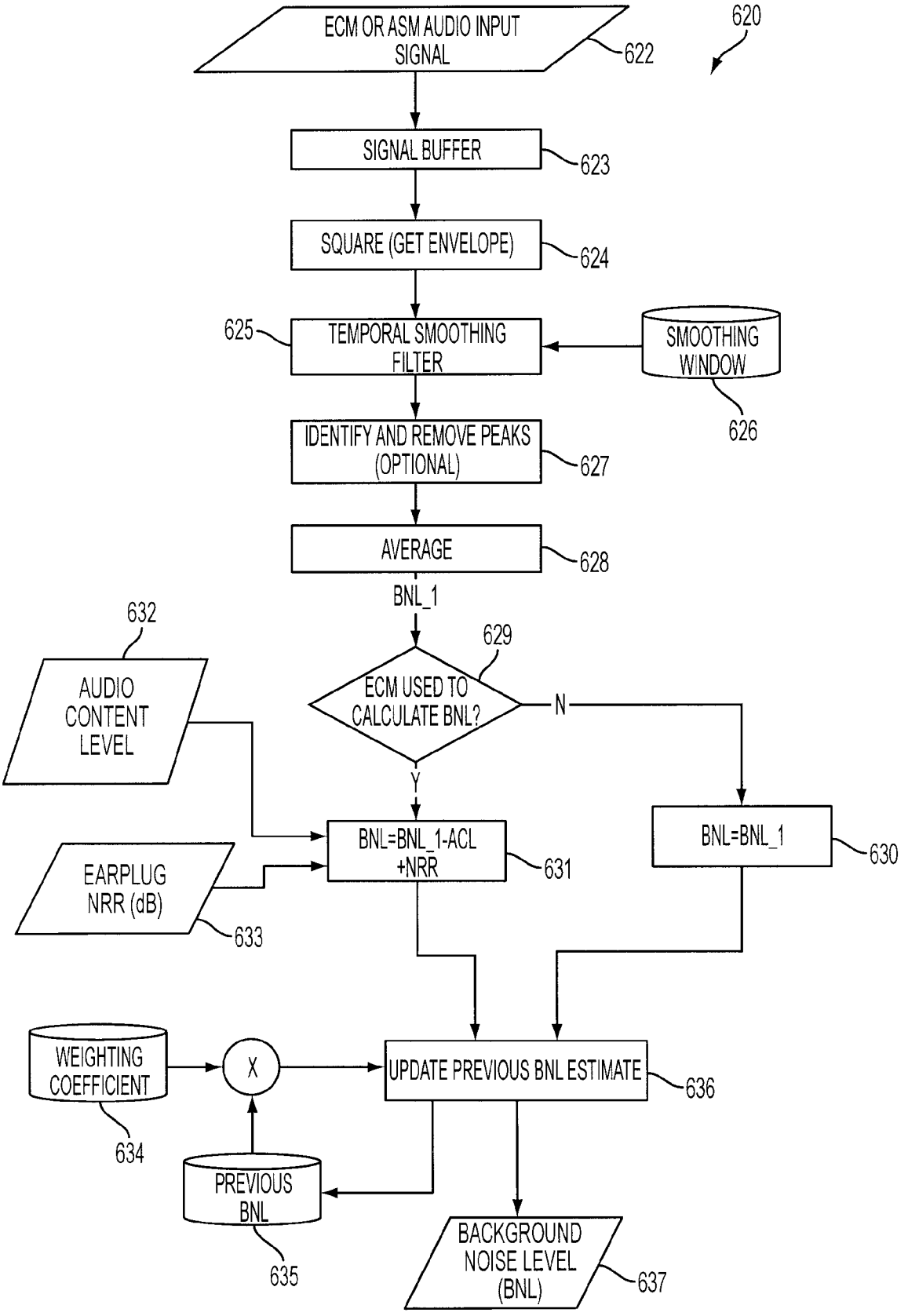

620
ECM OR ASM AUDIO INPUT SIGNAL
622
SIGNAL BUFFER
623
SQUARE (GET ENVELOPE)
624
625
TEMPORAL SMOOTHING FILTER
SMOOTHING WINDOW
626
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
627
AVERAGE
628
BNL_1
632
AUDIO CONTENT LEVEL
629
ECM USED TO CALCULATE BNL?
N
Y
BNL=BNL_1-ACL +NRR
631
BNL=BNL_1
630
EARPLUG NRR (dB)
633
WEIGHTING COEFFICIENT
X
UPDATE PREVIOUS BNL ESTIMATE
636
634
PREVIOUS BNL
635
BACKGROUND NOISE LEVEL (BNL)
637

FIG. 7

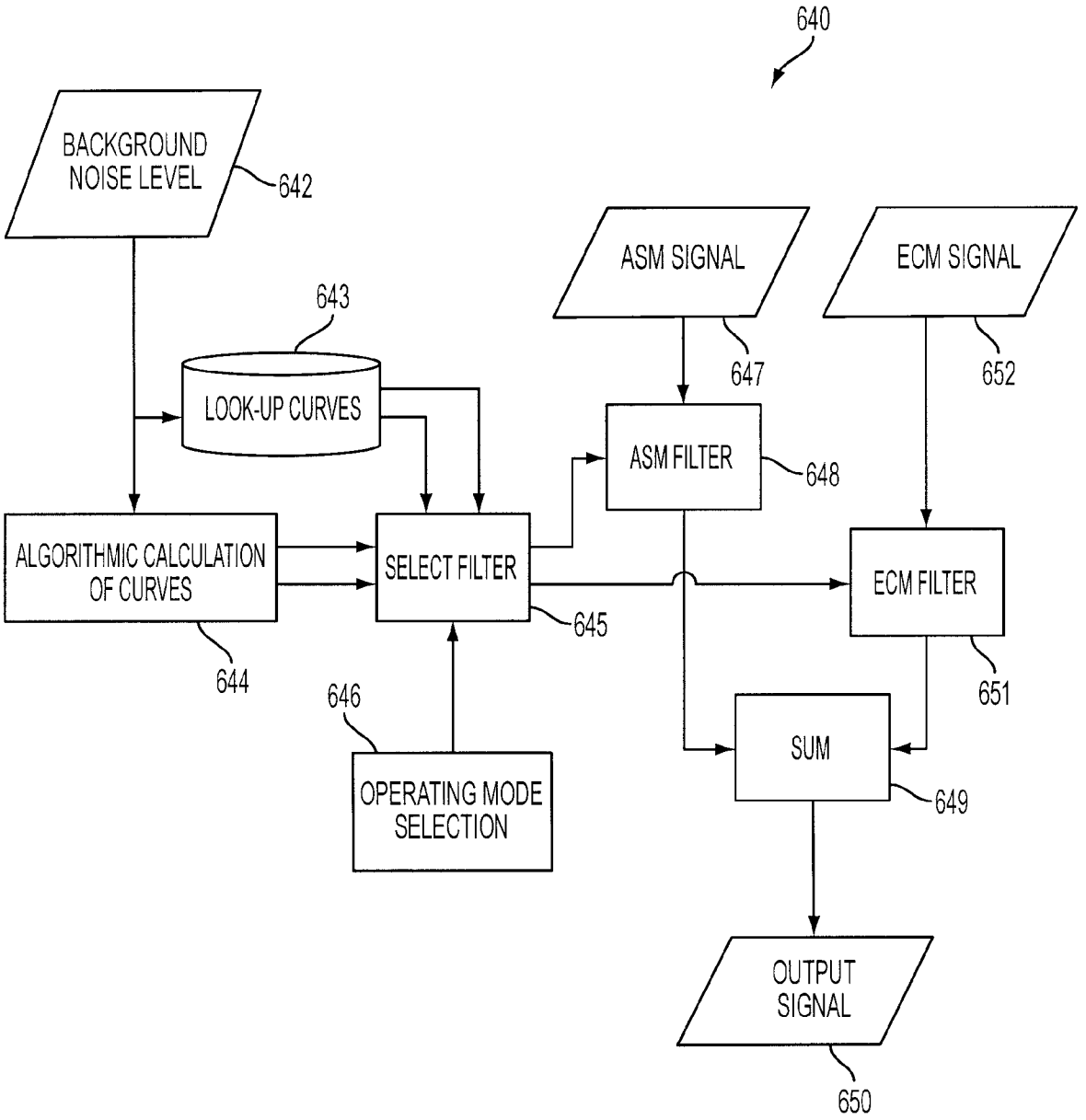

640
BACKGROUND NOISE LEVEL
642
643
LOOK-UP CURVES
ASM SIGNAL
647
ECM SIGNAL
652
ASM FILTER
648
ALGORITHMIC CALCULATION OF CURVES
644
SELECT FILTER
645
ECM FILTER
651
646
OPERATING MODE SELECTION
SUM
649
OUTPUT SIGNAL
650

FIG. 8

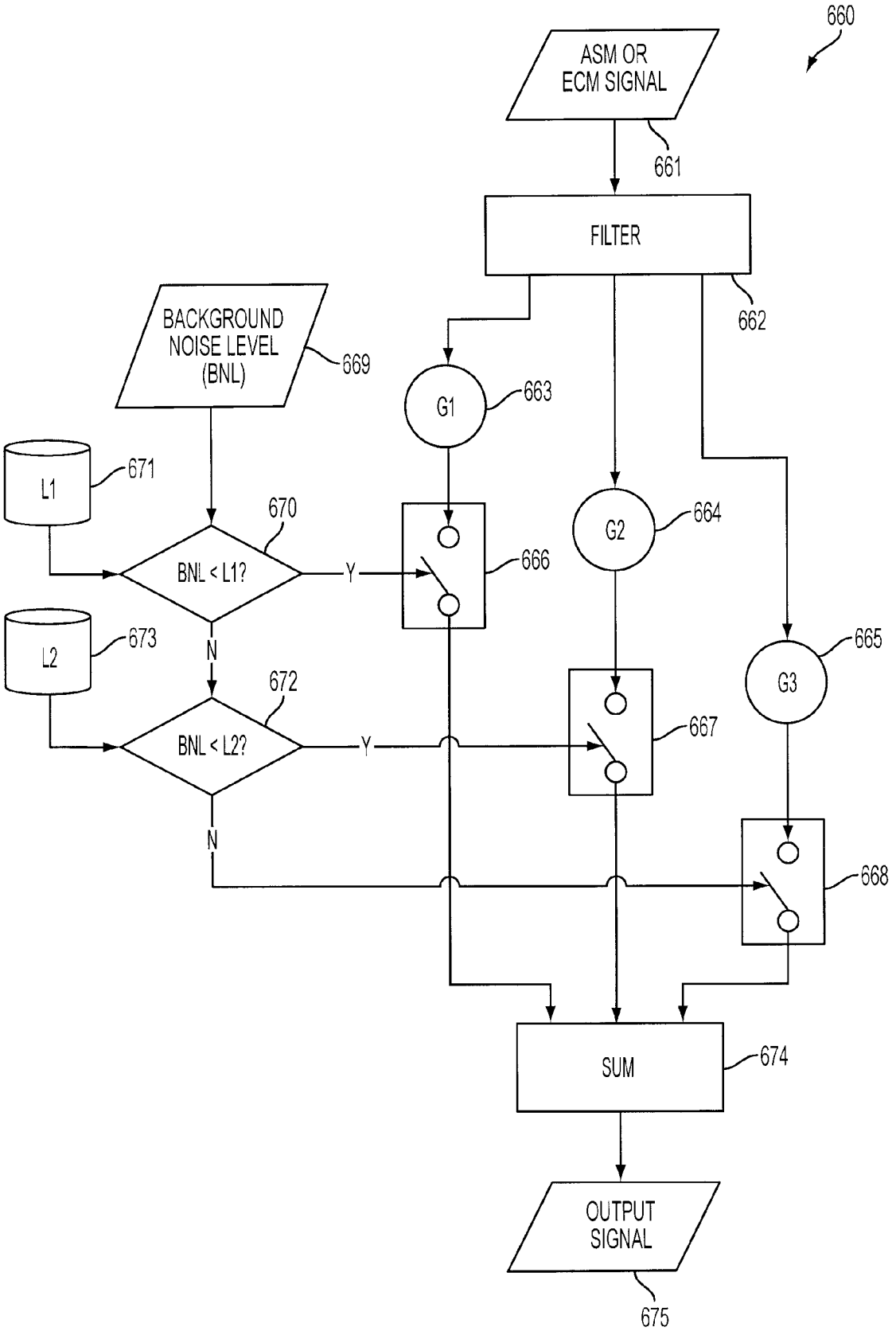
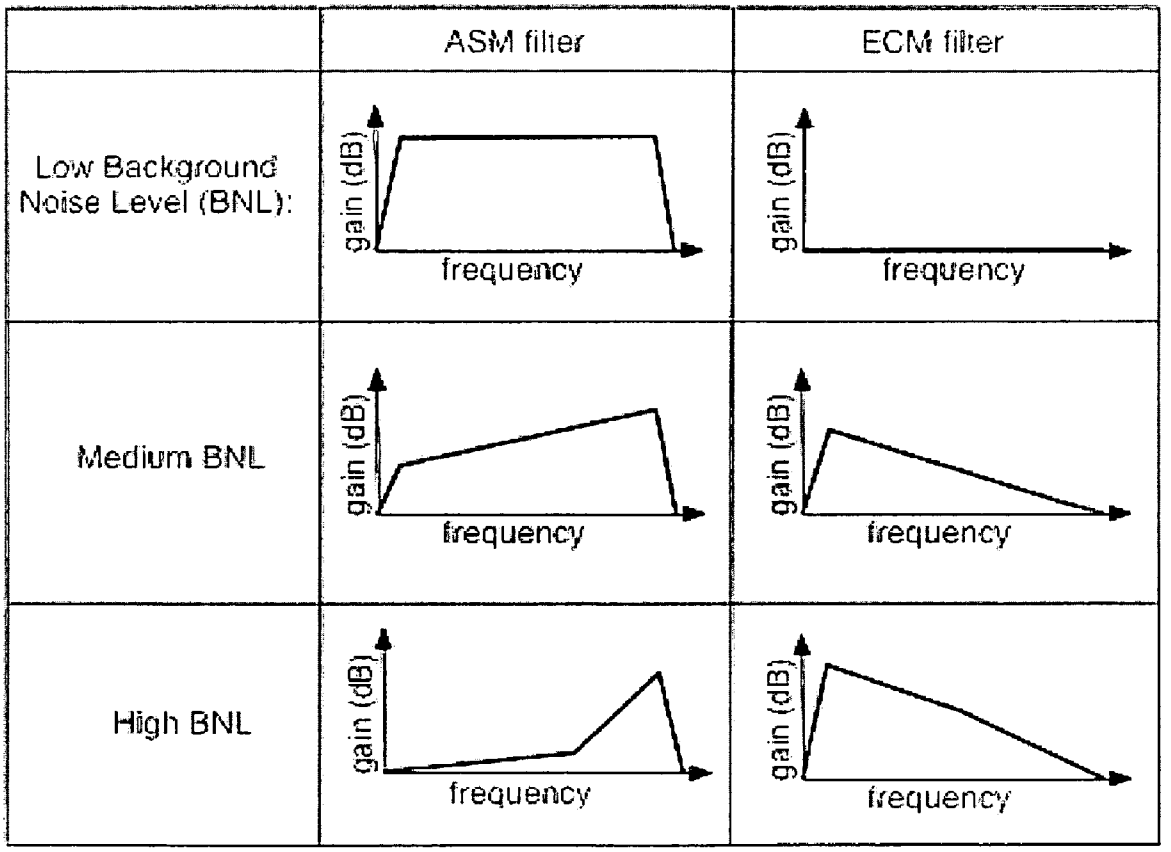

660
ASM OR ECM SIGNAL
661
FILTER
662
BACKGROUND NOISE LEVEL (BNL)
669
G1
663
L1
671
670
BNL < L1?
Y
666
G2
664
L2
673
N
672
BNL < L2?
Y
667
G3
665
N
668
SUM
674
OUTPUT SIGNAL
675

FIG. 9


ASM filter
ECM filter
Low Background Noise Level (BNL):
Medium BNL
High BNL
gain (dB)
frequency

900

FIG. 10

METHOD AND DEVICE FOR ACOUSTIC MANAGEMENT CONTROL OF MULTIPLE MICROPHONES

CROSS REFERENCE TO RELATED APPLICATIONS

This Application is a Non-Provisional and claims the priority benefit of Provisional Application No. 60/916,271 filed on May 4, 2007, the entire disclosure of which is incorporated herein by reference.

FIELD

The present invention pertains to sound reproduction, sound recording, audio communications and hearing protection using earphone devices designed to provide variable acoustical isolation from ambient sounds while being able to audition both environmental and desired audio stimuli. Particularly, the present invention describes a method and device for controlling a voice communication system by monitoring the user's voice with an ambient sound microphone and an ear canal microphone.

BACKGROUND

People use portable communication devices primarily for voice communications and music listening enjoyment. A mobile device or headset generally includes a microphone and a speaker. In noisy conditions, background noises can degrade the quality of the listening experience. Noise suppressors attempt to attenuate the contribution of background noise in order to enhance the listening experience.

In an earpiece, multiple microphones can be used to provide additional noise suppression. A need however exists for acoustic management control of the multiple microphones.

SUMMARY

Embodiments in accordance with the present invention provide a method and device for acoustic management control of multiple microphones.

In a first embodiment, a method for acoustic management control suitable for use in an earpiece can include the steps of capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal, measuring a background noise signal from the electronic ambient signal or the electronic internal signal, and mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal.

The method can include increasing an internal gain of the electronic internal signal while decreasing an external gain of the electronic ambient signal when the background noise levels increase. The method can similarly include decreasing an internal gain of the electronic internal signal while increasing an external gain of the electronic ambient signal when the background noise levels decrease. Frequency weighted selective mixing can also be performed when mixing the signals. The mixing can include filtering the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal, such as a level of the background noise level, a spectral profile, or an envelope fluctuation.

In a second embodiment, a method for acoustic management control suitable for use in an earpiece can include the steps of capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal, detecting a spoken voice signal generated by a wearer of the earpiece from the electronic ambient signal or the electronic internal signal, measuring a background noise level from the electronic ambient signal or the electronic internal signal when the spoken voice signal is not detected, and mixing the electronic ambient signal with the electronic internal signal as a function of the background noise level to produce a mixed signal.

In a third embodiment, an earpiece for acoustic management control can include an Ambient Sound Microphone (ASM) configured to capture ambient sound and produce an electronic ambient signal, an Ear Canal Receiver (ECR) to deliver audio content to an ear canal to produce an acoustic audio content, an Ear Canal Microphone (ECM) configured to capture internal sound in an ear canal and produce an electronic internal signal, and a processor operatively coupled to the ASM, the ECM and the ECR. The processor can be configured to measure a background noise signal from the electronic ambient signal or the electronic internal signal, and mix the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal.

The processor can filter the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal using filter coefficients stored in memory or filter coefficients generated algorithmically. An echo suppressor operatively coupled to the processor can suppress in the mixed signal an echo of spoken voice generated by a wearer of the earpiece when speaking. The processor can also generate a voice activity level for the spoken voice and applies gains to the electronic ambient signal and the electronic internal signal as a function of the background noise level and the voice activity level.

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 block diagram for an acoustic management module in accordance with an exemplary embodiment;

FIG. **4** is a schematic for the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal as a function of a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **5** is a more detailed schematic of the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **6** is a block diagram of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **7** is a block diagram of a method for calculating background noise levels in accordance with an exemplary embodiment;

FIG. **8** is a block diagram for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment;

FIG. **9** is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment; and

FIG. **10** is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL) 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.

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.

Various embodiments herein provide a method and device for automatically mixing audio signals produced by a pair of microphone signals that monitor a first ambient sound field and a second ear canal sound field, to create a third new mixed signal. An Ambient Sound Microphone (ASM) and an Ear Canal Microphone (ECM) can be housed in an earpiece that forms a seal in the ear of a user. The third mixed signal can be auditioned by the user with an Ear Canal Receiver (ECR) mounted in the earpiece, which creates a sound pressure in the occluded ear canal of the user. Alternatively, or additionally, the third mixed signal can be transmitted to a remote voice communications system, such as a mobile phone, personal media player, recording device, walkie-talkie radio, etc. Before the ASM and ECM signals are mixed, they can be subjected to different filters and at optional additional gains.

The characteristic responses of the ASM and ECM filter can differ based on characteristics of the background noise. In some exemplary embodiments, the filter response can depend on the measured Background Noise Level (BNL). A gain of a filtered ASM and a filtered ECM signal can also depend on the BNL. The (BNL) can be calculated using either or both the conditioned ASM and/or ECM signal(s). The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and can be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level (i.e. the high and low frequencies are attenuated before the level of the microphone signals are calculated).

For example, at low BNLs (e.g. <60 dBA), the ECM signal can be attenuated relative to the ASM signal. At medium BNL, a mixture of the ASM and ECM signals can be performed. Moreover the ASM filter can attenuate low frequencies of the ASM signal, and the ECM filter can attenuate high frequencies of the ECM signal. At high BNL (e.g. >85 dB), the ASM filter can attenuate low frequencies of the ASM signal, and the ECM filter can attenuate high frequencies of the ECM signal. In another embodiment, the ASM and ECM filters can be adjusted by the spectral profile of the background noise measurement. For instance, if there is a large Low Frequency noise in the ambient sound field of the user, then the ASM filter can attenuate the low-frequencies of the ASM signal, and boost the low-frequencies of the ECM signal using the ECM filter.

At least one exemplary embodiment of the invention is directed to an earpiece for voice operated control. Reference is made to FIG. **1** in which an earpiece device, generally indicated as earpiece **100**, is constructed and operates 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 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 **131**. This seal is also a basis for a 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 or partially 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 the earpiece **100**. In one arrangement, the ASM **111** can be housed in the in-the-ear assembly **113** to monitor 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**.

The earpiece **100** can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece **100** in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning

sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

The earpiece **100** can generate an Ear Canal Transfer Function (ECTF) to model the ear canal **131** using ECR **125** and ECM **123**, as well as an Outer Ear Canal Transfer function (OETF) using ASM **111**. For instance, the ECR **125** can deliver an impulse within the ear canal and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal. The earpiece **100** can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece **100** to safely administer and monitor sound exposure to the ear.

Referring to FIG. **2**, a block diagram **200** of the earpiece **100** in accordance with an exemplary embodiment is shown. As illustrated, the earpiece **100** can include the processor **121** 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 **121** 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 Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device **100**. The processor **121** can also include a clock to record a time stamp.

As illustrated, the earpiece **100** can include an acoustic management module **201** to mix sounds captured at the ASM **111** and ECM **123** to produce a mixed signal. The processor **121** can then provide the mixed signal to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor or communication device. The acoustic management module **201** can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the functionality of the acoustic management module **201** can be provided by way of software, such as program code, assembly language, or machine language.

The earpiece **100** can measure ambient sounds in the environment 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.

The memory **208** can also store program instructions for execution on the processor **206** as well as captured audio processing data and filter coefficient data. The memory **208** can be off-chip and external to the processor **121**, and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor **121** to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

The earpiece **100** can include an audio interface **212** operatively coupled to the processor **121** and acoustic management module **201** to receive audio content, for example from a media player, cell phone, or any other communication device, and deliver the audio content to the processor **121**. The processor **121** responsive to detecting spoken voice from the acoustic management module **201** can adjust the audio content delivered to the ear canal. For instance, the processor **121** (or acoustic management module **201**) can lower a volume of the audio content responsive to detecting a spoken voice. The processor **121** by way of the ECM **123** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the acoustic management module **201**.

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 location receiver **232** can utilize common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece **100**.

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 **121** 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 block diagram of the acoustic management module **201** in accordance with an exemplary embodiment. Briefly, the Acoustic management module **201** facilitates monitoring, recording and transmission of user-generated voice (speech) to a voice communication system. User-generated sound is detected with the ASM **111** that monitors a sound field near the entrance to a user's ear, and with the ECM **123** that monitors a sound field in the user's occluded ear canal. A new mixed signal **323** is created by filtering and mixing the ASM and ECM microphone signals. The filtering and mixing process is automatically controlled depending on the background noise level of the ambient sound field to enhance intelligibility of the new mixed signal **323**. For instance, when the background noise level is high, the acoustic management module **201** automatically increases the level of the ECM **123** signal relative to the level of the ASM **111** to create the new mixed signal **323**.

As illustrated, the ASM **111** is configured to capture ambient sound and produce an electronic ambient signal **426**, the ECR **125** is configured to pass, process, or play acoustic audio content **402** (e.g., audio content **321**, mixed signal **323**) to the ear canal, and the ECM **123** is configured to capture internal sound in the ear canal and produce an electronic internal signal **410**. The acoustic management module **201** is configured to measure a background noise signal from the electronic ambient signal **426** or the electronic internal signal **410**, and mix the electronic ambient signal **426** with the electronic

internal signal **410** in a ratio dependent on the background noise signal to produce the mixed signal **323**. The acoustic management module **201** filters the electronic ambient signal **426** and the electronic internal **410** signal based on a characteristic of the background noise signal using filter coefficients stored in memory or filter coefficients generated algorithmically.

In practice, the acoustic management module **201** mixes sounds captured at the ASM **111** and the ECM **123** to produce the mixed signal **323** based on characteristics of the background noise in the environment such as a level of the background noise level, a spectral profile, or an envelope fluctuation. In noisy ambient environments, the voice captured at the ASM **111** includes the background noise from the environment, whereas, the internal voice created in the ear canal **131** captured by the ECM **123** has less noise artifacts, since the noise is blocked due to the occlusion of the earpiece **100** in the ear. It should be however noted that the background noise can enter the ear canal if the earpiece **100** is not completely sealed. Accordingly, the acoustic management module **201** monitors the electronic internal signal **410** for background noise (e.g., spectral comparison with the electronic ambient signal). It should also be noted that voice generated by a user of the earpiece **100** is captured at both the external ASM **111** and the internal ECM **123**.

At low background noise levels, the acoustic management module **201** amplifies the electronic ambient signal **426** from the ASM **111** relative to the electronic internal signal **410** from the ECM **123** in producing the mixed signal **323**. At medium background noise levels, the acoustic management module **201** attenuates low frequencies in the electronic ambient signal **426** and attenuates high frequencies in the electronic internal signal **410**. At high background noise levels, the acoustic management module **201** amplifies the electronic internal signal **410** from the ECM **123** relative to the electronic ambient signal **426** from the ASM **111** in producing the mixed signal. As will be discussed ahead, the acoustic management module **201** can additionally apply frequency specific filters (see FIG. **10**) based on the characteristics of the background noise.

FIG. **4** is a schematic **300** of the acoustic management module **201** illustrating a mixing of the electronic ambient signal **426** with the electronic internal signal **410** as a function of a background noise level (BNL) and a voice activity level (VAL) in accordance with an exemplary embodiment. As illustrated, the acoustic management module **201** includes an Automatic Gain Control (AGC) **302** to measure background noise characteristics. The acoustic management module **201** also includes a Voice Activity Detector (VAD) **306**. The VAD **306** can analyze either or both the electronic ambient signal **426** and the electronic internal signal **410** to estimate the VAL. As an example, the VAL can be a numeric range such as 0 to 10 indicating a degree of voicing. For instance, a voiced signal can be predominately periodic due to the periodic vibrations of the vocal cords. A highly voiced signal (e.g., vowel) can be associated with a high level, and a non-voiced signal (e.g., fricative, plosive, consonant) can be associated with a lower level.

The acoustic management module **201** includes a first gain (G**1**) **304** applied to the AGC processed electronic ambient signal **426**. A second gain (G**2**) **308** is applied to the VAD processed electronic internal signal **410**. The acoustic management module **201** applies the first gain (G**1**) **304** and the second gain (G**2**) **308** as a function of the background noise level and the voice activity level to produce the mixed signal **323**, where

$$G1=f(BNL)+f(VAL) \text{ and } G2=f(BNL)+f(VAL)$$

As illustrated, the mixed signal is the sum **310** of the G**1** scaled electronic ambient signal and the G**2** scaled electronic internal signal. The mixed signal **323** can then be transmitted to a second communication device (e.g. second cell phone, voice recorder, etc.) to receive the enhanced voice signal. The acoustic management module **201** can also play the mixed signal **323** back to the ECR for loopback listening. The loopback allows the user to hear himself or herself when speaking, as though the earpiece **100** and associated occlusion effect were absent. The loopback can also be mixed with the audio content **321** based on the background noise level, the VAL, and audio content level. The acoustic management module **201** can also account for an acoustic attenuation level of the earpiece, and account for the audio content level reproduced by the ECR when measuring background noise characteristics.

FIG. **5** is a more detailed schematic of the acoustic management module **201** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment. In particular, the gain blocks for G**1** and G**2** of FIG. **4** are a function of the BNL and the VAL and are shown in greater detail. As illustrated, the AGC produces a BNL that can be used to set a first gain **322** for the processed electronic ambient signal **311** and a second gain **324** for the processed electronic internal signal **312**. For instance, when the BNL is low (<70 dBA), gain **322** is set higher relative to gain **324** so as to amplify the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. When the BNL is high (>85 dBA), gain **322** is set lower relative to gain **324** so as to attenuate the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. The mixing can be performed in accordance with the relation:

$$\text{Mixed signal}=(1-\beta)^{*}\text{electronic ambient signal}+(\beta)^{*}\text{electronic internal signal}$$

where $(1-\beta)$ is an external gain, (β) is an internal gain, and the mixing is performed with $0<\beta<1$.

As illustrated, the VAD produces a VAL that can be used to set a third gain **326** for the processed electronic ambient signal **311** and a fourth gain **328** for the processed electronic internal signal **312**. For instance, when the VAL is low (e.g., 0-3), gain **326** and gain **328** are set low so as to attenuate the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is not detected. When the VAL is high (e.g., 7-10), gain **326** and gain **328** are set high so as to amplify the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is detected.

The gain scaled processed electronic ambient signal **311** and the gain scaled processed electronic internal signal **312** are then summed at adder **320** to produce the mixed signal **323**. The mixed signal **323**, as indicated previously, can be transmitted to another communication device, or as loopback to allow the user to hear his or her self.

FIG. **6** is a block diagram **600** of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment.

As illustrated the mixing circuitry **613** (shown in center) receives an estimate of the background noise level **611** for mixing either or both the right earpiece ASM signal **602** and

the left earpiece ASM signal **604** with the left earpiece ECM signal **606**. (The right earpiece ECM signal can be used similarly.) An operating mode **612** selects a switching **608** (e.g., 2-in, 1-out) between the left earpiece ASM signal **604** and the right earpiece ASM signal **602**. As indicated earlier, the ASM signals and ECM signals can be first amplified with a gain system and then filtered with a filter system (the filtering may be accomplished using either analog or digital electronics). The audio input signals **602**, **604**, **606** are therefore taken after this gain and filtering process.

The Acoustic Echo Cancellation (AEC) system **610** can be activated with the operating mode selection system **612** when the mixed signal audio output **619** is reproduced with the ECR **125** in the same ear as the ECM **123** signal used to create the mixed signal audio output **619**. The acoustic echo cancellation platform **610** can also suppress an echo of a spoken voice generated by the wearer of the earpiece **100**. This ensures against acoustic feedback ("howlback").

The Voice Activated System (VOX) **614** in conjunction with a de-bouncing circuit **616** activates the electronic switch **618** to control the mixed signal output **619** from the mixing circuitry **613**; the mixed signal is a combination of the left ASM signal **604** or right ASM signal **602**, with the left ECM **606** signal. Though not shown, the same arrangement applies for the other earphone device for the right ear, if present. In a contra-lateral operating mode, as selected by operating mode selection system **612**, the ASM and ECM signal are taken from opposite earphone devices, and the mix of these signals is reproduced with the ECR in the earphone that is contra-lateral to the ECM signal, and the same as the ASM signal.

For instance, in the contra-lateral operating mode, the ASM signal from the Right earphone device is mixed with the ECM signal from the left earphone device, and the audio signal corresponding to a mix of these two signals is reproduced with the Ear Canal Receiver (ECR) in the Right earphone device. The mixed signal audio output **619** therefore contains a mix of the ASM and ECM signals when the user's voice is detected by the VOX. This mixed signal audio output can be used in loopback as a user Self-Monitor System to allow the user to hear their own voice as reproduced with the ECR **125**, or it may be transmitted to another voice system, such as a mobile phone, walkie-talkie radio etc. The VOX system **614** that activates the switch **618** may be one a number of VOX embodiments.

In a particular operating mode, specified by unit **612**, the conditioned ASM signal is mixed with the conditioned ECM signal with a ratio dependant on the BNL using audio signal mixing circuitry and the method described in either FIG. **8** or FIG. **9**. As the BNL increases, then the ASM signal is mixed with the ECM signal with a decreasing level. When the BNL is above a particular value, then a minimal level of the ASM signal is mixed with the ECM signal. When the VOX switch **618** is active, the mixed ASM and ECM signals are then sent to mixed signal output **619**. The switch de-bouncing circuit **616** ensures against the VOX **614** rapidly closing on and off (sometimes called chatter). This can be achieved with a timing circuit using digital or analog electronics. For instance, with a digital system, once the VOX has been activated, a time starts to ensure that the switch **618** is not closed again within a given time period, e.g. 100 ms. The delay unit **617** can improve the sound quality of the mixed signal audio output **619** by compensating for any latency in voice detection by the VOX system **614**. In some exemplary embodiments, the switch debouncing circuit **616** can be dependent by the BNL. For instance, when the BNL is high (e.g. above 85 dBA), the de-bouncing circuit can close the switch **618** sooner after the VOX output **615** determines that no user speech (e.g. spoken voice) is present.

FIG. **7** is a block diagram of a method **620** for calculating background noise levels in accordance with an exemplary embodiment. Briefly, the background noise levels can be calculated according to different contexts, for instance, if the user is talking while audio content is playing, if the user is talking while audio content is not playing, if the user is not talking but audio content is playing, and if the user is not talking and no audio content is playing. For instance, the system takes as its inputs either the ECM or ASM signal, depending on the particular system configuration. If the ECM signal is used, then the measured BNL accounts for an acoustic attenuation of the earpiece and a level of reproduced audio content.

As illustrated, modules **622-628** provide exemplary steps for calculating a base reference background noise level. The ECM or ASM audio input signal **622** can be buffered **623** in real-time to estimate signal parameters. An envelope detector **624** can estimate a temporal envelope of the ASM or ECM signal. A smoothing filter **625** can minimize abruptions in the temporal envelope. (A smoothing window **626** can be stored in memory). An optional peak detector **627** can remove outlier peaks to further smooth the envelope. An averaging system **628** can then estimate the average background noise level (BNL_**1**) from the smoothed envelope.

If at step **629**, it is determined that the signal from the ECM was used to calculate the BNL_**1**, an audio content level **632** (ACL) and noise reduction rating **633** (NRR) can be subtracted from the BNL_**1** estimate to produce the updated BNL **631**. This is done to account for the audio content level reproduced by the ECR **125** that delivers acoustic audio content to the earpiece **100**, and to account for an acoustic attenuation level (i.e. Noise Reduction Rating **633**) of the earpiece. For example, if the user is listening to music, the acoustic management module **201** takes into account the audio content level delivered to the user when measuring the BNL. If the ECM is not used to calculate the BNL at step **629**, the previous real-time frame estimate of the BNL **630** is used.

At step **636**, the acoustic management module **201** updates the BNL based on the current measured BNL and previous BNL measurements **635**. For instance, the updated BNL **637** can be a weighted estimate **634** of previous BNL estimates according to BNL=2*previous BNL+(1−w)*current BNL, where 0<W<1. The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and may be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level.

FIG. **8** is a block diagram **640** for mixing an external microphone signal with an internal microphone signal based on a background noise level to produce a mixed output signal in accordance with an exemplary embodiment. The block diagram can be implemented by the acoustic management module **201** or the processor **121**. In particular, FIG. **8** primarily illustrates the selection of microphone filters based on the background noise level. The microphone filters are used to condition the external and internal microphone signals before mixing.

As shown, the filter selection module **645** can select one or more filters to apply to the microphone signals before mixing. For instance, the filter selection module **645** can apply an ASM filter **648** to the ASM signal **647** and an ECM filter **651** to the ECM signal **652** based on the background noise level **642**. The ASM and ECM filters can be retrieved from memory based on the characteristics of the background noise. An operating mode **646** can determine whether the ASM and

ECM filters are look-up curves **643** from memory or filters whose coefficients are determined in real-time based on the background noise levels.

Prior to mixing with summing unit **649** to produce output signal **650**, the ASM signal **647** is filtered with ASM filter **648**, and the ECM signal **652** is filtered with ECM filter **651**. The filtering can be accomplished by a time-domain transversal filter (FIR-type filter), an IIR-type filter, or with frequency-domain multiplication. The filter can be adaptive (i.e. time variant), and the filter coefficients can be updated on a frame-by-frame basis depending on the BNL. The filter coefficients for a particular BNL can be loaded from computer memory using pre-defined filter curves **643**, or can be calculated using a predefined algorithm **644**, or using a combination of both (e.g. using an interpolation algorithm to create a filter curve for both the ASM filter **648** and ECM filter **651** from predefined filters).

Examples of filter response curves for three different BNL are shown in FIG. **10**, which is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL).

The basic trend for the ASM and ECM filter response at different BNLs is that at low BNLs (e.g. <60 dBA), the ASM signal is primarily used for voice communication. At medium BNL; ASM and ECM are mixed in a ratio depending on the BNL, though the ASM filter can attenuate low frequencies of the ASM signal, and attenuate high frequencies of the ECM signal. At high BNL (e.g. >85 dB), the ASM filter attenuates most al the low frequencies of the ASM signal, and the ECM filter attenuates most all the high frequencies of the ECM signal. In another embodiment of the Acoustic Management System, the ASM and ECM filters may be adjusted by the spectral profile of the background noise measurement. For instance, if there is a large Low Frequency noise in the ambient sound field of the user, then the ASM filter can reduce the low-frequencies of the ASM signal accordingly, and boost the low-frequencies of the ECM signal using the ECM filter.

FIG. **9** is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment.

In particular, FIG. **9** shows a method **660** for the filtering of the ECM and ASM signals using analog electronic circuitry prior to mixing. The analog circuit can process both the ECM and ASM signals in parallel; that is, the analog components apply to both the ECM and ASM signals. In one exemplary embodiment, the input audio signal **661** (e.g., ECM signal, ASM signal) is first filtered with a fixed filter **662**. The filter response of the fixed filter **662** approximates a low-pass shelf filter when the input signal **661** is an ECM signal, and approximates a high-pass filter when the input signal **661** is an ASM signal. In an alternate exemplary embodiment, the filter **662** is a unity-pass filter (i.e. no spectral attenuation) and the gain units G**1**, G**2** etc instead represent different analog filters. As illustrated, the gains are fixed, though they may be adapted in other embodiments. Depending on the BNL **669**, the filtered signal is then subjected to one of three gains; G**1** **663**, G**2** **664**, or G**3** **665**. (The analog circuit can include more or less than the number of gains shown.)

For low BNLs (e.g. when BNL<L**1670**, where L**1** is a predetermined level threshold **671**), a G**1** is determined for both the ECM signal and the ASM signal. The gain G**1** for the ECM signal is approximately zero; i.e. no ECM signal would be present in the output signal **675**. For the ASM input signal, G**1** would be approximately unity for low BNL.

For medium BNLs (e.g. when BNL<L**2** **672**, where L**2** is a predetermined level threshold **673**), a G**2** is determined for both the ECM signal and the ASM signal. The gain G**2** for the ECM signal and the ASM signal is approximately the same. In another embodiment, the gain G**2** can be frequency dependent so as to emphasize low frequency content in the ECM and emphasize high frequency content in the ASM signal in the mix. For high BNL; G**3** **665** is high for the ECM signal, and low for the ASM signal. The switches **666**, **667**, and **668** ensure that only one gain channel is applied to the ECM signal and ASM signal. The gain scaled ASM signal and ECM signal are then summed at junction **674** to produce the mixed output signal **675**.

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 acoustic management control suitable for use in an earpiece, the method comprising the steps of:

capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal;

capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal;

measuring a background noise signal from the electronic ambient signal or the electronic internal signal; and

mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal.

2. The method of claim **1**, comprising

increasing an internal gain of the electronic internal signal as background noise levels increase, while

decreasing an external gain of the electronic ambient signal as the background noise levels increase.

3. The method of claim **1**, comprising

decreasing an internal gain of the electronic internal signal as background noise levels decrease, while

increasing an external gain of the electronic ambient signal as the background noise levels decrease.

4. The method of claim **1**, where the step of mixing includes filtering the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal,

where the characteristic is a level of a background noise level, a spectral profile, or an envelope fluctuation.

5. The method of claim **4**, wherein the filtering is performed by a High-Pass Filter for the electronic ambient signal and a Low-Pass Filter for the electronic internal signal.

6. The method of claim **4**, where filter coefficients for a particular background noise level or a particular spectral profile are loaded from a memory containing pre-defined filter curves.

7. The method of claim **4**, where filter coefficients are algorithmically determined for a particular background noise level or a particular spectral profile.

8. The method of claim 4, comprising

at low background noise levels, amplifying the electronic ambient signal from the ASM relative to the electronic internal signal from the ECM in producing the mixed signal,

at medium background noise levels, attenuating low frequencies in the electronic ambient signal and attenuating high frequencies in the electronic internal signal, and

at high background noise levels, amplifying the electronic internal signal from the ECM relative to the electronic ambient signal from the ASM in producing the mixed signal.

9. The method of claim 1, where the mixing is performed in accordance with the relation:

mixed signal=(1−β) *electronic ambient signal+(β)*electronic internal signal, where (1−β) is an external gain, (β) is an internal gain, and the mixing is performed with $0<\beta<1$.

10. The method of claim 1, further comprising

estimating a voice activity level from the electronic internal signal or the electronic ambient signal; and

scaling the electronic internal signal and the electronic ambient signal in accordance with the voice activity level.

11. The method of claim 10, wherein the mixing is performed by

applying a first gain (G1) to the electronic ambient signal, and

applying a second gain (G2) to the electronic internal signal, where the first gain and the second gain are a function of a background noise level (BNL) and the voice activity level (VAL), according to the relation:

$$G1=f(BNL)+f(VAL) \text{ and } G2=f(BNL)+f(VAL).$$

12. The method of claim 10, where the step of measuring the background noise signal includes

accounting for an acoustic attenuation level of the earpiece, and

accounting for an audio content level reproduced by an Ear Canal Receiver (ECR) that delivers acoustic audio content to the earpiece.

13. A method for acoustic management control suitable for use in an earpiece, the method comprising the steps of:

capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal;

capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal;

detecting a spoken voice signal generated by a wearer of the earpiece from the electronic ambient signal or the electronic internal signal;

measuring a background noise level from the electronic ambient signal or the electronic internal signal when the spoken voice signal is not detected; and

mixing the electronic ambient signal with the electronic internal signal as a function of the background noise level to produce a mixed signal.

14. The method of claim 13, comprising

delivering audio content to the ear canal by way of an Ear Canal Receiver (ECR); and

adjusting the mixing based on a level of the audio content, the background noise level, and an acoustic attenuation level of the earpiece.

15. The method of claim 14, wherein the audio content is at least one among a phone call, a voice message, a music signal, and the spoken voice signal.

16. The method of claim 13, comprising

suppressing in the mixed signal an echo of the spoken voice signal generated by the wearer of the earpiece, and

producing a modified electronic internal signal containing primarily the spoken voice signal.

17. The method of claim 16, wherein the suppressing is performed by way of a normalized least mean squares algorithm.

18. The method of claim 13, comprising

generating a voice activity level of the spoken voice signal, and

mixing the electronic ambient signal with the electronic internal signal as a function of the voice activity level and the background noise level.

19. An earpiece for acoustic management control, comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound and produce an electronic ambient signal;

an Ear Canal Receiver (ECR) to deliver audio content to an ear canal to produce an acoustic audio content;

an Ear Canal Microphone (ECM) configured to capture internal sound in the ear canal and produce an electronic internal signal; and

a processor operatively coupled to the ASM, the ECM and the ECR where the processor is configured to

measure a background noise signal from the electronic ambient signal or the electronic internal signal; and

mix the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal.

20. The earpiece of claim 19, wherein the processor filters the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal using filter coefficients stored in a memory or generated algorithmically.

21. The earpiece of claim 20, further comprising a transceiver operatively coupled to the processor to transmit the mixed signal to a communication device, where the processor also plays the mixed signal back to the ECR for loopback listening.

22. The earpiece of claim 20, further comprising an echo suppressor operatively coupled to the processor to suppress an echo of a spoken voice generated by a wearer of the earpiece when speaking.

23. The earpiece of claim 22, further comprising a voice activity detector operatively coupled to the echo suppressor to detect the spoken voice generated by the wearer in the presence of the background noise signal.

24. The earpiece of claim 22, where the processor generates a voice activity level for the spoken voice and applies gains to the electronic ambient signal and the electronic internal signal as a function of a background noise level and the voice activity level.

25. The earpiece of claim 23, further comprising a control unit operatively coupled to the voice activity detector to freeze weights of a Least Mean Squares (LMS) system in the echo suppressor during the speaking of the spoken voice.

* * * * *

Exhibit B

US008805692B2

(12) **United States Patent**
Goldstein

(10) **Patent No.: US 8,805,692 B2**
(45) **Date of Patent: Aug. 12, 2014**

(54) **PERSONAL AUDIO ASSISTANT DEVICE AND METHOD**

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(72) Inventor: **Steven Wayne Goldstein**, Delray Beach, FL (US)

(73) Assignee: **Personics Holdings, LLC**, Boca Raton, 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.

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(22) Filed: **Jan. 7, 2014**

(65) **Prior Publication Data**

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(60) Provisional application No. 60/806,769, filed on Jul. 8, 2006.

(51) **Int. Cl.**
G10L 15/00 (2013.01)

(52) **U.S. Cl.**
USPC **704/275**; 704/270; 455/575.2

(58) **Field of Classification Search**
USPC 704/270, 275; 455/575.2
See application file for complete search history.

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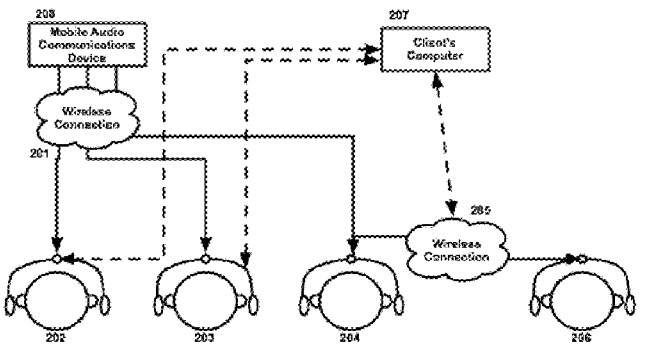
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(74) *Attorney, Agent, or Firm* — Pablo Meles

(57) **ABSTRACT**

A wearable device includes a housing for the wearable device, a first microphone within or on the housing, a communication module within or on the housing, a logic circuit communicatively coupled to the first microphone, a memory storage unit communicatively coupled to the logic circuit and an interaction element. The interaction element and logic circuit cooperatively initiate control of media content or initiate operations of the wearable device. Other embodiments are disclosed.

20 Claims, 5 Drawing Sheets



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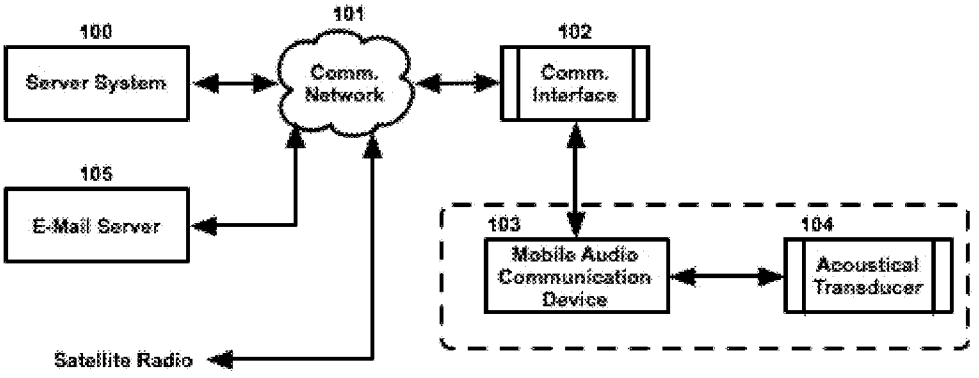

100
Server System
101
Comm. Network
102
Comm. Interface
105
E-Mail Server
103
Mobile Audio Communication Device
104
Acoustical Transducer
Satellite Radio

Fig. 1

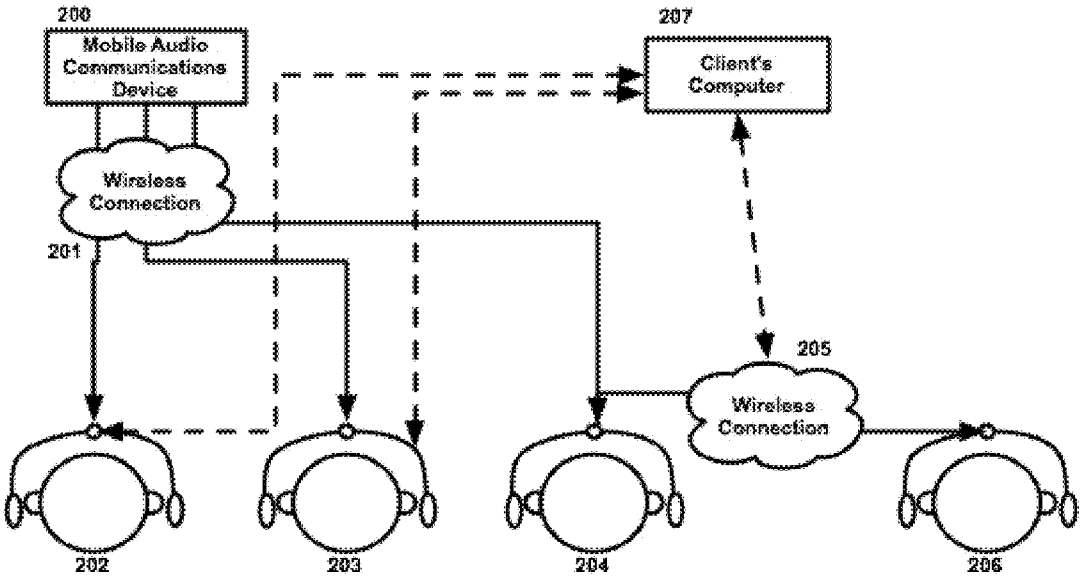

200
Mobile Audio Communications Device
207
Client's Computer
Wireless Connection
201
205
Wireless Connection
202
203
204
206

Fig. 2

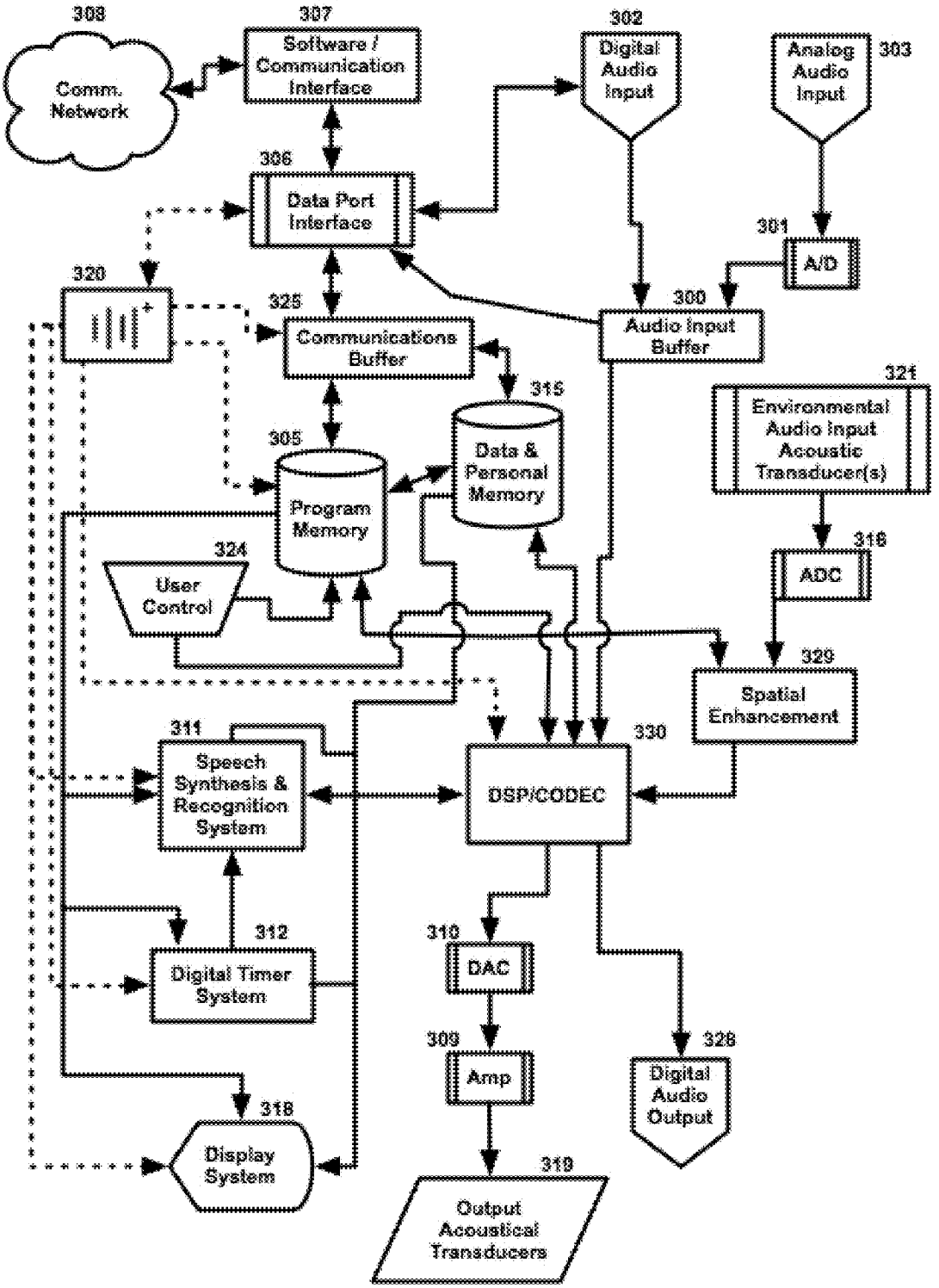

308
Comm. Network
307
Software / Communication Interface
302
Digital Audio Input
303
Analog Audio Input
306
Data Port Interface
301
A/D
320
325
Communications Buffer
300
Audio Input Buffer
315
321
Environmental Audio Input Acoustic Transducer(s)
305
Program Memory
Data & Personal Memory
316
ADC
324
User Control
329
Spatial Enhancement
311
Speech Synthesis & Recognition System
330
DSP/CODEC
312
Digital Timer System
310
DAC
309
Amp
328
Digital Audio Output
318
Display System
319
Output Acoustical Transducers

Fig. 3

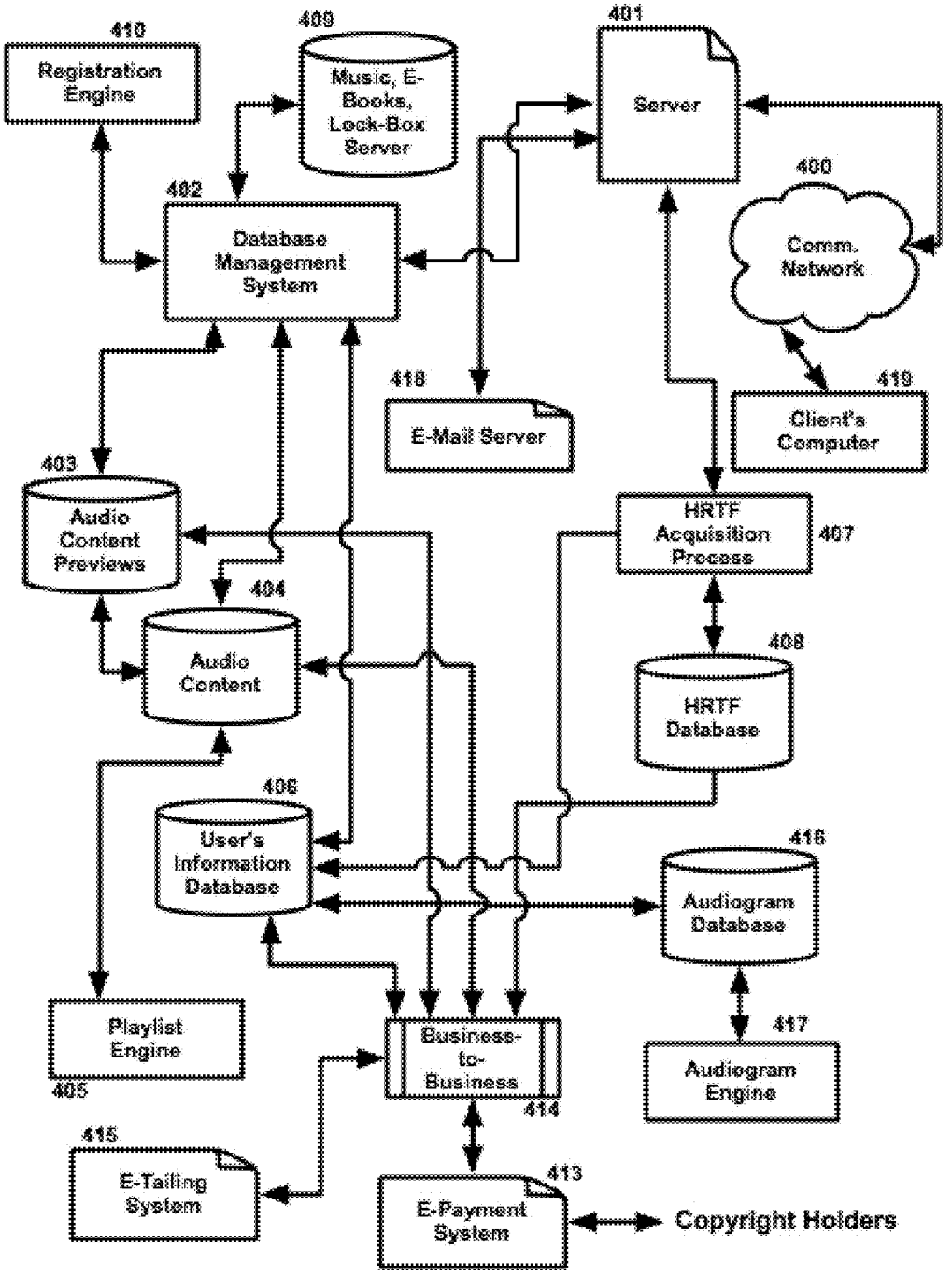

410
Registration Engine
409
Music, E-Books, Lock-Box Server
401
Server
402
Database Management System
400
Comm. Network
418
E-Mail Server
419
Client's Computer
403
Audio Content Previews
HRTF Acquisition Process
407
404
Audio Content
408
HRTF Database
406
User's Information Database
416
Audiogram Database
Playlist Engine
405
Business-to-Business
414
417
Audiogram Engine
415
E-Tailing System
413
E-Payment System
Copyright Holders

Fig. 4

FIG. 5A

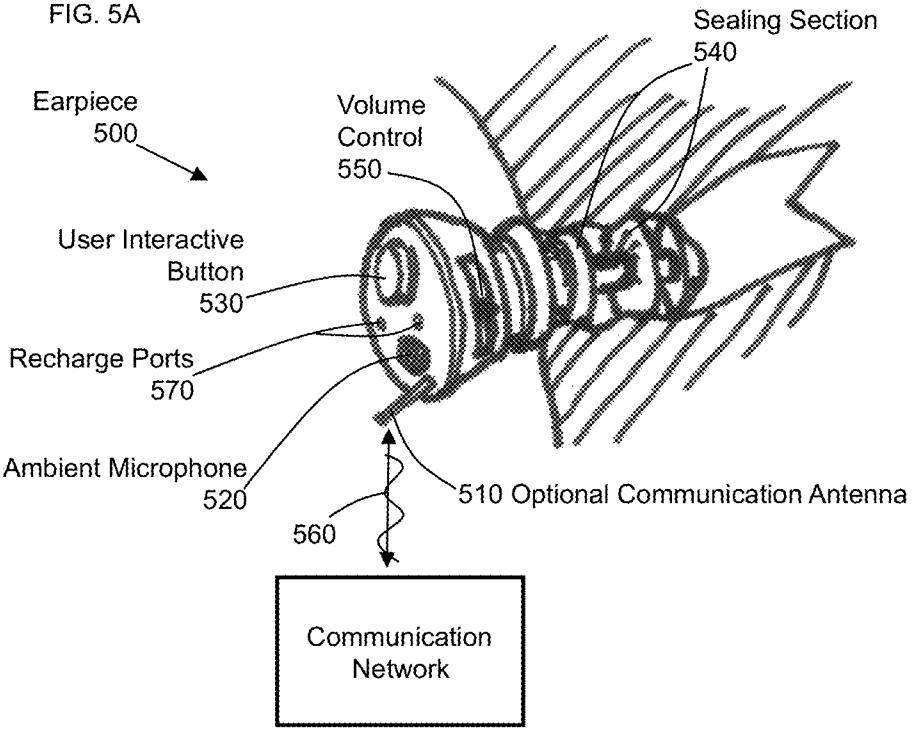

Earpiece
500
Volume
Control
550
Sealing Section
540
User Interactive
Button
530
Recharge Ports
570
Ambient Microphone
520
560
510 Optional Communication Antenna
Communication
Network

FIG. 5B

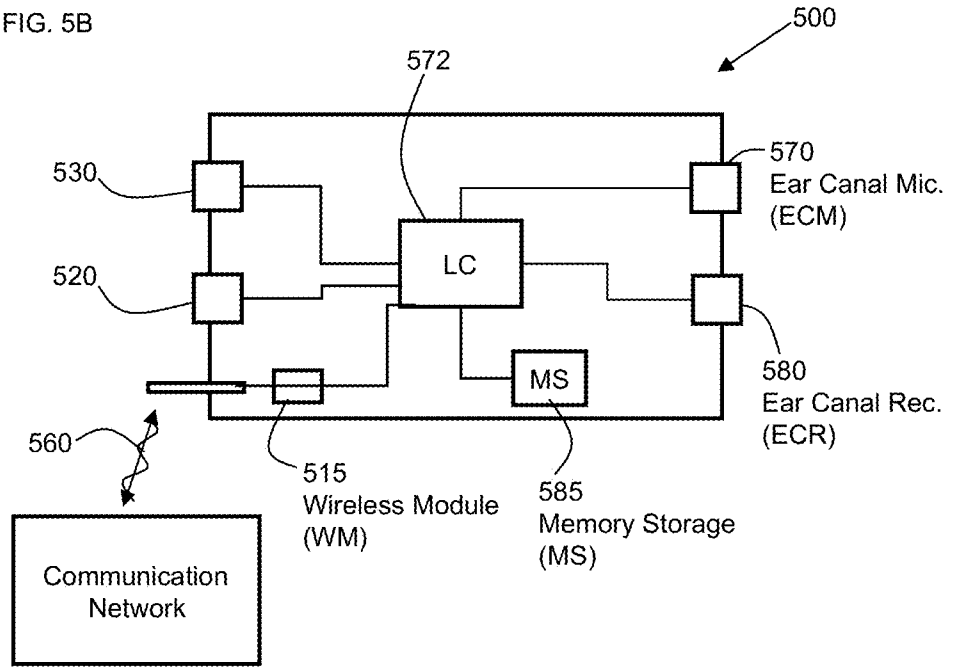
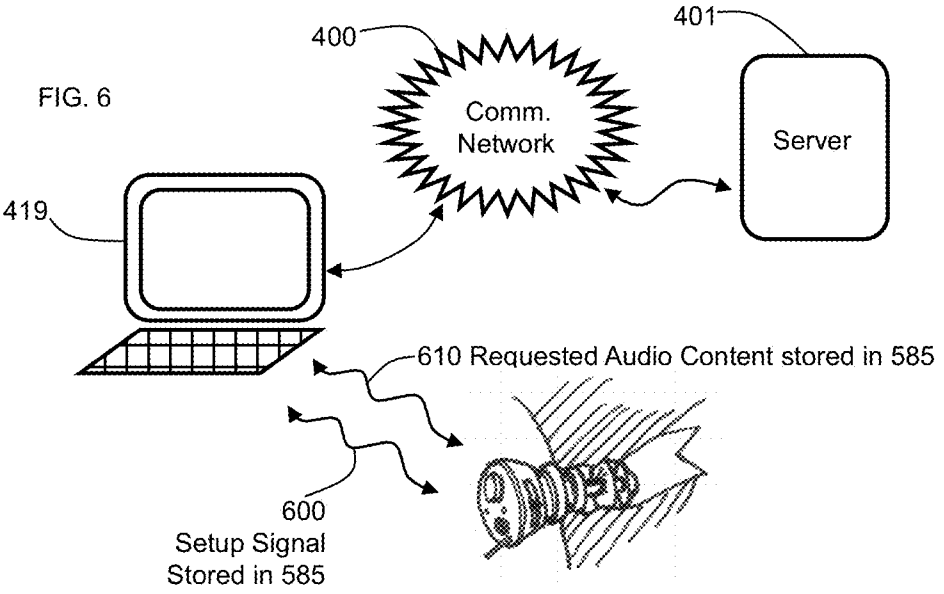

500
572
530
570
Ear Canal Mic.
(ECM)
520
LC
MS
580
Ear Canal Rec.
(ECR)
560
515
Wireless Module
(WM)
585
Memory Storage
(MS)
Communication
Network

FIG. 6


400
Comm. Network
401
Server
419
610 Requested Audio Content stored in 585
600
Setup Signal
Stored in 585

PERSONAL AUDIO ASSISTANT DEVICE AND METHOD

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the priority benefit of U.S. Provisional Application No. 60/806,769, under 35 U.S.C. §119(e), filed 8 Jul. 2006, and U.S. Non-Provisional application Ser. No. 11/774,965 filed 9 Jul. 2007, both of which are incorporated herein by reference in their entireties.

FIELD

The claimed embodiments relate in general to methods and devices for the storage and recall of audio content via an earpiece, and in particular, though not exclusively, for the storage and playing of music or verbal content on a system that is built into a headphone.

BACKGROUND

Present audio content playing devices are separated from the headphone system that normally contains the speakers (also referred to as receivers). The reason for this has typically been that audio content has been stored on disks that require a separate playing system. However, even with the advent of storing audio content on non-disk RAM (Random Access Memory) storage systems, the audio content player has been separated from the earpiece system (e.g., plug in headphones or earbuds). Combining the capacity for audio download and playing in an earpiece system is not obvious over related art since the user interaction system (e.g., play button, keyboard system) does not readily appear compatible with the size of an earpiece device and the difficulty of user interaction.

Additionally, no system currently exists for registration and download of audio content into an earpiece.

BRIEF DESCRIPTION OF THE DRAWINGS

Examples of the claimed embodiments will become apparent from the following detailed description, taken in conjunction with the drawings in which:

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network;

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity);

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment;

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer;

FIG. **5**A illustrates an earpiece that can store and download audio content in accordance with at least one exemplary embodiment;

FIG. **5**B illustrates a block diagram of the earpiece of FIG. **5**A; and

FIG. **6** illustrates a user interface for setting the parameters of the Personal Audio Assistant.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS

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

Processes, methods, materials and devices known by one of ordinary skill in the relevant arts can not be discussed in detail but are intended to be part of the enabling discussion where appropriate for example the generation and use of transfer functions.

Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it can not be discussed for following figures.

Note that herein when referring to correcting or corrections of an error (e.g., noise), a reduction of the error and/or a correction of the error is intended.

SUMMARY OF EXEMPLARY EMBODIMENTS

At least one exemplary embodiment is directed to a system for Personalized Services delivered to a Personal Audio Assistant incorporated within an earpiece (e.g., earbuds, headphones). Personalized Services include content such as music files (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech, speech-to-text processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction, academic instruction, time and date information, speech-to-speech delivery, instructions from a GPS system and others. A Personal Audio Assistant can be an audio playback platform for providing the user with Personalized Services.

At least one exemplary embodiment is directed to a Personal Audio Assistant system that is included as part of an earpiece (e.g., Headphone system). The Personal Audio Assistant is capable of digital audio playback, mitigating the need to carry a personal music player. Furthermore, a subscription-based service provides audio content to the user through the Personal Audio Assistant. The type of audio content, which is automatically provided to the user, is based on the user's preferences, which are obtained through a registration process.

The audio content, which is seamlessly downloaded to the Personal Audio Assistant in the background, is managed from a Server system and is only available on the Personal Audio Assistant for a predetermined period of time or for a fixed number of playback counts. However, the user can purchase any music file or electronic book directly from the Personal Audio Assistant with a simple one-click control interface, storing the purchased audio content on the Personal Audio Assistant as well as storing the content permanently in a user storage lock-box location on the Server system.

The system provides for audio content to be new and "fresh" each time the user auditions the content. As such, the content is typically auditioned in a first-in: first-out scenario. In one such example, the user has turned on the Personal Audio Assistant at 8:00 am and by 10:00 am has auditioned 2 hours of content that were created for the user as a manifestation of the user's choices of their preferences of genre, artist, their demographics, day of the week, time of day and purchase history. The system also provides for the elimination of a particular song or playlist in situ.

As the user's Listening History Envelope is updated based on experience, subsequent downloads will only contain content incorporating these revised preferences. The Personal Audio Assistant provides for ample memory, thus permitting hours of uninterrupted playback without the need to download additional content from the server. When in need, the Personal Audio Assistant automatically interrogates various communication platforms as it searches for connections.

Once a connection is made, the Listener History Envelope file is uploaded to the server, and a new set of personalized playlist content is downloaded to the Personal Audio Assistant. Accordingly, as the Personal Audio Assistant content is auditioned and thus depleted, the communications system provides for constant replenishment.

In another embodiment, the Personal Audio Assistant also provides for a new set of business solutions to be offered to the music industry. As the personalized audio content is only available for audition for a limited period of time, and can not be sent to the user again from for weeks to months, the user's purchasing behavior can be demonstrated as spontaneous. The basic model of "Try before you buy" is the expected outcome. In another iteration, the distributor of the music can choose to offer discounts, which can be time-sensitive or quantity-sensitive in nature, in effect promoting greater purchase activity from the user.

In another iteration, while in audition a user can wish to place the desired content in a hold status. The hold status forms the basis of a "wish list," thus allowing the user to hold for future consideration audio content while it is being auditioned. This content resides in the memory of the Personal Audio Assistant for a defined period, and is automatically erased, or the user can do so manually. The selected content will also appear on the user's computer via a URL address; here it resides on the server ready for audition or purchase and download.

The system is designed to operate as simply as possible. Using a single button, which has multiple contacts, the interface allows the user to purchase, delete, skip to next, and add to a wish list and even control a listening level.

In another iteration, the user can download their own music to the Personal Audio Assistant for audition. The Personal Audio Assistant system is capable of text-to-speech processing and can interface with personal scheduling software to provide auditory schedule reminders for the user. Auditory reminders relating to the user's medication schedule are also generated by the system.

At least one exemplary embodiment includes input Acoustic Transducers (microphones) for capturing user's speech as well as Environmental Audio. In further embodiments, stereo input Acoustic Transducers capture Environmental Audio, and, mixing it with the audio signal path, present the ambient sound field to the user, mitigating the need to remove the Headphone apparatus for normal conversation.

Additional exemplary embodiments are directed to various scenarios for the delivery and consumption of audio content. The Personal Audio Assistant can store and play back audio content in compressed digital audio formats. In one embodiment, the storage memory of the Personal Audio Assistant is completely closed to the end-user and controlled from the Server. This allows for audio content to be distributed on a temporary basis, as part of a subscription service. In another iteration of the present embodiments, the storage memory of the Personal Audio Assistant is not completely closed to the end-user, allowing the user to transfer audio content to the Personal Audio Assistant from any capable device such as a Personal Computer or a Personal Music Player.

In at least one exemplary embodiment the Personal Audio Assistant automatically scans for other Bluetooth®-enabled audio playback systems and notifies the user that additional devices are available. These additional devices can include a Bluetooth® video system, television system, personal video player, video camera, cell phone, another Personal Audio Assistant and others.

In another iteration, the Personal Audio Assistant can be directly connected to a Terrestrial Radio receiver, or have such a receiver built in to the system.

In another exemplary embodiment, a technique known as Sonification can be used to convey statistical or other numerical information to a headphone. For example, the user would be able to receive information about the growth or decline of a particular stock, groups of stocks or even sectors of the markets though the Personal Audio Assistant. Many different components can be altered to change the user's perception of the sound, and in turn, their perception of the underlying information being portrayed. An increase or decrease in some level of share price or trading levels can be presented to the user. A stock market price can be portrayed by an increase in the frequency of a sine tone as the stock price rose, and a decline in frequency as it fell. To allow the user to determine that more than one stock was being portrayed, different timbres and spatial locations might be used for the different stocks, or they can be played to the user from different points in space, for example, through different sides of their headphones. The user can act upon this auditory information and use the controls built-in to the headphone to either purchase or sell a particular stock position.

Furthermore, specific sonification techniques and preferences can be presented to the user as "themes" from which the user can select. For example, one theme might auralize the current trading price of one stock with an ambient sine tone in the left ear, the price of another stock in the right ear, their respective trade volumes as perceived elevation using personalized head-related transfer function binauralization, and the current global index or other market indicator as the combined perceptual loudness of both tones. Such a scheme affords ambient auditory display in this example of five dimensions of financial data without compromising the user's ability to converse or work on other tasks. In another embodiment, the system affords users the ability to customize themes to their liking and to rapidly switch among them using simple speech commands. Additionally, the user can search the web from voice commands and receive results via a text to speech synthesizer.

In yet another exemplary embodiment the Personal Audio Assistant (PAA) functions as a dictation device for medical professionals for dictating clinical information to a patient's medical record, or write prescriptions for medication or devices. Conversely, the PAA can function as text-to-speech allowing the clinician to audition information from a medical record, rather than reading. The PAA can save considerable time preparing clinician interaction with a patient.

In another iteration, the Personal Audio Assistant can function as a tool to locate other users of Personal Audio Assistant who share common interests, or who are searching for particular attributes of other users. Whereas the first user has stored specific personal information in the Public Data memory of the Personal Audio Assistant, an example of which might be related to schools attended, marital status, profession etc, or the first user can be in search of another user with these attributes and whereas a second user of a Personal Audio Assistant comes within communication range of the first user, the individual Personal Audio Assistants communicate with each other, access the personal information stored in each of their respective Public Data memories to ascertain if these users have common interests. If a match occurs, each unit can contain both audible and visual indicators announcing that a match has been made and thus each user can start dialog either physically or electronically via the environmental microphones.

EXAMPLES OF TERMINOLOGY

Note that the following non-limiting examples of terminology are solely intended to aid in understanding various exemplary embodiments and are not intended to be restrictive of the meaning of terms nor all inclusive.

Acoustic Isolation Cushion: An "Acoustic Isolation Cushion" shall be defined as a circum-aural or intra-aural device that provides acoustic isolation from Environmental Noise. Acoustic Isolation Cushions can be included as part of a Headphones system, allowing the output of the acoustical transducers to reach the ear unimpeded, but still providing acoustic isolation from Environmental Noise.

Acoustic Transducer: An "Acoustic Transducer" shall be defined as a device that converts sound pressure level variations into electronic voltages or vice versa. Acoustic Transducers include microphones, loudspeakers, Headphones, and other devices.

Audio Playback: "Audio Playback" shall be defined as the auditory stimuli generated when Playback Hardware reproduces audio content (music, speech, etc) for a listener or a group of listeners listening to Headphones.

Audition: "Audition" shall be defined as the process of detecting sound stimulus using the human auditory system. This includes the physical, psychophysical, psychoacoustic, and cognitive processes associated with the perception of acoustic stimuli.

Client: A "Client" shall be defined as a system that communicates with a Server, usually over a communications network, and directly interfaces with a user. Examples of Client systems include personal computers and mobile phones.

Communications Port: A Communication Port shall be defined as an interface port supporting bidirectional transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others) via a communications network (e.g., the Internet, cellular networks).

Control Data: "Control Data" shall be defined as information that dictates the operating parameters for a system or a set of systems.

Earcon: An Earcon shall be defined as a Personalized Audio signal that informs the User of a pending event typically inserted in advance of the upcoming audio content.

Ear Mold Style: "Ear Mold Style" shall be defined as a description of the form factor for an intra-aural device (e.g., hearing aids). Ear Mold Styles include completely in the canal (CIC), in the canal (ITC), in the ear (ITE), and behind the ear (BTE).

Environmental Audio: "Environmental Audio" shall be defined as auditory stimuli of interest to the user in the environment where the user is present. Environmental Audio includes speech and music in the environment.

Environmental Noise: "Environmental Noise" shall be defined as the auditory stimuli inherent to a particular environment where the user is present and which the user does not wish to audition. The drone of highway traffic is a common example of Environmental Noise. Note that Environmental Noise and Audio Playback are two distinct types of auditory stimuli. Environmental Noise does not typically include Music or other audio content.

E-Tailing System: An "E-tailing System" shall be defined as a web-based solution through which a user can search, preview and acquire some available product or service. Short for "electronic retailing," E-tailing is the offering of retail goods or services on the Internet. Used in Internet discussions as early as 1995, the term E-tailing seems an almost inevitable addition to e-mail, e-business, and e-commerce. E-tailing is synonymous with business-to-consumer (B2C) transactions. Accordingly, the user can be required to register by submitting personal information, and the user can be required to provide payment in the form of Currency or other consideration in exchange for the product or service. Optionally, a sponsor can bear the cost of compensating the E-tailer, while the user would receive the product or service.

Generic HRTF: A "Generic HRTF" shall be defined as a set of HRTF data that is intended for use by any Member. A Generic HRTF can provide a generalized model of the parts of the human anatomy relevant to audition and localization, or simply a model of the anatomy of an individual other than the Member. The application of Generic HRTF data to Audio Content provides the least convincing Spatial Image for the Member, relative to Semi-Personalized and Personalized HRTF data. Generic HRTF data is generally retrieved from publicly available databases such as the CIPIC HRTF database.

Headphones: "Headphones" (also known as earphones, earbuds, stereophones, headsets, Canalphones, or the slang term "cans") are a pair of transducers that receive an electrical signal from a media player, communication receivers and transceivers, and use speakers placed in close proximity to the ears (hence the name earphone) to convert the signal into audible sound waves. Headphones are intended as personal listening devices that are placed either circum-aural or intra-aural according to one of the Ear Mold Styles, as well as other devices that meet the above definition such as advanced eyewear that includes Acoustical Transducers (i.e. Dataview). Headphones can also include stereo input Acoustic Transducers (microphones) included as part of the Ear Mold Style form factor.

HRTF: "HRTF" is an acronym for head-related transfer function—a set of data that describes the acoustical reflection characteristics of an individual's anatomy relevant to audition. Although in practice they are distinct (but directly related), this definition of HRTF encompasses the head-related impulse response (HRIR) or any other set of data that describes some aspects of an individual's anatomy relevant to audition.

Informed Consent: "Informed Consent" shall be defined as a legal condition whereby a person can provide formal consent based upon an appreciation and understanding of the facts and implications associated with a specific action. For minors or individuals without complete possession of their faculties, Informed Consent includes the formal consent of a parent or guardian.

Listening History Envelope: "Listening History Envelope" shall be defined as a record of a user's listening habits over time. The envelope includes system data, time system was turned off, time the system is presenting content, when the system doesn't audition, system transducers, when the user auditions content, time stamp of content being auditioned, content which is: skipped, deleted, played multiple times, saved in the Wish List, and time between listening sessions.

Music: "Music" shall be defined as a form of expression in the medium of time using the structures of tones and silence to create complex forms in time through construction of patterns and combinations of natural stimuli, principally sound. Music can also be referred to as audio media or audio content.

Playback Hardware: Any device used to play previously recorded or live streaming audio. Playback Hardware includes Headphones, loudspeakers, personal music players, mobile phones, and other devices.

Personal Audio Assistant: A "Personal Audio Assistant" shall be defined as a portable system capable of interfacing with a communications network, directly or through an intermediate, to transmit and receive audio signals and other data.

Personal Computer: "Personal Computer" shall be defined as any piece of hardware that is an open system capable of compiling, linking, and executing a programming language (such as C/C++, Java™, etc.).

Personal Music Player: "Personal Music Player" shall be defined as any portable device that implements perceptual audio decoder technology but is a closed system in that users are not generally allowed or able to write software for the device.

Personalized HRTF: A "Personalized HRTF" shall be defined as a set of HRTF data that is measured for a specific Member and unique to that Member. The application of Personalized HRTF data to Audio Content creates, by far, the most convincing Spatial Image for the Member (Begault et. al. 2001, D. Zotkin, R. Duraiswami, and L. Davis 2002).

Personalized Services: "Personalized Services" shall be defined as services customized to better meet the needs of an individual. Personalized Services include media content (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction [real-time foreign language translation], academic instruction, time and date information, and others.

Public Data: "Public Data" shall be defined as data which contains specific and personal information about the registered user of the Personal Audio Assistant. The registered user chooses which portions of their complete Registration Process data they wish to include in this subset. This data becomes distributed to other users who have compliant devices thus allowing other users to know specific details of the registered user.

Registration Process: "Registration Process" includes the acquisition of the user's preference via a web page. Typically, the process would include the items to be captured: Age, demographics, email, gender, Relative Audiogram, Personal Preferences, banking information, credit card information, wake-up and sleep times, music preferences by genre, artist, preferences for writers and authors, desire to receive advertising, turn-on listening level, equalization, email preferences, parental control setup as well as other user-controlled settings.

Relative Audiogram: A "Relative Audiogram" shall be defined as a measured set of data describing a specific individual's hearing threshold level as a function of frequency. A Relative Audiogram is only an approximate Audiogram, leaving more complete Audiogram analysis to qualified audiologists.

Semi-Personalized HRTF: A "Semi-Personalized HRTF" shall be defined as a set of HRTF data that is selected from a database of known HRTF data as the "best-fit" for a specific user. Semi-Personalized HRTF data is not necessarily unique to one user; however, interpolation and matching algorithms can be employed to modify HRTF data from the database to improve the accuracy of a Semi-Personalized HRTF. The application of Semi-Personalized HRTF data to Audio Content provides a Spatial Image that is improved compared to that of Generic HRTF data, but less effective than that of Personalized HRTF data. The embodiments within speak to a variety of methods for determining the best-fit HRTF data for a particular Member including anthropometrical measurements extracted from photographs and deduction.

Server: A "Server" shall be defined as a system that controls centrally held data and communicates with Clients.

Sonification: "Sonification" shall be defined as the use of non-speech audio to convey information or to aurally perceptualize non-acoustic data (auralize). Due to a variety of phenomena involving human cognition, certain types of information can be better or more efficiently conveyed using auditory means than, for example, visual means.

Exemplary Embodiments

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network (**101**) via communication interface **102**, which can be operatively connected (via a wired or wireless connection) to a server system (**100**) and/or an e-mail server (**105**). Additionally a radio signal (e.g., satellite radio) can be input into the earpiece **500** (FIG. **5**B) via a communication module (e.g., Bluetooth® wireless module **515**).

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity). For example multiple users (e.g., **202**, **203**, **204**, and **206**) can send signals to each individual earpiece (e.g., **500**) when in range (e.g., via a wireless connection **205**) or to a mobile audio communications device **200** via a wireless connection (**201**) with each earpiece (**500**). Additionally information (e.g., audio content, software download) can be sent via a client's computer **207** to each earpiece, either directly (e.g., **205**), or via **200**. For example audio content can be retrieved on a user's computer and sent to the earpieces that have authorization to use it.

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment. The earpiece can include all or some of the elements illustrated in FIG. **5**B. For example the logic circuit **572** or the operatively connected memory storage device **585**, can include spatial enhancement software **329**, a DSP codec **330**, a speech synthesis and recognition system **311**, and a digital timer system **312**. Additional elements can be connected to the logic circuit **572** as needed, for example a power supply **320**, a software communication interface **307** (e.g., wireless module **515**) (which may be connected to communication network **308**), data port interface **306**, audio input buffers **300** connected to digital audio input **302** and/or analog audio input **303** converted to digital via an ADC **301**, environmental audio input acoustic transducer(s) **321** converted to digital via an ADC **316**, user control **324**, digital audio output **328**, output acoustic transducers **319** (which receive signals converted to analog via a DAC **310** and amplified via amplifier **309**), display system **318**, communication buffers **325**, program memory **305**, data and personal memory **315**, as well as other electronic devices as known by one of ordinary skill in the relevant arts.

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer **419**, via a communication network **400** (e.g., Internet connection) connected to many various database and registration systems as illustrated and labeled in FIG. **4**. For example, server **401**, database management system **402**, audio content preview database **403**, audio content database **404**, playlist engine **405**, user's information database **406**, HRTF acquisition process module **407**, HRTF database **408**, lock-box server **409**, registration engine **410**, e-payment system **413**, business-to-business module **414**, e-tailing system **415**, audiogram database **416**, audiogram engine **417** and/or e-mail server **418**.

FIG. **5**A illustrates an earpiece **500** having sealing section **540** in an ear canal that can store and download audio content **560** in accordance with at least one exemplary embodiment. The earpiece **500**, can include a first user interaction element

530 (e.g., a button), that can be used to turn the earpiece 500 on, or if on then activate an audio play command to start playing saved audio content. The earpiece 500 can also include a second user interaction element 550 (e.g., a slide control) that can be used for example to control the volume. The earpiece 500 can also include recharge ports 570, that can accept two wires of varying voltage that can be inserted into the recharge ports 570 to recharge any batteries in the earpiece 500. The earpiece 500 can include an ambient microphone 520 and an optional communication antenna 510, that if needed can aid in the communication between the earpiece 500 and a communication network.

FIG. 5B illustrates a block diagram of the earpiece of FIG. 5A, illustrating the first user interaction element 530, the ambient microphone (AM) 520, that can be used to pick up ambient audio content, an ear canal microphone (ECM) 590 that can pick up audio in the ear canal region, an ear canal receiver (ECR) 580 that can direct audio content to the ear drum, all of which can be connected operatively to a logic circuit 572. A memory storage device 585 can be operatively connected to the logic circuit (LC) 572, and can store data such as registration, preference, and audio content data. The optional communication antenna 510 can be connected to a communication module (e.g., wireless module 515), and can receive or transmit information 560 to a communication network.

FIG. 6 illustrates a user interface for setting the parameters stored in the memory storage device 585. For example a user can use his/her computer 419 to communicate with a server 401 (e.g., via a communication network 400) to start the user's registration (e.g., with an audio content provider). The registration information can then be transmitted 600 to set the stored parameters in the memory storage device 585 of the earpiece 500. Additionally a requested (e.g., bought) audio content can be downloaded 610 into the memory storage device 585 of the earpiece 500.

At least one exemplary embodiment is directed to an earpiece comprising: an ambient microphone; an ear canal microphone; an ear canal receiver; a sealing section; a logic circuit; a communication module; a memory storage unit; and a user interaction element, where the user interaction element is configured to send a play command to the logic circuit when activated by a user where the logic circuit reads registration parameters stored on the memory storage unit and sends audio content to the ear canal receiver according to the registration parameters.

In at least one exemplary embodiment the audio content is stored in the memory storage unit. The earpiece according to claim 2, where the communications module is a wireless communications module. Additionally the earpiece can include a second user interaction element configured to alter the volume of the audio content that is emitted from the ear canal receiver.

Upon a play command being received by the logic circuit the logic circuit can check registration parameters stored in the memory storage device for example one of the registration parameters can be whether the audio content is a sample audio content or a fully purchased audio content, or the allowed number of times an audio content can be played, and a counter value that keeps track of the number of times the audio content has been played.

The earpiece can send an auditory warning to be emitted by the ear canal receiver when the counter value is greater than or equal to the allowed number of times the audio content can be played, and where the logic circuit does not send the audio content to the ear canal receiver.

Further Exemplary Embodiments

At least one exemplary embodiment is directed to a system for the delivery of Personalized Services to Personal Audio Assistants, the system comprising: a Personal Audio Assistant system for presenting Personalized Services to the user as Audio Playback; a Server system for user registration, Personalized Service management, and communication; a Registration Process for collecting detailed registration information from users, including the information necessary for creating Personalized Services; a communications protocol (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to the Server or connecting the Personal Audio Assistant to other Personal Audio Assistants (peer-to-peer behavior).

In at least one exemplary embodiment a Personal Computer acts as an intermediate, connecting to the Server system over a communications network and connecting to the Personal Audio Assistant over a local connection. At least one exemplary embodiment includes a Personal Hearing Damage Intervention System (e.g., USPTO—60/805,985—Goldstein).

In at least one exemplary embodiment a Personal Audio Assistant system included as part of a Headphone system, the system comprising: a Communications Port supporting a communications protocol enabling communication with the Server system, peer devices, and other capable devices; a non-volatile program memory storage system for storing Control Data, dictating system behavior; a data memory storage system for storing data and audio content; an analog audio input/output and corresponding ADC/DAC; a digital audio input/output and a digital audio signal path; a user control system allowing the user to adjust the level of the audio output and control the behavior of the system; a user control system allowing the user to purchase the content being auditioned in real time; a user control system allowing the user to control, delete, fast forward, output level control, scan, advance, the data stored both stored in memory as well as new streaming data emails and reminders; a display system for presenting information to the user(s) visually using any method familiar to those skilled in the art (LED, LCD, or other); a display system for presenting information to the user(s) (e.g., using Earcons and other sound files); a speech synthesis system for converting text-to-speech and generating speech signals; a speech recognition system for converting speech to-text to respond and send emails and to interface with the control language as to provide navigational commands; a digital timer system; a power supply system in the form of a battery; a unique identification number for each Personal Audio Assistant; Input Acoustic Transducers; an Output Acoustic Transducer; an Audio amplification system; Acoustic Isolation Cushions conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE; see definitions) and other elements common to Headphone systems; a digital signal processor (DSP) system; and a CODEC processor capable of improving the perceptual sound quality of the content to be auditioned while governed by delivering the correct SPL dose.

In at least one exemplary embodiment the system is independent of a Headphone array or can be included and imbedded as part of a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio sys-

tem, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the various processing needed to derive the intended functions are distributed among any combination of a Server system, a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the Personal Audio Assistant system can exchange audio signals with a mobile phone via the Communications Port, allowing the Personal Audio Assistant to function as a mobile phone accessory.

In at least one exemplary embodiment a communications buffer is included. For example when a network connection is available, the communications buffer uploads stored content (e.g., Listening Habits Envelope) and stores incoming transmissions (e.g., music, electronic books, and updates to the firmware or operating system) from the Communications Port; The contents of the communications buffer are then transmitted whenever a network connection becomes available. At least one exemplary embodiment includes a perceptual audio codec decoding technology in the DSP, enabling the storage and playback of compressed digital audio formats (e.g., MP3, AAC, FLAG, etc.). At least one exemplary embodiment is compliant and compatible with DRM, FairPlay® and other forms of digital content governance.

At least one exemplary embodiment includes a user control system for selecting and playing back audio content stored in memory that operates using any combination of the following methods: a button or tactile interface which upon auditioning a song can be pressed to order content; a button, tactile and/or voice controlled interface which, when pressed once, commanded to, activates playback of short audio clips or audio thumbnails of the audio content stored in memory; When the button is pressed again during audio thumbnail playback, the current audio content selection is played in its entirety; The behavior of this interface is similar to the "scan" button interface common in FM/AM radio devices; a button, tactile and/or voice controlled interface that, when pressed or commanded to, skips to the next piece of audio content, which is selected randomly from all available audio content that has a play count equal to or less than the play count of the piece of audio content currently playing; The behavior of this interface is similar to the "shuffle" behavior found in some personal music players; an interface for browsing audio content storage devices familiar to those skilled in the art; and a process to allow for increased data memory storage capacity for storing audio content.

In at least one exemplary embodiment the contents of the data memory are encrypted and controlled by the Server system only, prohibiting the end-user from loading unauthorized audio content into the data memory. Further the contents of the data memory can be manipulated by the end-user, allowing the user to transfer audio content to the Personal Audio Assistant system from any device capable of interfacing with the communications port; For example, audio content can be transferred to the system from a Personal Music Player or a Personal Computer. According to at least one exemplary embodiment, audio content (or other media content) updates are retrieved from the Server system any time a connection is detected by the communications port. Furthermore, an exemplary embodiment can include an acoustical and/or visual indicator informing the user when a transfer of data is activated.

In at least one exemplary embodiment radio wave transmissions are used to implement some communications protocol and the communications port acts as a radio receiver. Additionally the Personal Audio Assistant can include: interfaces with some personal scheduling software through the communications port; a speech synthesis system which generates speech-signal reminders corresponding to information from the scheduling software, where the digital timer system triggers the presentation of the speech-signal reminders at the appropriate time.

Additionally the Personal Audio Assistant can interface with an email platform through the communications port; The speech synthesis system converts the email in text to speech and provides email to the user in aural presentation format. The system further comprising: a process in the Registration engine allowing the user to optimize their personalization process of incoming emails by associating a specific Earcon with the importance of the incoming email. As such, normal priority email contains an introduction sound announcing to the user the level of importance the sender associated with their email; a speech recognition system for converting speech-to-text which interfaces with the control language as to provide navigational commands allowing the user to respond and send emails.

In at least one exemplary embodiment the communications port system makes use of some wireless communications protocol (802.11, Bluetooth®, A2DP, or other) to transmit and receive digital audio data for playback, the system further comprising: an audio codec to encode and decode digital audio transmissions; a wireless communications system (802.11, Bluetooth®, A2DP, etc.) for transmitting and receiving data (digital audio transmissions, Control Data, etc.); a method for pairing two or more Personal Audio Assistants through a wireless communications protocol to provide a secure exchange of audio content, data such as the user's Public Data; an audio warning signal or visual display system output that notifies the user anytime a compatible transmission becomes available; and a user control system enabling the user to switch between available compatible transmissions.

In at least one exemplary embodiment the system enables listeners to share digital audio transmissions, the system further comprising: a method for scanning for available digital audio transmissions within range; a user control interface for specifying digital audio transmission behavior; a method for employing the system as a relay to other compliant devices; re-broadcasting digital audio transmissions to increase wireless range. In at least one exemplary embodiment multiple systems are capable of sharing the contents of their program and data memory using the wireless communications protocol.

In at least one exemplary embodiment of the system, the input Acoustic Transducer is used to record audio content to the data memory storage system, the system further comprising: an implementation of some perceptual audio codec technology in the DSP, enabling the storage of compressed audio formats (e.g., MP3, AAC, FLAG, etc); and an increased data memory storage capacity for storing recorded audio content.

In at least one exemplary embodiment, the stereo input Acoustic Transducers are ultimately connected to the audio signal path at the DSP, allowing the user to audition Environmental Audio (e.g., speech or music) and mitigating the need for the user to remove the Headphone apparatus to audition Environmental Audio, the system further comprising: a stereo

pair of input Acoustic Transducers placed close to the user's ear canal input, conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE, see definitions); and by mounting the input Acoustic Transducers in a CIC or ITC configuration, spatial-acoustic cues are preserved, creating a spatially-accurate Environmental Audio input signal—essentially a personal binaural recording; a method for acoustically compensating for the non-linear frequency response characteristics of the Acoustical Isolation Cushions of a given Headphone system by applying corresponding inverse filters to the Environmental Audio input signal at the DSP; With this method, the system acts as a linear-frequency-response hearing protection apparatus (e.g., USPTO—60/805,985—Goldstein).

At least one exemplary embodiment includes a system for first attenuating Audio Playback and then mixing the Environmental Audio input signals, at a louder listening level, with the audio signal path using the DSP, where the system is activated by any combination of the following methods: a manual switch to activate/deactivate the system; a speech-detection apparatus to activate the system when speech is detected as the principal component of the Environmental Audio input; and a music-detection apparatus to activate the system when music is detected as the principal component of the Environmental Audio input.

At least one exemplary embodiment can include active noise reduction, echo cancellation and signal conditioning that can be environmentally customized through the registration process to better meet the user's specific needs (i.e., occupation-related noise cancellation); A typical application would be a special set of noise cancellation parameters tuned to the drilling equipment used by a dentist.

In at least one exemplary embodiment the input Acoustic Transducers are instead mounted within circum-aural, intra-aural BTE, or intra-aural ITE molds (see Ear Mold Style), the system further comprising: a spatial audio enhancement system for supplementing the spatial-acoustic cues captured by the stereo pair of input Acoustical Transducers to provide improved spatial perception of Environmental Audio using any combination of the following methods: the application of Generic, Semi-Personalized, or Personalized HRTF data to the Environmental Audio input signal; the application of binaural enhancement algorithms, familiar to those skilled in the art, to the Environmental Audio input signals; the application of a pinna simulation algorithm to the Environmental Audio input signal; and a synthetic pinna apparatus placed just before the stereo input Acoustic Transducers.

At least one exemplary embodiment includes a Server system for the creation, Registration, management, and delivery of Personalized Services, the system comprising: a communications system for interfacing with public communication networks to exchange data with Personal Audio Assistants, a Client's computer, mobile phones, PDAs or other capable devices; a database and database management system for storing and retrieving information relating to user Registration, Personalized Services, audio content, Control Data, and other data; a Registration interface system for collecting, storing, and applying information provided by users; a method for creating Personalized Services based on user Registration information; an end-user audio content Lock-Box storage system, providing every registered user access to their purchased media content; a business-to-business interface system for acquiring audio content with record labels, copyright holders, and other businesses; an E-tailing system including an electronic transactions system enabling users to purchase content, items offered for sale or pay subscription fees electronically; an E-Payment system compensating the various copyholders upon purchase of content by user; a Playlist engine, which acquires the user's Registration information, Listening History Envelope and then creates audio playlists, which is optimized for the user preferences and further refinements; and an Email server, which distributes communications to the user and others, regarding marketing data, the status of the user weekly SPL dose, and other information.

At least one exemplary embodiment includes machine-learning techniques employed to better optimize the user's preferences relating to audio content and other media content, the system further comprising: a method for tracking the purchase history of each user, relating the purchase history to media content preferences, and using the purchase history to make media content recommendations; a method for examining a user's digital media library, stored on a Personal Computer, Personal Music Player, or Personal Audio Assistant, from the Server system, and relating media content preferences and media content recommendations to the user's digital media library; and a method for examining a user's Listening History Profile.

At least one exemplary embodiment includes a Registration system for collecting a wide variety of information from users, including information necessary for creating Personalized Services, the system comprising: a Server system; an interface system for querying the user to collect registration information including demographics (age, gender), Playback Hardware information, Headphone information, occupational information, home and work locations, medication information, music-related preferences, video-related preferences, and other information; a method for customizing Control Data based on registration information; and a method for creating Personalized Services based on registration information.

In at least one exemplary embodiment a fast HRTF acquisition process is included as part of the Registration process, the system further comprising a method for the fast acquisition of Semi-Personalized HRTF data via a deduction process, the method comprising: a database system containing indexed, clustered HRTF data sets; an auditory test signal with distinctive spatial characteristics, where two or more distinct sound source locations exist; a system for the application of potential HRTF matches to the auditory test signal; and a feedback system, allowing the user to select the best listening experience from a number of candidate listening experiences, based on the spatial quality perceived in the HRTF-processed auditory test signal.

In at least one exemplary embodiment Personalized HRTF data is measured and used instead of Semi-Personalized HRTF data, by any method familiar to those skilled in the art.

In at least one exemplary embodiment the user is provided some Personal Audio Assistant free-of-charge or at a discount, given the user agrees to a subscription service commitment to receive Personalized Services for a certain amount of time.

In at least one exemplary embodiment, as part of the Personalized Services, the user is provided with temporary audio content corresponding to the preferences indicated during the registration process; Further, the user is given the option to purchase the audio content permanently; Otherwise, the audio content is replaced with new audio content from the Server, after a predetermined amount of time or a predetermined number of playback counts, the system comprising: a Personal Audio Assistant with an enhanced user control system, enabling a registered user to purchase media content directly from the Personal Audio Assistant with a button; and a Personal Audio Assistant with an enhanced user control

system, enabling a registered user to store a reference to media content that can be purchased by the user at a later time.

In at least one exemplary embodiment, video or gaming content is included as well as audio content, the system further comprising: a Personal Audio Assistant with an enhanced visual display system, capable of playing video and/or gaming content.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives medication reminders in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; Medication reminders are generated by the Server system based on the user's registration information.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives stock market information in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; The stock market information is selected by the Server system based on the user's registration information, the system further comprising: the user having successfully registered their Personal Audio Assistant with a brokerage firm, or other stock trading engines, the user can then purchase or sell a stock by use of a user button or a speech command.

Further in at least one exemplary embodiment, the user is able to request specific media content to be transferred temporarily or permanently to the user's Personal Audio Assistant, the system further comprising: an interface system operating on the Server allowing users to request specific media content by artist, title, genre, format, keyword search, or other methods familiar to those skilled in the art; and a media content search engine system.

In at least one exemplary embodiment a Relative Audiogram compensation filter is applied to audio signal path by the digital signal processor, the system either (e.g., USPTO—60/805,985—Goldstein): (a) Retrieves Relative Audiogram compensation information from a remote Server after a registration process (during transmission, the information can include HIPAA compliant encoding); or (b) calculates a compensation filter from Relative Audiogram information obtained by the system locally. For example U.S. Pat. No. 6,840,908—Edwards, and U.S. Pat. No. 6,379,314—Horn, discuss methods for the acquisition of an individual's Relative Audiogram.

In at least one exemplary embodiment a Satellite Radio transmitter/receiver (transceiver) is incorporated within the Headphone proper, allowing the user to at least: receive XM®, Sirius® and other broadcasts for playback over the system; select radio stations for playback over the system via the control system, the control system comprising either a single-click tactile interface or the speech-controlled circuitry; store selected portions of such broadcasts in memory for later recall and playback via the control systems; engage a novel commercial-skip feature for attenuating the playback level of suspected sales commercials broadcasts; and engage a speech-skip feature for attenuating the playback of speech (e.g., news, announcements, etc.).

At least one exemplary embodiment includes a Walkie-Talkie mode, which broadcasts input to the system's built-in microphone, whereby the user's speech can be detected by the input acoustic transducer and remotely broadcast where at least one of the following occurs: the Walkie-Talkie mode receives input via AM/FM broadcasts (as well as digital communications protocols) from a nearby user; the Walkie-Talkie mode allows nearby users to engage in conversation with increased perceptual clarity in noisy environments (e.g., aircraft cockpits), using for example a noise-cancellation system; selectively engage and disengage the Walkie-Talkie mode using the control system; detect other users of the system within a given range; and alert the user of the system when other detected systems contain certain Public Data and contain a predefined Public Message Key (e.g., "If the detected system belongs to a single male between the ages of 25 and 30 and whose favorite sport is tennis, then broadcast the message, 'I like tennis also; would you like to have coffee?'" or "If the detected system belongs to a user who attended Princeton University, then broadcast the message, 'Go Tigers!'").

At least one exemplary embodiment can use other communications to accomplish this service rather than AM/FM; as such the system can incorporate communications transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other protocols) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to other Personal Audio Assistants. At least one exemplary embodiment can selectively control the broadcast of public data and public message keys via the control system.

At least one exemplary embodiment includes a Sonification algorithm within the Headphone, which enables auditory display of digitally received data, including for example financial data, news, GPS data, the system further containing a variety of sonification "themes" selected during the registration process that map requested data (e.g., current average trading price of AAPL stock, the Dow Jones Industrial Index, and the Nasdaq Composite®) to corresponding audio content (e.g., the frequency of a sine tone presented in the left ear, the frequency of a sine tone presented in the right ear, and the global amplitude of both sine tones, respectively).

At least one exemplary embodiment includes an auditory display, which is synthesized by the onboard Digital Signal Processor. In at least one exemplary embodiment the auditory display is created through the digital audio signal processing effects applied to any other acoustic data the system is capable of reproducing (e.g., terrestrial radio, prepurchased audio content in the user's digital library, electronic books, etc.); For example, a sudden listening level increase in the playback level of a song to which the user was listening can be triggered by a predefined alert condition (e.g., Nasdaq Composite® has exceeded 2200 points).

At least one exemplary embodiment includes the ability to create themes using a computer program and uploading a file to the Headphone system.

At least one exemplary embodiment includes a speech recognition system for converting speech to HTML (Voice Browser), whereby the user can access the Internet, provide navigational commands, perform searches and receive results via the Headphones through a text (HTML)-speech synthesize.

Additionally, the Personal Audio Assistant can be totally incorporated with a mobile cell phone, or any portable technology which incorporates any of the following protocols, TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others known to those of ordinary skill in the arts via a communications network (e.g., the Internet, cellular networks), the system further comprising: an Acoustic Transducer constructed as part of the mobile cell phone or a series of Acoustic Transducers, which are constructed as part of mobile cell phone; a commutations path incorporated into the mobile cell phone providing for bi-directional communication with a Headphone array; the incorporation of the mobile cell phone's microphone(s) to act as the Environmental Audio Acoustical Transducer(s); and the incorporation of the mobile cell phone's keyboard or touch sensitive screen to

function as a manual input or to complement speech commands and that can act in a way to respond to Personalized Services offered to a user.

While the present embodiments have been described with reference to exemplary embodiments, it is to be understood that the claimed embodiments are 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 such modifications and equivalent structures and functions.

What is claimed is:

1. A wearable device, comprising:
- a housing for the wearable device;
- an ambient microphone within or on the housing;
- a communication module within or on the housing;
- a logic circuit communicatively coupled to the ambient microphone;
- a memory storage unit communicatively coupled to the logic circuit; and
- an interaction element, the interaction element and logic circuit cooperatively initiate control of media content from the wearable device or initiate operations of the wearable device;
- wherein the logic circuit is configured to cooperatively initiate control of media content of the wearable device in response to analysis of audio captured at the wearable device at the ambient microphone and analyzed at a remote server in response to instructions received from the remote server.

2. The wearable device of claim **1**, wherein the interaction element and logic circuit are configured to initiate control of access to the media content played back at the wearable device or mobile device coupled thereto.

3. The wearable device of claim **2**, wherein the memory storage unit and logic circuit are remote from the housing.

4. The wearable device of claim **2**, wherein the memory storage unit or logic circuit or both are remote from the housing.

5. The wearable device of claim **1**, wherein the wearable device comprises eye-wear.

6. The wearable device of claim **1**, wherein the interaction element comprises a voice controlled user interface.

7. The wearable device of claim **1**, wherein the wearable device further comprises a button with multiple contacts coupled to the logic circuit that forms a portion of the interaction element.

8. The wearable device of claim **1**, wherein the initiating of the operations of the wearable device is in response to a trigger of a predetermined alert condition.

9. The wearable device of claim **1**, wherein the wearable device comprises an earphone.

10. The wearable device of claim **1**, wherein the wearable device comprises at least a portion of a personal computer system, a personal music player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

11. The wearable device of claim **1**, wherein the wearable device further comprises speaker for providing an audible output.

12. The wearable device of claim **1**, wherein the wearable device further comprises a GPS receiver and wherein the communication module forms a peer-to-peer network using location information from the GPS receiver.

13. The wearable device of claim **1**, wherein the logic circuit is configured to perform a search of the web in response to a voice command and provide a web search result to the wearable device.

14. The wearable device of claim **1**, wherein the logic circuit is configured to cooperatively initiate control of the media content of the wearable device and initiate operations of the wearable device in response to analysis of the audio captured at the wearable device at the ambient microphone and analyzed at the remote server in response to instructions received from the remote server.

15. The wearable device of claim **1**, wherein the initiating of operations comprises commercial transactional operations.

16. The wearable device of claim **1**, wherein the logic circuit is configured to retrieve specific media content in response to voice commands captured at the wearable device using the ambient microphone, and the wearable device receives results from the remote server in response to speech recognition processing of the voice command at the remote server.

17. The wearable device of claim **1**, wherein the wearable device further comprises a presentation device operatively coupled to the wearable device for presenting one or more of text, images, webpages, location information, scheduling information, or audible outputs to complement detected voice commands at the wearable device and speech recognition results received at the wearable device from the remote server in response to analysis of the detected voice commands at the remote server.

18. A wearable device, comprising:
- a housing for the wearable device;
- an ambient microphone mounted on or within the housing;
- a communication module communicatively coupled to the ambient microphone;
- one or more processors communicatively coupled to the ambient microphone;
- a computer readable medium communicatively coupled to the one or more processors and containing instructions which when executed by the one or more processors, cause the one or more processors to perform functions comprising initiating control of audio content or initiating at least one operation of the wearable device; and
- wherein the one or more processors is configured to cooperatively initiate control of media content of the wearable device in response to analysis of audio captured at the wearable device at the ambient microphone and analyzed at a remote server in response to instructions received from the remote server.

19. The wearable device of claim **18**, wherein the one or more processors initiates control of the media content in response to a voice command received and processed by a voice controlled user interface, wherein a remote server receives and analyzes the processed voice command and sends results back to the wearable device.

20. A wearable device, comprising:
- a wearable structure having at least an ambient microphone;
- a communication module communicatively coupled to the ambient microphone;
- one or more processors communicatively coupled to the ambient microphone;
- a computer readable medium communicatively coupled to one or more processors and containing instructions which when executed by the one or more processors, cause the one or more processors to perform functions comprising initiating control of audio content accessed

at the wearable device in response to detected voice commands in the captured audio; and

wherein the one or more processors is configured to cooperatively initiate control of media content of the wearable device in response to analysis of audio captured at the wearable device at the ambient microphone and analyzed at a remote server in response to instructions received from the remote server.

* * * * *

Exhibit C

US008150044B2

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

(10) **Patent No.: US 8,150,044 B2**
(45) **Date of Patent: Apr. 3, 2012**

(54) **METHOD AND DEVICE CONFIGURED FOR SOUND SIGNATURE DETECTION**

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(73) Assignee: **Personics Holdings Inc.**, Boca Raton, FL (US)

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(22) Filed: **Dec. 28, 2007**

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H03G 3/20 (2006.01)
H04R 29/00 (2006.01)
H04R 25/00 (2006.01)

(52) **U.S. Cl.** **381/57**; 381/56; 381/60; 381/317

(58) **Field of Classification Search** 381/56, 381/57, 60, 317
See application file for complete search history.

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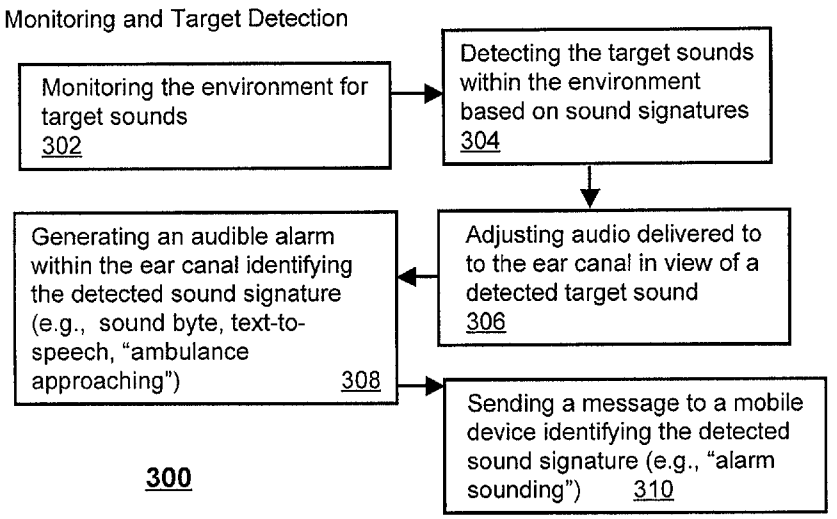
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Primary Examiner — Victor A Mandala
(74) *Attorney, Agent, or Firm* — RatnerPrestia

(57) **ABSTRACT**

Methods for personalized listening which can be used with an earpiece are provided. A method includes capturing ambient sound from an Ambient Sound Microphone (ASM) of an earpiece partially or fully occluded in an ear canal, monitoring the ambient sound for a target sound, and adjusting by way of an Ear Canal Receiver (ECR) in the earpiece a delivery of audio to an ear canal based on a detected target sound. A volume of audio content can be adjusted upon the detection of a target sound, and an audible notification can be presented to provide a warning.

21 Claims, 5 Drawing Sheets



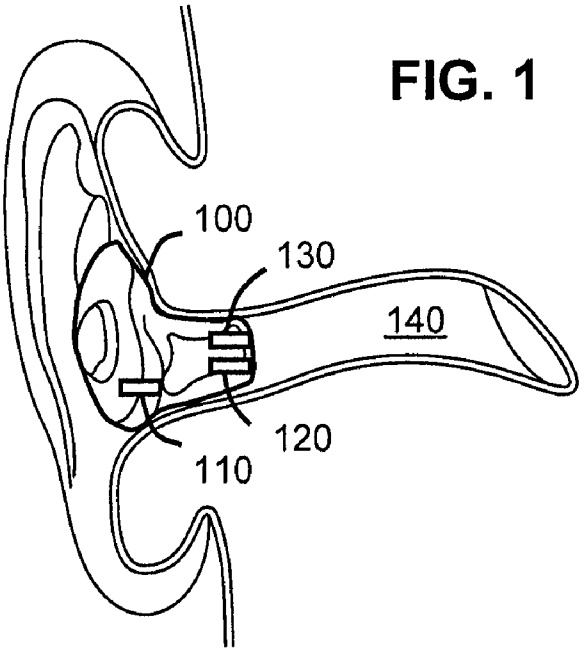

100
130
140
120
110

FIG. 1

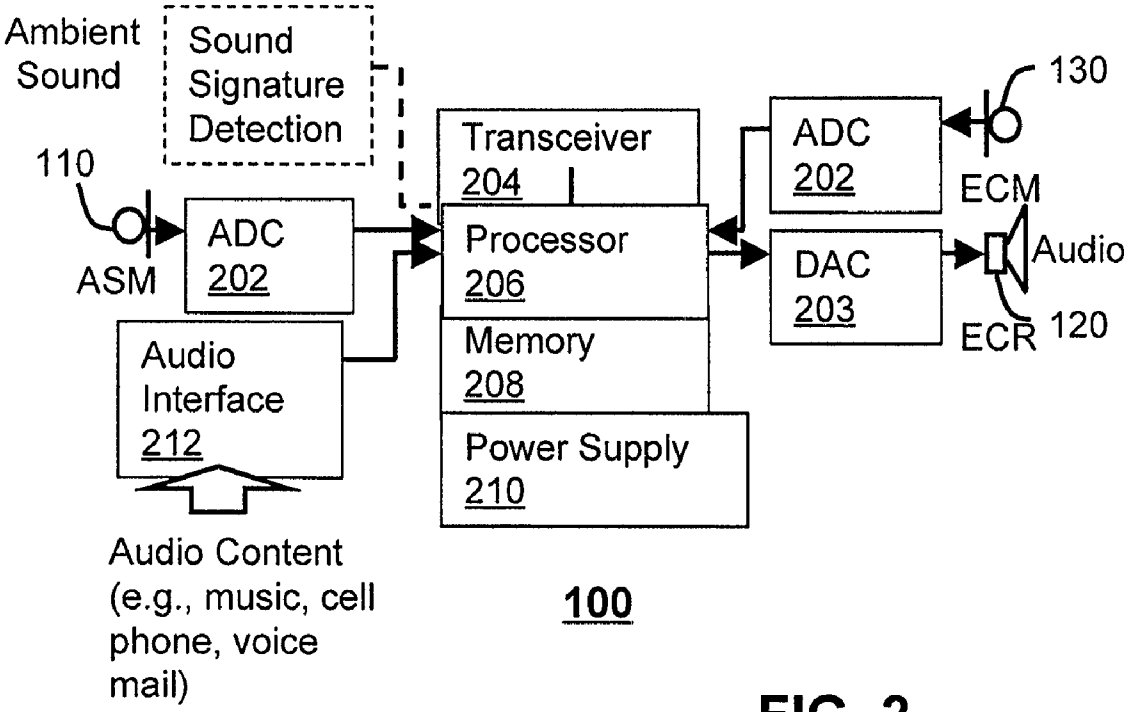

Ambient Sound
Sound Signature Detection
110
ASM
ADC 202
Audio Interface 212
Audio Content (e.g., music, cell phone, voice mail)
Transceiver 204
Processor 206
Memory 208
Power Supply 210
100
ADC 202
130
ECM
DAC 203
Audio
ECR
120

FIG. 2

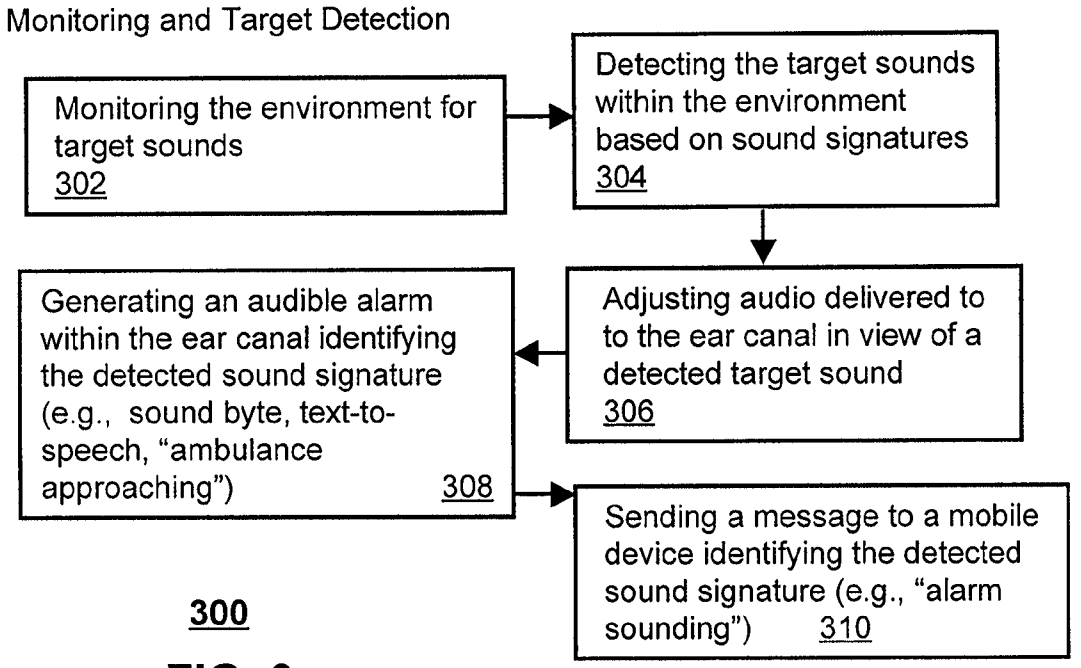

Monitoring and Target Detection
Monitoring the environment for target sounds 302
Detecting the target sounds within the environment based on sound signatures 304
Adjusting audio delivered to to the ear canal in view of a detected target sound 306
Generating an audible alarm within the ear canal identifying the detected sound signature (e.g., sound byte, text-to-speech, "ambulance approaching") 308
Sending a message to a mobile device identifying the detected sound signature (e.g., "alarm sounding") 310
300

FIG. 3

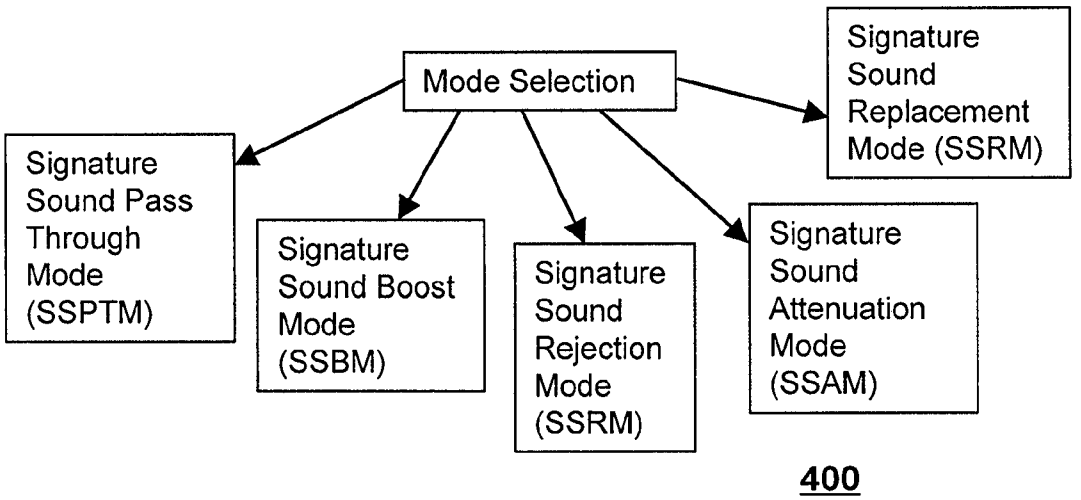

Mode Selection
Signature Sound Pass Through Mode (SSPTM)
Signature Sound Boost Mode (SSBM)
Signature Sound Rejection Mode (SSRM)
Signature Sound Attenuation Mode (SSAM)
Signature Sound Replacement Mode (SSRM)
400

FIG. 4

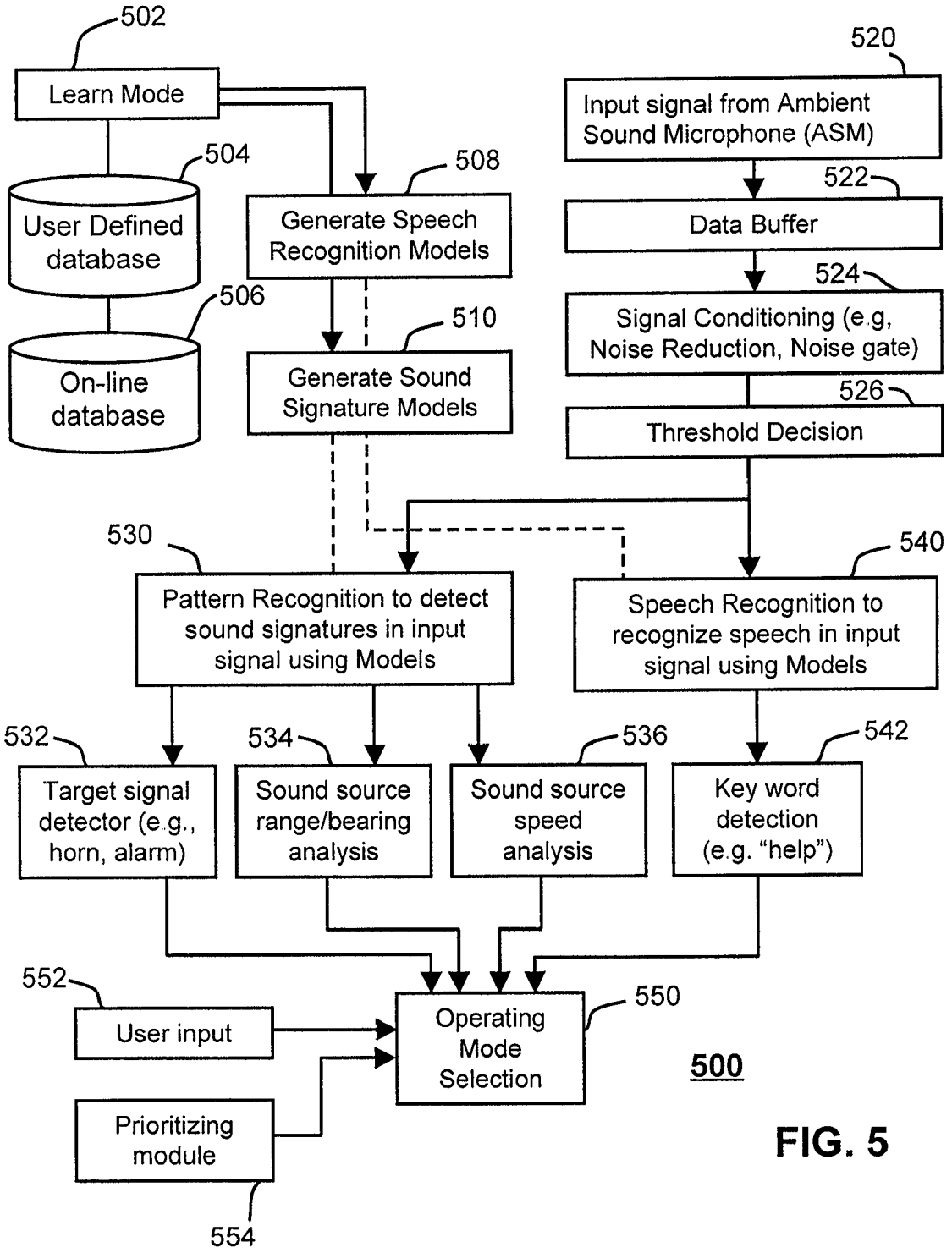

502
Learn Mode
504
User Defined database
506
On-line database
508
Generate Speech Recognition Models
510
Generate Sound Signature Models
520
Input signal from Ambient Sound Microphone (ASM)
522
Data Buffer
524
Signal Conditioning (e.g, Noise Reduction, Noise gate)
526
Threshold Decision
530
Pattern Recognition to detect sound signatures in input signal using Models
540
Speech Recognition to recognize speech in input signal using Models
532
Target signal detector (e.g., horn, alarm)
534
Sound source range/bearing analysis
536
Sound source speed analysis
542
Key word detection (e.g. "help")
552
User input
550
Operating Mode Selection
500
Prioritizing module
554

FIG. 5

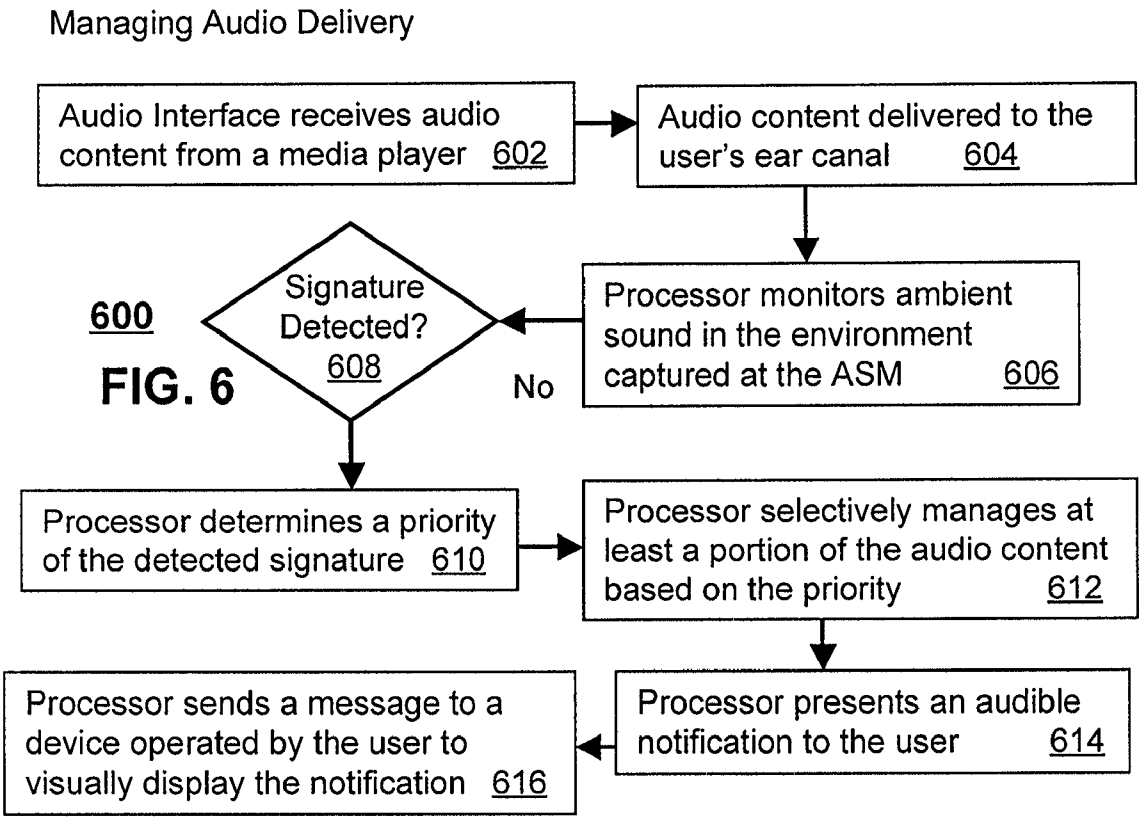

Managing Audio Delivery
Audio Interface receives audio content from a media player 602
Audio content delivered to the user's ear canal 604
Processor monitors ambient sound in the environment captured at the ASM 606
Signature Detected? 608
No
600
Processor determines a priority of the detected signature 610
Processor selectively manages at least a portion of the audio content based on the priority 612
Processor presents an audible notification to the user 614
Processor sends a message to a device operated by the user to visually display the notification 616

FIG. 6

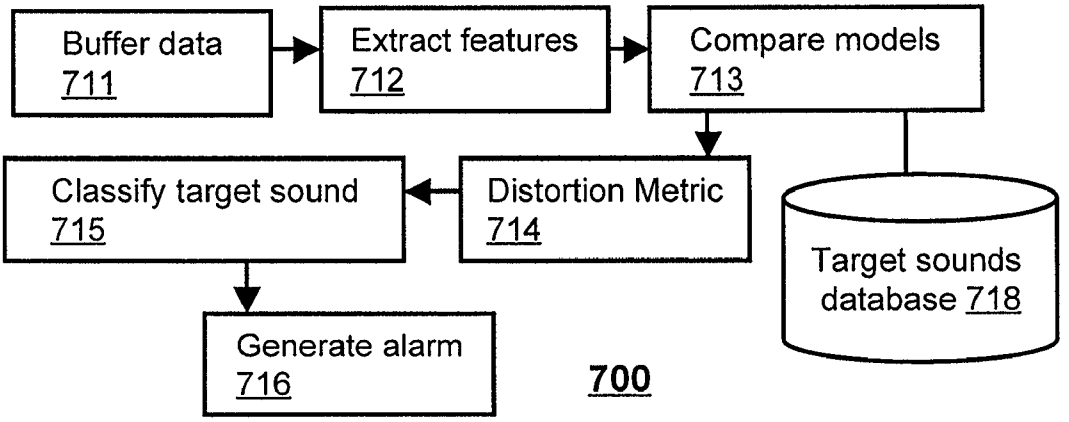
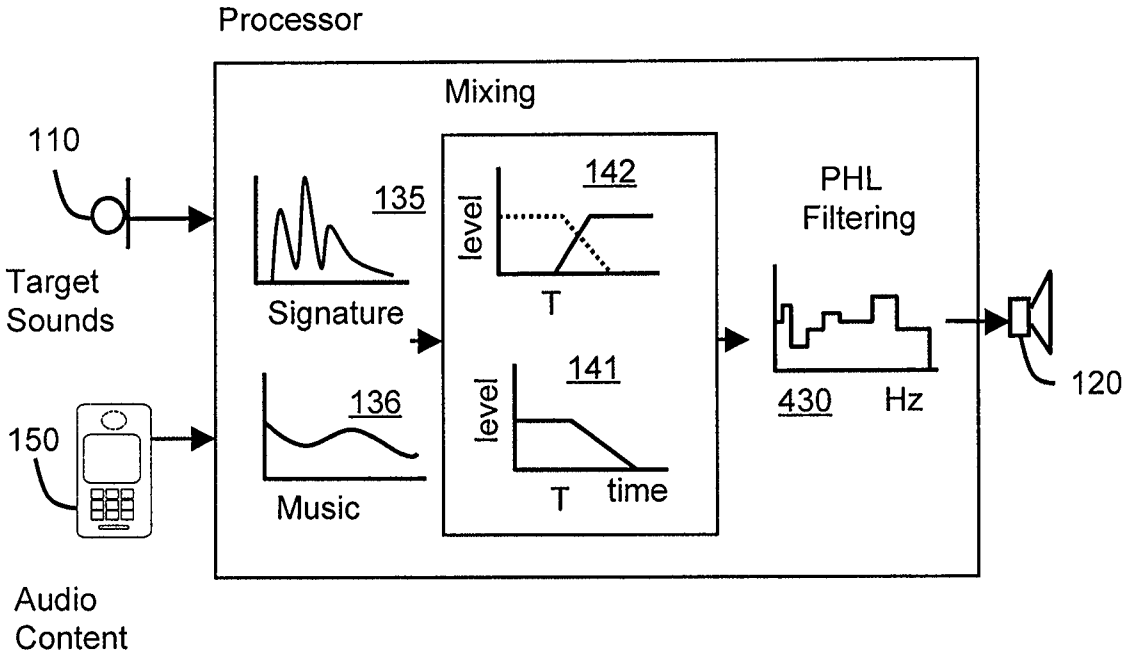

Buffer data 711
Extract features 712
Compare models 713
Classify target sound 715
Distortion Metric 714
Target sounds database 718
Generate alarm 716
700

FIG. 7


Processor
Mixing
110
Target
Sounds
135
Signature
142
level
T
PHL
Filtering
120
150
136
Music
141
level
T
time
430
Hz
Audio
Content

800

FIG. 8

METHOD AND DEVICE CONFIGURED FOR SOUND SIGNATURE DETECTION

CROSS REFERENCE TO RELATED APPLICATIONS

This Application is a Non-Provisional and claims the priority benefit of Provisional Application No. 60/883,013 filed on Dec. 31, 2006, the entire disclosure of which is incorporated herein by reference.

FIELD

The present invention relates to a device that monitors target (e.g. warning) sounds, and more particularly, though not exclusively, to an earpiece and method of operating an earpiece that detects target sounds.

BACKGROUND

Excess noise exposure can generate auditory fatigue, possibly compromising 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. Some of the sounds in the environment may correspond to warnings, such as those associated with an alarm or siren. A person that can hear the warning sounds can generally react in time to avoid danger. In contrast, a person that cannot adequately hear the warning sounds, or whose hearing faculties have been compromised due to auditory fatigue, may be susceptible to danger.

Environmental noise can mask warning sounds and impair a person's judgment. Moreover, when people wear 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 target sounds within their environment. In some cases, the user may even turn up the volume to hear their personal audio over environmental noises. This can put the user in a compromising situation since they may not be aware of target sounds in their environment. It also puts them at high sound exposure risk, which can potentially cause long term hearing damage.

A need therefore exists for enhancing the user's ability to hear target sounds in their environment without compromising his hearing.

SUMMARY

At least one exemplary embodiment is directed to a method and device for sound signature detection.

In at least one exemplary embodiment, an earpiece, can include an Ambient Sound Microphone (ASM) configured to capture ambient sound, at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal, and a processor operatively coupled to the ASM and the at least one ECR to monitor target sounds in the ambient sound. Target (e.g., warning) sounds can be amplified, attenuated, or reproduced and reported to the user by way of the ECR. As an example, the target (e.g., warning) sound can be an alarm, a horn, a voice, or a noise. The processor can detect sound signatures in the ambient sound to identify the target (e.g., warning) sounds and adjust the audio delivered to the ear canal based on detected sound signatures.

In a second exemplary embodiment, a method for personalized listening suitable for use with an earpiece is provided.

The method can include capturing ambient sound from an Ambient Sound Microphone (ASM) of an earpiece that is partially or fully occluded in an ear canal, monitoring the ambient sound for a target sound, and adjusting by way of an Ear Canal Receiver (ECR) in the earpiece a delivery of audio to an ear canal based on a detected target sound. The method can include passing, amplifying, attenuating, or reproducing the target sound for delivery to the ear canal.

In a third exemplary embodiment a method for personalized listening suitable for use with an earpiece can include the steps of capturing ambient sound from an Ambient Sound Microphone (ASM) of an earpiece that is partially or fully occluded in an ear canal, detecting a sound signature within the ambient sound that is associated with a target sound, and mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound. A direction and speed of a sound source generating the target sound can be determined, and presented as a notification to a user of the earpiece. The method can include detecting a spoken utterance in the ambient sound that corresponds to a verbal warning or help request.

In a fourth exemplary embodiment a method for sound signature detection can include capturing ambient sound from an Ambient Sound Microphone (ASM) of an earpiece, and receiving a directive to learn a sound signature within the ambient sound. The method can include receiving a voice command or detecting a user interaction with the earpiece to initiate the step of capturing and learning. A sound signature can be generated for a target sound in the environment and saved to a memory locally on the earpiece or remotely on a server.

In a fifth exemplary embodiment a method for personalized listening can include capturing ambient sound from an Ambient Sound Microphone (ASM) of an earpiece that is partially or fully occluded in an ear canal, detecting a sound signature within the ambient sound that is associated with a target sound, and mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound and a personalized hearing level (PHL). The method can include retrieving from a database learned models, comparing the sound signature to the learned models, and identifying the target sound from the learned models in view of the comparison. Auditory queues in the target sound can be enhanced relative to the audio content based on a spectrum of the ambient sound captured at the ASM. A perceived direction of a sound source generating the target sounds can be spatialized using Head Related Transfer Functions (HRTFs).

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 ambient sound monitoring and target detection in accordance with an exemplary embodiment;

FIG. **4** illustrates earpiece modes in accordance with an exemplary embodiment;

FIG. **5** illustrates a flowchart of a method for sound signature detection in accordance with an exemplary embodiment;

FIG. **6** is a flowchart of a method for managing audio delivery based on detected sound signatures in accordance with an exemplary embodiment;

FIG. **7** is a flowchart for sound signature detection in accordance with an exemplary embodiment; and

FIG. **8** is a pictorial diagram for mixing ambient sounds and target sounds with audio content 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 target 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. Earpiece **100** includes an Ambient Sound Microphone (ASM) **110** to capture ambient sound, an Ear Canal Receiver (ECR) **120** to deliver audio to an ear canal **140**, and an ear canal microphone (ECM) **130** to assess a sound exposure level within the ear canal. The earpiece **100** can partially or fully occlude the ear canal **140** to provide various degrees of acoustic isolation.

The earpiece **100** can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality to ensure safe reproduction levels. The earpiece **100** in various exemplary embodiments can provide listening tests, filter sounds in the environment, monitor target sounds in the environment, present notifications based on identified target sounds, adjust audio content levels with respect to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL). The earpiece **100** is suitable for use with users having healthy or abnormal auditory functioning. 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. Accordingly, the earpiece **100** can be partially or fully occluded in the ear canal.

As part of its operation, the earpiece **100** can generate an Ear Canal Transfer Function (ECTF) to model the ear canal **140** using ECR **120** and ECM **130**. The ECTF can be used to establish a personalized hearing level profile. The earpiece **100** can also determine a sealing profile with the user's ear to compensate for any sound leakage. In one configuration, the earpiece **100** can provide personalized full-band width general audio reproduction within the user's ear canal via timbral equalization based on the ECTF to account for a user's hearing sensitivity. The earpiece **100** also provides Sound Pressure Level dosimetry to estimate sound exposure of the ear and associated recovery times from excessive sound exposure. This permits the earpiece **100** to safely administer and monitor sound exposure to the ear.

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**, ECR **120**, and ECM **130** 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** for target sounds in the environment, such as an alarm (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 memory **208** can store sound signatures for previously learned target sounds from which the processor **206** refers to for detecting target sounds. The sound signatures can be resident in the memory **208** or downloaded to the earpiece **100** via the transceiver **204** during operation as needed. Upon detecting a target sound, the processor **206** can report the target to the user via audio delivered from the ECR **120** to the ear canal.

The earpiece **100** can also include an audio interface **212** operatively coupled to the processor **206** to receive audio content, for example from a media player, and deliver the audio content to the processor **206**. The processor **206** responsive to detecting target sounds can adjust the audio content and the target sounds delivered to the ear canal. The processor **206** can 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 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 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 target 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 exemplary 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 earpiece monitoring and target detection 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 device.

The method **300** can start in a state wherein the earpiece **100** has been inserted and powered on. As shown in step **302**, the processor **206** can monitor the environment for target sounds, such as an alarm, a horn, a voice, or a noise. Each of the target sounds can have certain identifiable features that characterize the sound. The features can be collectively referred to as a sound signature which can be used for recognizing the target sound. As an example, the sound signature may include statistical properties or parametric properties of the target sound. For example, a sound signature can describe prominent frequencies with associated amplitude and phase information. As another example, the sound signature can contain principal components identifying the most likely recognizable features of a target sound.

The processor **206** at step **304** can then detect the target sounds within the environment based on the sound signatures. As will be shown ahead, feature extraction techniques are applied to the ambient sound captured at the ASM **110** to generate the sound signatures. Pattern recognition approaches are applied based on known sound signatures to detect the target sounds from their corresponding sound signatures. More specifically, sound signatures can then be compared to learned models to identify a corresponding target sound. Notably, the processor **206** can detect sound signatures from the ambient sound regardless of the state of the earpiece **100**. For example, the earpiece **100** may be in a listening state wherein ambient sound is transparently passed to the ECR **120**, in a media state wherein audio content is delivered from the audio interface **212** to the ECR **120**, or in an active listening state wherein sounds in the environment are selectively enhanced or suppressed.

At step **306**, the processor **206** can adjust sound delivered to the ear canal in view of a detected target sound. For instance, if the earpiece is in a listening state, the processor **206** can amplify detected target sounds in accordance with a Personalized Hearing Level (PHL). The PHL establishes comfortable and uncomfortable levels of hearing, and can be referenced by the processor **206** to set the volume level of the target sound (or ambient sound) so as not to exceed the user's preferred listening levels. As another example, if the earpiece is in a media state, the processor **206** can attenuate the audio content delivered to the ear canal, and amplify the target sounds in the ear canal. The PHL can also be used to properly mix the volumes of the different sounds. As yet another example, if the earpiece **100** is in an active state, the processor **206** can selectively adjust the volume of the target sounds relative to background noises in the environment.

The processor **206** can also compensate for an ear seal leakage due to a fitting of the earpiece **100** with the ear canal. An ear seal profile can be generated by evaluating amplitude and phase difference between the ASM **110** and the ECM **130** for known signals produced by the ECR **120**. That is, the processor **206** can monitor and report transmission levels of frequencies through the ear canal **140**. The processor **206** can take into account the ear seal leakage when performing audio enhancement, or other spectral enhancement techniques, to maintain minimal audibility of the ambient noise while audio content is playing.

Upon detecting a target sound in the ambient sound of the user's environment, the processor **206** at step **308** can generate an audible alarm within the ear canal that identifies the detected sound signature. The audible alarm can be a reproduction of the target sound, an amplification of the target sound (or the entire ambient sound), a text-to-speech message (e.g. synthetic voice) identifying the target sound, a haptic vibration via a motor in the earpiece **100**, or an audio clip. For example, the earpiece **100** can play a sound bite (i.e., audio clip) corresponding to the detected target sound such as an ambulance, fire engine, or other environmental sound. As another example, the processor **206** can synthesize a voice to describe the detected target sound (e.g., "ambulance approaching"). At step **310**, a message may be sent to a mobile device identifying the detected sound signature (e.g., "alarm sounding").

FIG. **4** illustrates earpiece modes **400** in accordance with an exemplary embodiment. The earpiece mode can be manually selected by the user, for example, by pressing a button, or automatically selected, for example, when the earpiece **100** detects it is in an active listening state or in a media state. As shown in FIG. **4**, the earpiece mode can correspond to Signature Sound Pass Through Mode (SSPTM), Signature Sound Boost Mode (SSBM), Signature Sound Rejection Mode (SSRM), Signature Sound Attenuation Mode (SSAM), and Signature Sound Replacement Mode (SSRM).

In SSPTM mode, ambient sound captured at the ASM **110** is passed transparently to the ECR **120** for reproduction within the ear canal. In this mode, the sound produced in the ear canal sufficiently matches the ambient sound outside the ear canal, thereby providing a "transparency" effect. That is, the earpiece **100** recreates the sound captured at the ASM **110** to overcome occlusion effects of the earpiece **100** when inserted within the ear. The processor **206** by way of sound measured at the ECM **130** 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, as though the earpiece **100** were absent in the ear canal. In one configuration, the processor **206** can predict an approximation of an equalizing filter to provide the transparency by comparing an ASM **110** signal and an ECM **130** signal transfer function.

In SSBM, target sounds and/or ambient sounds are amplified upon the processor **206** detecting a target sound. The target sound can be amplified relative to the normal level received, or amplified above an audio content level if audio content is being delivered to the ear canal. As noted previously, the target sound can also be amplified in accordance with a user's PHL to be within safe hearing levels, and within subjectively determined listening levels.

In SSRM, target sounds detected in the environment can be replaced with audible warning messages. For example, the processor **206** upon detecting a target sound can generate synthetic speech identifying the target sound (e.g., "ambulance detected"). In such regard, the earpiece **100** audibly reports the target sound identified thereby relieving the user from having to interpret the target sound. The synthetic speech can be mixed with the ambient sound (e.g., amplified, attenuated, cropped, etc.), or played alone with the ambient sound muted.

In SSAM, sounds other than target sounds can be attenuated. For instance, annoying sounds or noises not associated with target sounds can be suppressed. For instance, by way of a learning session, the user can establish what sounds are considered target sounds (e.g., "ambulance") and which sounds are non-target sounds (e.g. "jackhammer"). The processor **206** upon detecting non-target sounds can thus attenuate these sounds within the occluded or partially occluded ear canal.

FIG. **5** is a flowchart of a method **500** for a method for sound signature detection in accordance with an exemplary embodiment. 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 device.

The method can start at step **502**, in which the earpiece **100** can enter a learn mode. Notably, the earpiece upon completion of a learning mode or previous learning configuration can start instead at step **520**. In the learning mode of step **502**, the earpiece **100** can actively generate and learn sound signatures from ambient sounds within the environment. In learning mode, the earpiece **100** can also receive previously trained learning models to use for detecting target sounds in the environment. In an active learning mode, the user can press a button or otherwise (e.g. voice recognition) initiate a recording of ambient sounds in the environment. For example, the user can upon hearing a new target sound in the environment ("car horn"), activate the earpiece **100** to learn the new target sound. Upon generating a sound signature for the new target sound, it can be stored in the user defined database **504**. In another arrangement, the earpiece **100** upon detecting a unique sound, characteristic to a target sound, can ask the user if they desire to have the sound signature for the unique sound learned. In such regard, the earpiece **100** actively senses sounds and queries the user about their environment to learn the sounds. Moreover, the earpiece can organize learned sounds based on environmental context, for example, in outdoor (e.g. traffic, car, etc.) or indoor (e.g., restaurant, airport) environments.

In another learning mode, trained models can be retrieved from an on-line database **506** for use in detecting target sounds. The previously learned models can be transmitted on a scheduled basis to the earpiece, or as needed, depending on the environmental context. For example, upon the earpiece **100** detecting traffic noise, sound signature models associated with target sounds (e.g., ambulance, police car) in traffic can be retrieved. In another exemplary embodiment, upon the earpiece **100** detecting conversational noise (e.g. people talking), sound signature models for verbal warnings ("help", "police") can be retrieved. Groups of sound signature models can be retrieved based on the environmental context or on user directed action.

As shown in step **508**, the earpiece can also generate speech recognition models for target sounds corresponding to voice, such as "help", "police", "fire", etc. The speech recognition models can be retrieved from the on-line database **506** or the user defined database **504**. In the latter for example, the user can say a word or enter a text version of a word to associate with a verbal warning sound. For instance, the user can define a set of words of interest along with mappings to their meanings, and then use keyword spotting to detect their occurrences. If the user enters an environment wherein another individual says the same word (e.g., "help") the earpiece **100** can inform the user of the verbal warning sound. For other acoustic sounds, the earpiece **100** can generate sound signature models as shown in step **510**. Notably, the earpiece **100** itself can generate the sound signature models, or transmit the captured target sounds to external systems (e.g., remote server) that generate the sound signature models. Such learning can be conducted off-line in a training phase, and the earpiece **100** can be uploaded with the new learning models.

It should also be noted that the learning models can be updated during use of the earpiece, for example, when the earpiece **100** detects target sounds. The detected target sounds can be used to adapt the learning models as new target sound variants are encountered. For example, the earpiece **100** upon detecting a target sound, can use the sound signature of the target sound to update the learned models in accordance with the training phase. In such an exemplary embodiment a first learned model is adapted based on new training data collected in the environment by the earpiece. In such regard, for example, a new set of "horn" target sounds could be included in real-time training without discarding the other "horn" sounds already captured in the existing model.

Upon completion of learning, uploading, or retrieval of sound signature models, the earpiece **100** can monitor and report target sounds within the environment. As shown in step **520**, ambient sounds (e.g. input signal) within the environment are captured by the ASM **110**. The ambient sounds can be digitized by way of the ADC **202** and stored temporarily to a data buffer in memory **208** as shown in step **522**. The data buffer holds enough data to allow for generation of a sound signature as will be described ahead in FIG. **7**.

In another configuration, the processor **206** can implement a "look ahead" analysis system by way of the data buffer for reproduction of pre-recorded audio content, using a data buffer to offset the reproduction of the audio signal. The look-ahead system allows the processor to analyze potentially harmful audio artifacts (e.g. high level onsets, bursts, etc.) either received from an external media device, or detected with the ambient microphones, in-situ before it is reproduced. The processor **206** can thus mitigate the audio artifacts in advance to reduce timbral distortion effects caused by, for instance, attenuating high level transients.

At step **524**, signal conditioning techniques can be applied to the ambient sound for example to suppress noise or gate the noise to a predetermined threshold. Other signal processing steps such as threshold detection shown in step **526** can be employed to determine whether ambient sounds should be evaluated for target sounds. For instance, to conserve computational processing resources (e.g., battery, processor) only ambient sounds that exceed a predetermined power level are evaluated for target sounds. Other metrics such as signal spectrum, duration, and stationarity are considered in determining whether the ambient sound is analyzed for target sounds. Notably, other metrics (e.g., context aware) can also be employed to determine when the ambient sound should be processed for target sound detection.

If at least one property (e.g., power, spectral shape, duration, etc) of the ambient sound exceeds a threshold (or adaptive threshold), the earpiece **100** at step **530** can proceed to generate a sound signature for the ambient sound. In one exemplary embodiment the sound signature is a feature vector which can include statistical parameters or salient features of the ambient sound. An ambient sound with a target sound (e.g. "bell", "siren"), such as shown in step **532**, is generally expected to exhibit features similar to sound signatures for similar target sounds (e.g. "bell", "siren") stored in the user defined database **504** or the on-line database **506**. The earpiece **100** can also identify a direction and speed of the sound source if it is moving, for example, by evaluating Doppler shift as shown in step **534** and **536**. The earpiece **100**, by way of beam-forming among multiple ASM microphones can also estimate a direction of a sound source generating the target sound. In another arrangement, when dual earpieces **100** are used, or when multiple ASMs are employed, the distance and bearing of a sound source can be calculated by frequency dependent magnitude and phase between ASMs **110** (e.g. left

and right). The speed and bearing of the sound source can also be estimated using pitch analysis to detect changes predicted by Doppler effect, or alternatively by an analysis in changes in relative phase and magnitude between the two ASM signals. The earpiece **100**, by way of a sound recognition engine, can detect general target signals such as car horns or emergency sirens (and other signals referenced by ISO 7731) using spectral and temporal analysis.

The earpiece **100** can also analyze the ambient sound to determine if a verbal target (e.g. "help", "police", "excuse me") is present. As shown in step **540**, the sound signature of the ambient sound can be analyzed for speech content. For instance, the sound signature can be analyzed for voice information, such as vocal cord pitch periodicities, time-varying voice formant envelopes, or other articulation parameter attributes. Upon detecting the presence of voice in the ambient sound, the earpiece **100** can perform key word detection (e.g. "help") in the spoken content as shown in step **542**. Speech recognition models as well as language models can be employed to identify key words in the spoken content. As previously noted, the user can themselves say or enter in one or more target sounds that can be mapped to associated learning models for sound signature detection.

As shown in step **552**, the user can also provide user input to direct operation of the earpiece, for example, to select an operational mode as shown in **550**. As one example, the operation mode can enable, disable or adjust monitoring of target sounds. For instance, in listening mode, the earpiece **100** can mix audio content with ambient sound while monitoring for target sounds. In quiet mode, the earpiece **100** can suppress all noises except detected target sounds. The user input may be in the form of a physical interaction (e.g., button press) or a vocalization (e.g., spoken command). The operating mode can also be controlled by a prioritizing module as shown in step **554**. The prioritizing module prioritizes target sounds based on severity and context. For example, if the user is in a phone call, and a target sound is detected, the earpiece **100** can audibly inform the user of the warning and/or present a text message of the target sound. If the user is listening to music, and a target sound is detected, the earpiece **100** can automatically shut off the music and alert the user. The user, by way of a user interface or administrator, can rank target sounds and instruct the earpiece **100** how to respond to targets in various contexts.

FIG. **6** is a flowchart of a method **600** for managing audio delivery based on detected sound signatures in accordance with an exemplary embodiment. 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 device.

As noted previously, the audio interface **212** can supply audio content (e.g., music, cell phone, voice mail, etc) to the earpiece **100**. In such regard, the user can listen to music, talk on the phone, receive voice mail, or perform other audio related tasks while the earpiece **100** additionally monitors target sounds in the environment. During normal use, when a target sound is not present, the earpiece **100** can operate normally to recreate the sound experience requested by the user. If however the earpiece **100** detects a target sound, the earpiece **100** can manage audio content delivery to notify the user of the target sound. Managing audio content delivery can include adjusting or overriding other current audio settings.

By way of example, as shown in step **602**, the audio interface **212** receives audio content from a media player, such as a portable music player, or cell phone. The audio content can be delivered to the user's ear canal by way of the ECR **120** as shown in step **604**. The processor **206** can regulate the delivery of audio to the ear canal such that the sound pressure level dose is within safe limits. For instance, the processor **206** can adjust the audio level in accordance with a personalized hearing level (PHL) previously established for the user. The PHL provides upper and lower volume bounds across frequency for establishing comfortable listening levels.

At step **606**, the processor **206** monitors ambient sound in the environment captured at the ASM **110**. Ambient sound can be sampled at sufficiently data rates (e.g. 8, 16, and 32 KHz) to allow for feature extraction of sound signatures. Moreover, the processor **206** can adjust the sampling rate based on the information content of the ambient signal. For example, upon the ambient sound exceeding a first threshold, the sampling rate can be set to a first rate (e.g. 4 KHz). As the ambient sound increases in volume, or as prominent features are identified, the sampling rate can be increased to a second rate (e.g. 8 KHz) to increase signal resolution. Although, the higher sampling rate improves resolution of features, the lower sampling rate can preserve use of computational resources for minimally sufficient feature resolution (e.g., battery, processor).

If at step **608**, a sound signature is detected, the processor **206** can then determine a priority of the detected sound signature (at step **610**). The priority establishes how the earpiece **100** manages audio content. Notably, target sounds for various environmental conditions and user experiences can be learned. Accordingly, the user or an administrator, can establish priorities for target sounds. Moreover, these priorities can be based on environmental context. For example, if a user is in a warehouse where loading vehicles emit a beeping sound, sound signatures for such vehicles can be given the highest priority. A user can also prioritize learned target sounds for example via a user interface on a paired device (e.g., cell phone), or via speech recognition (e.g., "prioritize—'ambulance'—high").

Upon detecting a target sound and identifying a priority, the processor **206** at step **612** selectively manages at least a portion of the audio content based on the priority. For example, if the user is listening to music during the time a target sound is detected, the processor **206** can decrease the music volume to present an audible notification. This is one indication that the earpiece **100** has detected a target sound. At step **614**, the processor can further present an audible notification to the user. For instance, upon detecting a "horn" sound, a speech-to-text message can be presented to the user to audibly inform them that a horn sound has been detected (e.g., "horn detected"). Information related to the target sound (e.g., direction, speed, priority, etc.) can also be presented with the audible notification.

In a further arrangement, the processor **206** can send a message to a device operated by the user to visually display the notification as shown in step **616**. For example, if the user has disengaged audible notification, the earpiece **100** can transmit a text message to a paired device (e.g. cell phone) containing the audible warning. Moreover, the earpiece **100** can beacon out an audible alarm to other devices within a vicinity, for example via Wi-Fi (e.g., IEEE 802.16x). Other devices in the proximity of the user can sign up to receive audible alarms from the earpiece **100**. In such regard, the earpiece **100** can beacon a warning notification to other devices in the area to share warning information with other users.

FIG. **7** is a flowchart of a method **700** further describing sound signature detection in accordance with an exemplary embodiment. The method **700** can be practiced with more or less than the number of steps shown and is not limited to the order shown. The method **700** can begin in a state in which the earpiece **100** is actively monitoring target sounds in the environment.

At step **711**, ambient sound captured from the ASM **110** can be buffered into short term memory as frames. As an example, the ambient sound can be sampled at 8 KHz with 10-20 ms frame sizes (80 to 160 samples). The frame size can also vary depending on the energy level of the ambient sound. For example, the processor **206** upon detecting low level sounds (e.g., 70-74 dB SPL) can use a frame size of 30 ms, and update the frame size to 10 ms as the power level increases (e.g. >86 dB SPL). The processor **206** can also increase the sampling rate in accordance with the power level and/or a duration of the ambient sound. (A longer frame size with lower sampling can compromise resolution for computational resources.) The data buffer is of sufficient length to hold a history of frames (e.g. 10-15 frames) for short-term historical analysis.

At step **712**, the processor **206** can perform feature extraction on the frame as the ambient sound is buffered into the data buffer. As one example, feature extraction can include performing a filter-bank analysis and summing frequencies in auditory bandwidths. Features can also include Fast Fourier Transform (FFT) coefficients, Discrete Cosine Transform (DCT) coefficients, cepstral coefficients, PARCOR coefficients, wavelet coefficients, statistical values (e.g., energy, mean, skew, variance), parametric features, or any other suitable data compression feature set. Additionally, dynamic features, such as derivatives of any order, can be added to the static feature set. As one example, mel-frequency-cepstral analysis can be performed on the frame to generate between 10-16 mel-frequency-cepstral coefficients. The small number of coefficients represent features that can be compactly stored to memory for that particular frame. Such front end feature extraction techniques reduce the amount of data needed to represent the data frame.

At step **713**, the features can be incorporated as a sound signature and compared to learned models, for example, those retrieved from the target sounds database **718** (e.g., user defined database **504** or the on-line database **506** of FIG. **5**). A sound signature can be defined as a sound in the user's ambient environment which has significant perceptual saliency. As an example, a sound signature can correspond to an alarm, an ambulance, a siren, a horn, a police car, a bus, a bell, a gunshot, a window breaking, or any other target sound, including voice. The sound signature can include features characteristic to the sound. As an example, the sound signature can be classified by statistical features of the sound (e.g., envelope, harmonics, spectral peaks, modulation, etc.).

Notably, each learned model used to identify a sound signature has a set of features specific to a target sound. For example, a feature vector of a learned model for an "alarm" is sufficiently different from a feature vector of a learned model for a "bell sound". Moreover, the learned model can describe interconnectivity (e.g., state transitions, emission probabilities, initial probabilities, synaptic connections, hidden layers) among the feature vectors (e.g. frames). For instance, the features of a "bell" sound may change in a specific manner compared to the features of an "alarm" sound. The learned model can be a statistical model such as a Gaussian mixture model, a Hidden Markov Model (HMM), a Bayes Classifier, or a Neural Network (NN) that requires training.

In the foregoing, a Gaussian Mixture Model (GMM) is presented, although it should be noted that any of the above models can be used for sound signature detection. In this case, each target sound can have an associated GMM used for detecting the target sound. As an example, the target sound for an "alarm" will have its own GMM, and a target sound for a "bell" will have its own GMM. Separate GMMs can also be used as a basis for the absence of the sounds ("anti-models"), such as "not alarm" or "not bell." Each GMM provides a model for the distribution of the feature statistics for each target sound in a multi-dimensional space. Upon presentation of a new feature vector, the likelihood of the presence of each target sound can then be calculated. In order to detect a target sound, each target sound's GMM is evaluated relative to its anti-model, and a score related to the likelihood of that target sound is computed. A threshold can be applied directly to this score to decide whether the target sound is present or absent. Similarly, the sequence of scores can be relayed to yet another module which uses a more complex rule to decide presence or absence. Examples of such rules include linear smoothing or median filtering.

As previously noted, a HMM model or NN model with their associated connection logic can be used in place of each GMM for each learning model. For instance, each target sound in the database (**718** see FIG. **7**) can have a corresponding HMM. A sound signature for a target sound captured at the ASM **110** in ambient sound can be processed through a lattice network (e.g. Viterbi network) for comparison to each HMM to determine which HMM corresponds to the target sound, if any. Alternatively, in a trained NN, the sound signature can be input to the NN wherein the output states of the NN correspond to target sound indices. The NN can include various topologies such as a Feed-Forward, Radial Basis Function, Hopfield, Time-Delay Recurrent, or other optimized topologies for real-time sound signature detection.

At step **714**, a distortion metric is performed with each learned model to determine which learned models are closest to the captured feature vector (e.g., sound signature). The learned model with the smallest distortion (e.g., mathematical distance) is generally considered the correct match, or recognition result. It should also be noted that the distortion can be calculated as part of the model comparison in step **713**. This is because the distortion metric may depend on the type of model used (e.g., HMM, NN, GMM, etc) and in fact may be internal to the model (e.g. Viterbi decoding, back-propagation error update, etc). The distortion module is merely presented in FIG. **7** as a separate component to suggest use with other types of pattern recognition methods or learning models.

Upon evaluating the feature vector (e.g. sound signature) against the candidate target sound learned models, the ambient sound at step **715** can be classified as a target sound. Each of the learned models can be associated with a score. For example, upon the presentation of a sound signature, each GMM will produce a score. The scores can be evaluated against a threshold, and the GMM with the highest score can be identified as the detected target sound. For instance, if the learned model for the "alarm" sound produces the highest score (e.g., smallest distortion result) compared to other learned models, the ambient sound is classified as an "alarm" target sound.

The classification step **715** also takes into account likelihoods (e.g. recognition probabilities). For instance, as part of the step of comparing the sound signature of the unknown ambient sound against all the GMMs for the learned models, each GMM can produce a likelihood result, or output. As an example, these likelihood results can be evaluated against

each other or in the context in a logical context to determine the GMM considered "most likely" to match the sound signature of the target sound. The processor **206** can then select the GMM with the highest likelihood or score via soft decisions.

The earpiece **100** can continually monitor the environment for target sounds, or monitor the environment on a scheduled basis. In one arrangement, the earpiece **100** can increase monitoring in the presence of high ambient noise possibly signifying environmental danger or activity. Upon classifying an ambient sound as a target sound the processor **206** at step **716** can generate an alarm. As previously noted, the earpiece **100** can mix the target sound with audio content, amplify the target sound, reproduce the target sound, and/or deliver an audible message. As one example, spectral bands of the audio content that mask the target sound can be suppressed to increase an audibility of the target sound. This serves to notify the user of a target sounded detected in the environment, to which the user may not be aware depending on their environmental context.

As an example, the processor **206** can present an amplified audible notification to the user via the ECR **120**. The audible notification can be a synthetic voice identifying the target sound (e.g. "car alarm"), a location or direction of the sound source generating the target sound (e.g. "to your left"), a duration of the target sound (e.g., "3 minutes") from initial capture, and any other information (e.g., proximity, severity level, etc.) related to the target sound. Moreover, the processor **206** can selectively mix the target sound with the audio content based on a predetermined threshold level. For example, the user can prioritize target sound types for receiving various levels of notification, and/or identify the sound types as desirable of undesirable.

FIG. **8**, presents a pictorial diagram **800** for mixing ambient sounds and target sounds with audio content. In the illustration show, the earpiece **100** is playing music **136** to the ear canal via ECR **120** while simultaneously monitoring target sounds in the environment. At time, T, the processor **206** upon detecting a target sound (signature **135**) can lower the music volume from the media player **150** (graph **141**), and increase the volume of the ambient sound received at the ASM **110** (graph **142**). Other mixing arrangements are herein contemplated. In such regard, the user hears a smooth audio transition between the music and the target sound. Notably, the ramp up and down times can also be adjusted based on the priority of the target sound. For example, in an extreme case, the processor **206** can immediately shut off the music, and present the audible warning. Other various implementations for mixing audio and managing audio content delivery have been herein contemplated. Moreover, the audio content can be managed with other media devices (e.g., cell phone). For instance, upon detecting a target sound, the processor **206** can inform the user and the called party of a target sound. In such regard, the user does not need to inform the called party since they also receive the notification which can save them time to explain an emergency situation.

As one example, the processor **206** can spectrally enhance the audio content in view of the ambient sound. Moreover, a timbral balance of the audio content can be maintained by taking into account level dependent equal loudness curves and other psychoacoustic criteria (e.g., masking) associated with the personalized hearing level (PHL). For instance, auditory queues in a received audio content can be enhanced based on the PHL **430** and a spectrum of the ambient sound captured at the ASM **110**. Frequency peaks within the audio content can be elevated relative to ambient noise frequency levels and in accordance with the PHL to permit sufficient audibility of the ambient sound. The PHL reveals frequency dynamic ranges that can be used to limit the compression range of the peak elevation in view of the ambient noise spectrum.

In one arrangement, the processor **206** can compensate for a masking of the ambient sound by the audio content. Notably, the audio content if sufficiently loud, can mask auditory queues in the ambient sound, which can i) potentially cause hearing damage, and ii) prevent the user from hearing target sounds in the environment (e.g., an approaching ambulance, an alarm, etc.) Accordingly, the processor **206** can accentuate and attenuate frequencies of the audio content and ambient sound to permit maximal sound reproduction while simultaneously permitting audibility of ambient sounds. In one arrangement, the processor **206** can narrow noise frequency bands within the ambient sound to permit sensitivity to audio content between the frequency bands. The processor **206** can also determine if the ambient sound contains salient information (e.g., target sounds) that should be un-masked with respect to the audio content. If the ambient sound is not relevant, the processor **206** can mask the ambient sound (e.g., increase levels) with the audio content until target sounds are detected.

Note that in at least one exemplary embodiment the ASM is not part of an earpiece and is configured to measure the environment. Additionally in at least one exemplary embodiment the ECR is not part of an earpiece but can be a speaker that emits a notification signal. Note that at least one exemplary embodiment is an acoustic device (e.g., non-earpiece) that includes the ASM, optionally an ECR, and optionally ECM.

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 acoustic device, comprising:

an Ambient Sound Microphone (ASM) configured to capture ambient sound;

at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal; and

a processor operatively coupled to the ASM and the ECR, where the processor monitors the ambient sound for a target sound and adjusts the audio delivered to the ear canal based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.

2. The acoustic device of claim **1**, where the acoustic device is an earpiece, wherein the target sound is at least one among an alarm, a horn, and a noise.

3. The acoustic device of claim **1**, where the acoustic device is an earpiece, wherein the processor monitors the ambient sound for spoken words associated with verbal warnings.

4. The acoustic device of claim **1**, where the acoustic device is an earpiece, further comprising a memory to store, responsive to a directive by a user of the acoustic device, at least one target sound captured by the ASM for learning.

5. The acoustic device of claim 1, where the acoustic device is an earpiece, further comprising an audio interface operatively coupled to the processor configured to receive audio content from a media player or cell phone,

wherein the processor selectively adjusts a volume of the audio content delivered to the ear canal when the target sound is detected.

6. A method for personalized listening, the method comprising:

capturing ambient sound with an Ambient Sound Microphone (ASM);

monitoring the ambient sound for a target sound by generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound; and

adjusting a delivery of audio by an Ear Canal Receiver (ECR) in an earpiece to an ear canal based on the target sound.

7. The method of claim 6, further comprising: passing the target sound to the ECR for delivery to the ear canal.

8. The method of claim 6, further comprising: amplifying the target sound for delivery to the ear canal.

9. The method of claim 6, further comprising: attenuating the target sound for delivery to the ear canal.

10. The method of claim 6, further comprising: generating an audible message for delivery to the ear canal.

11. The method of claim 6, further comprising:

mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound.

12. The method of claim 6, further comprising: detecting and reporting from the sound signature a direction or a speed of a sound source generating the target sound.

13. The method of claim 6, further comprising: detecting and reporting from the sound signature a spoken utterance in the ambient sound associated with verbal warnings.

14. The method of claim 6, further comprising: identifying the target sound from the sound signature and transmitting a warning notification to other devices.

15. The method of claim 6, wherein the target sound is at least one among an alarm, a horn, a voice, and a noise.

16. A method for sound signature detection, the method comprising:

capturing ambient sound with an Ambient Sound Microphone (ASM); and

receiving a directive to learn a sound signature within the ambient sound,

where a voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.

17. The method of claim 16, further comprising: saving the sound signature locally on an earpiece or remotely to a server.

18. A method for personalized listening, the method comprising:

capturing ambient sound via an earpiece that is at least partially occluded in an ear canal;

detecting a sound signature within the ambient sound that is associated with a target sound; and

mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound and a personalized hearing level (PHL),

where learned models are retrieved from a database, the sound signature is compared to the learned models, and the target sound is identified from the learned models in view of the comparison.

19. The method of claim 18, further comprising: enhancing auditory queues in the target sound relative to the audio content based on a spectrum of the ambient sound captured at an ambient sounds microphone (ASM).

20. A sound detection device comprising:

an ambient sound microphone configured to measure an ambient sound;

an ear canal microphone; and

a processor configured to compare the ambient sound to at least one target sound signature, and where the processor identifies an onset of an identified target sound signature in the ambient sound,

where the ear canal microphone is configured to emit an auditory warning when the processor identifies the onset.

21. The sound detection device according to claim 20, where the ear canal microphone is operatively connected to an earpiece.

* * * * *

Exhibit D

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

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(54) **METHOD AND APPARATUS FOR IN-EAR CANAL SOUND SUPPRESSION**

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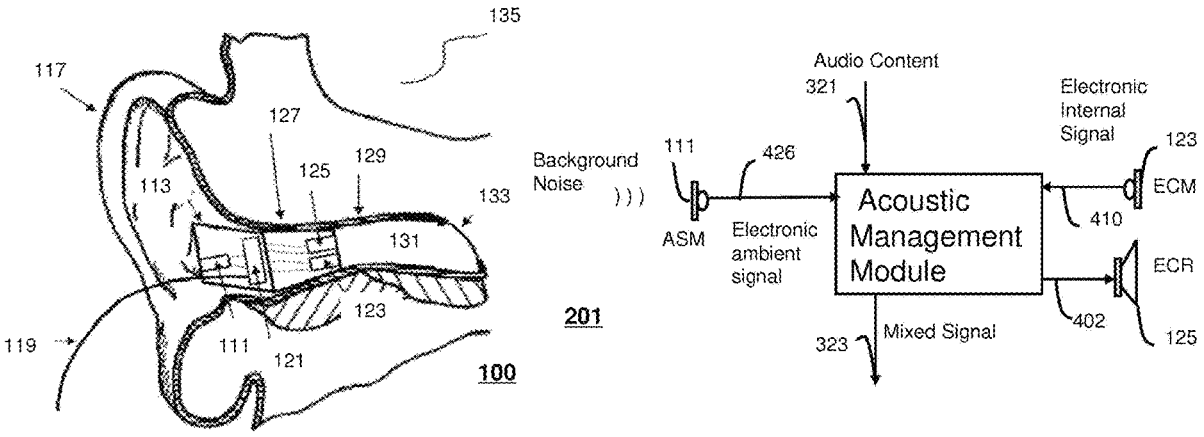
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(57) **ABSTRACT**

A method and system of conferencing can include the steps of initiating a conference call at a communication device with two or more communication devices and selecting to suppress a voice communication of at least one communication device on the conference call where a modified electronic signal is generated with the selected at least one communication device so that the voice communication from the selected at least one communication device is inaudible. The method or system further includes sending the modified electronic signal to at least one other communication device on the conference call. Other embodiments are disclosed.

15 Claims, 8 Drawing Sheets



Related U.S. Application Data

continuation of application No. 14/943,001, filed on Nov. 16, 2015, now Pat. No. 10,194,032, which is a continuation of application No. 12/245,316, filed on Oct. 3, 2008, now Pat. No. 9,191,740, which is a continuation-in-part of application No. 12/115,349, filed on May 5, 2008, now Pat. No. 8,081,780.

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See application file for complete search history.

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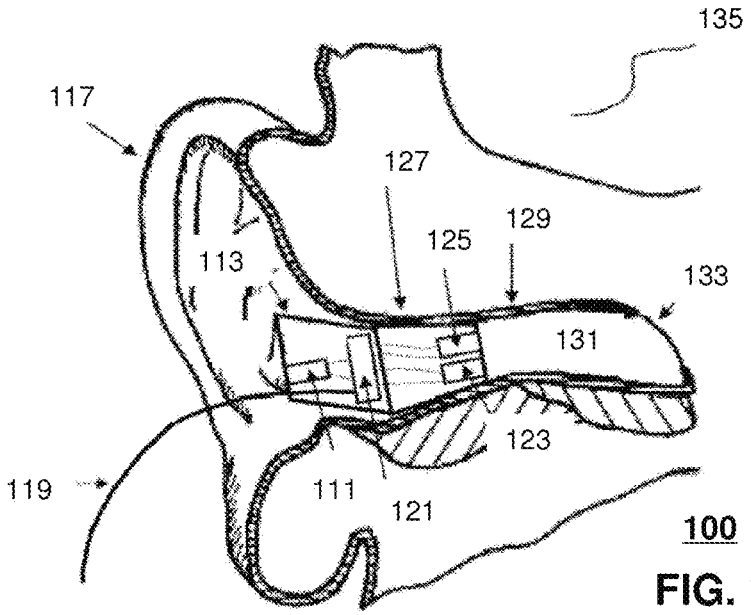

135
117
127
129
125
113
133
131
123
119
111
121
100

FIG. 1

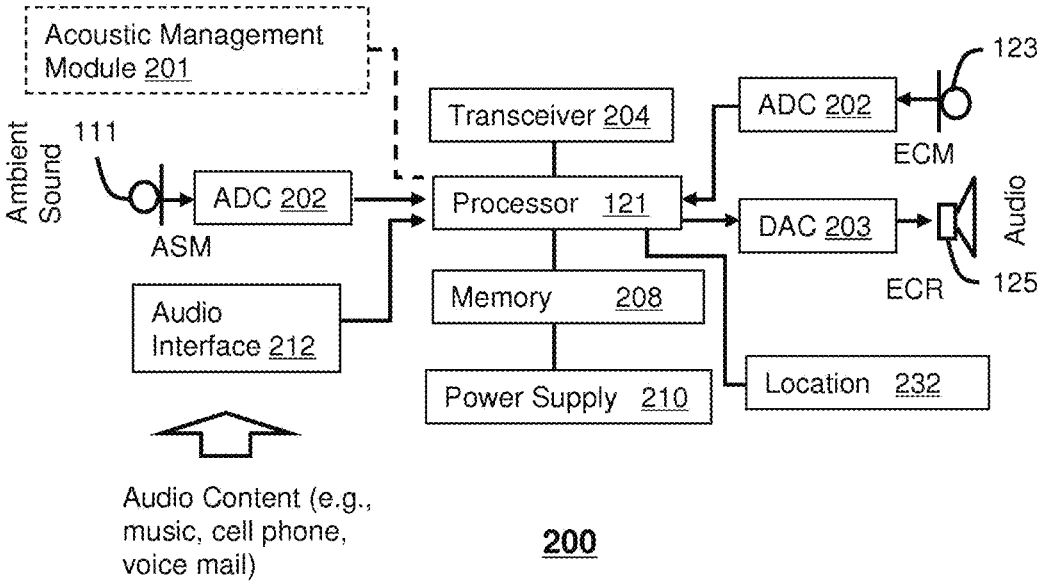

Acoustic Management Module 201
Ambient Sound
111
ASM
ADC 202
Transceiver 204
Processor 121
Memory 208
Power Supply 210
Audio Interface 212
Audio Content (e.g., music, cell phone, voice mail)
ADC 202
123
ECM
DAC 203
Audio
ECR
125
Location 232
200

FIG. 2

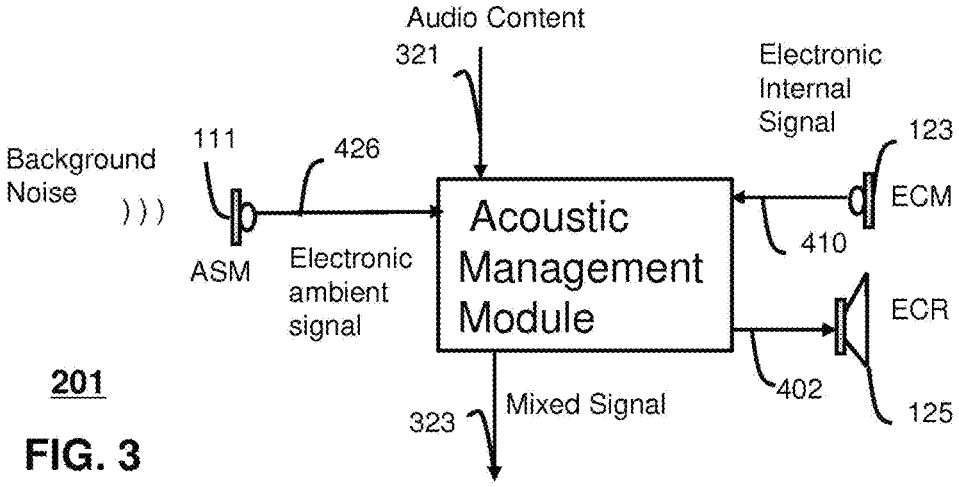

Audio Content
321
Electronic Internal Signal
123
Background Noise
111
426
ECM
Acoustic Management Module
410
ASM
Electronic ambient signal
ECR
402
125
201
323
Mixed Signal

FIG. 3

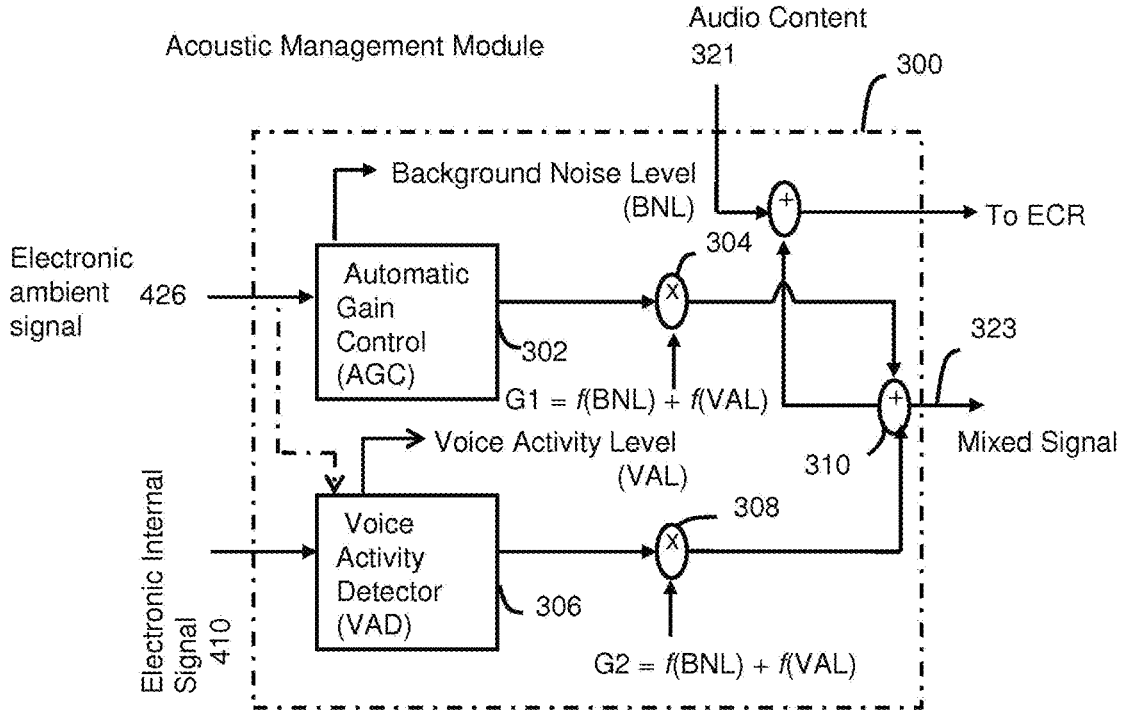

Acoustic Management Module
Audio Content
321
300
Background Noise Level (BNL)
To ECR
304
Electronic ambient signal
426
Automatic Gain Control (AGC)
302
323
G1 = f(BNL) + f(VAL)
Voice Activity Level (VAL)
310
Mixed Signal
308
Electronic Internal Signal
410
Voice Activity Detector (VAD)
306
G2 = f(BNL) + f(VAL)

FIG. 4

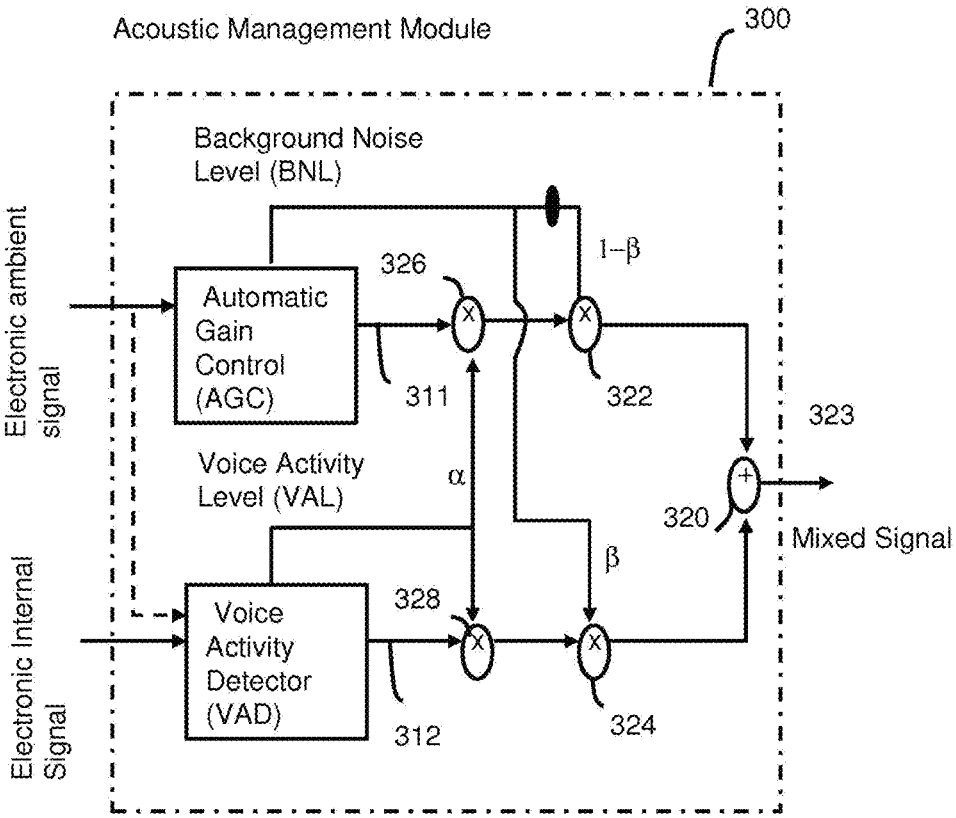

Acoustic Management Module
300
Background Noise Level (BNL)
Electronic ambient signal
Automatic Gain Control (AGC)
326
311
1−β
322
323
Voice Activity Level (VAL)
α
320
Mixed Signal
β
Electronic Internal Signal
Voice Activity Detector (VAD)
328
312
324

FIG. 5

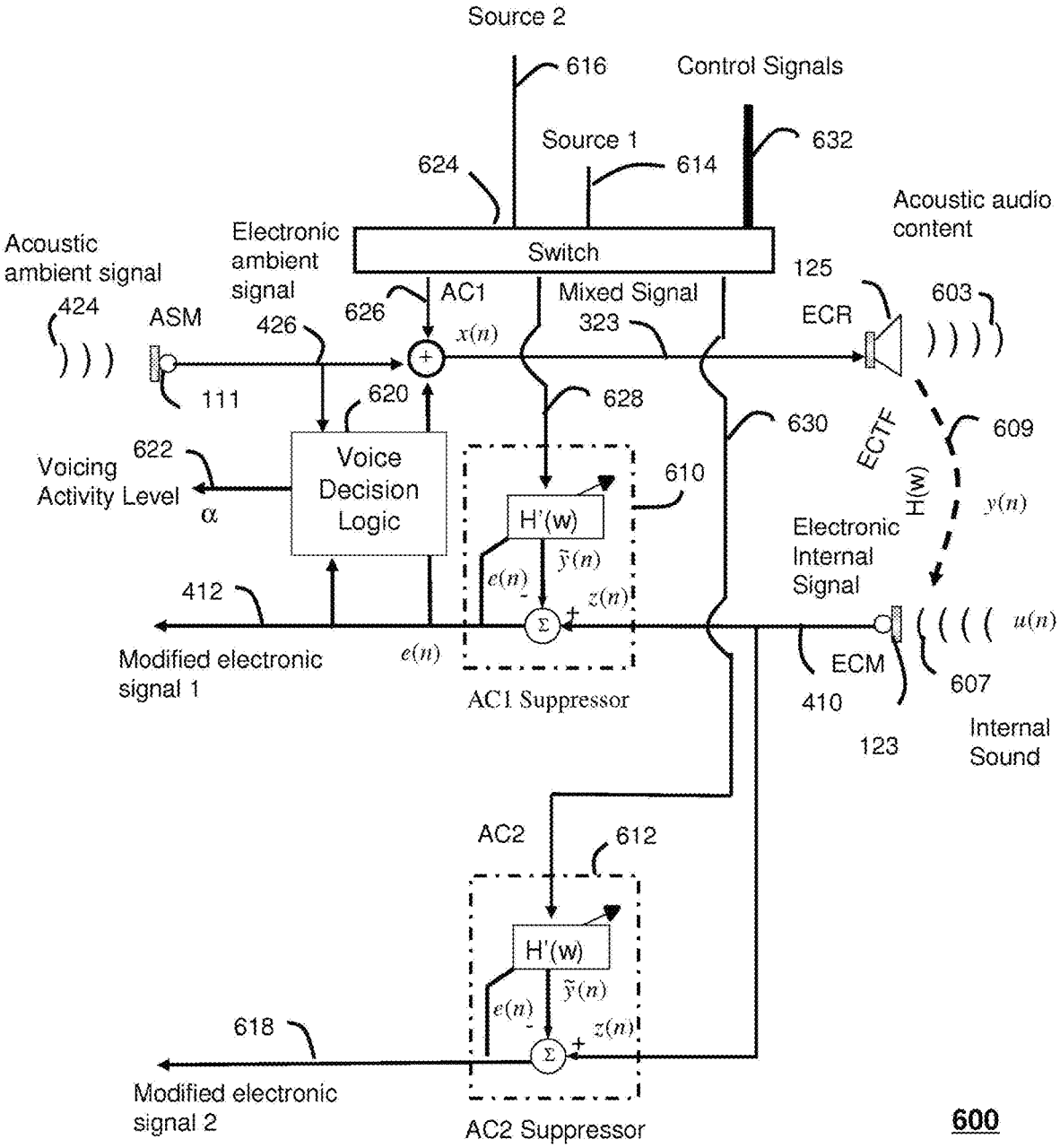

Source 2
616
Control Signals
Source 1
614
632
624
Acoustic audio content
Acoustic ambient signal
Electronic ambient signal
Switch
125
AC1
Mixed Signal
603
424
ASM
426
626
x(n)
323
ECR
111
620
628
630
609
622
Voice Decision Logic
610
ECTF
H(w)
y(n)
Voicing Activity Level
α
H'(w)
Electronic Internal Signal
412
e(n)
ỹ(n)
z(n)
u(n)
Modified electronic signal 1
e(n)
ECM
607
AC1 Suppressor
410
123
Internal Sound
AC2
612
H'(w)
618
Modified electronic signal 2
AC2 Suppressor
600

FIG. 6

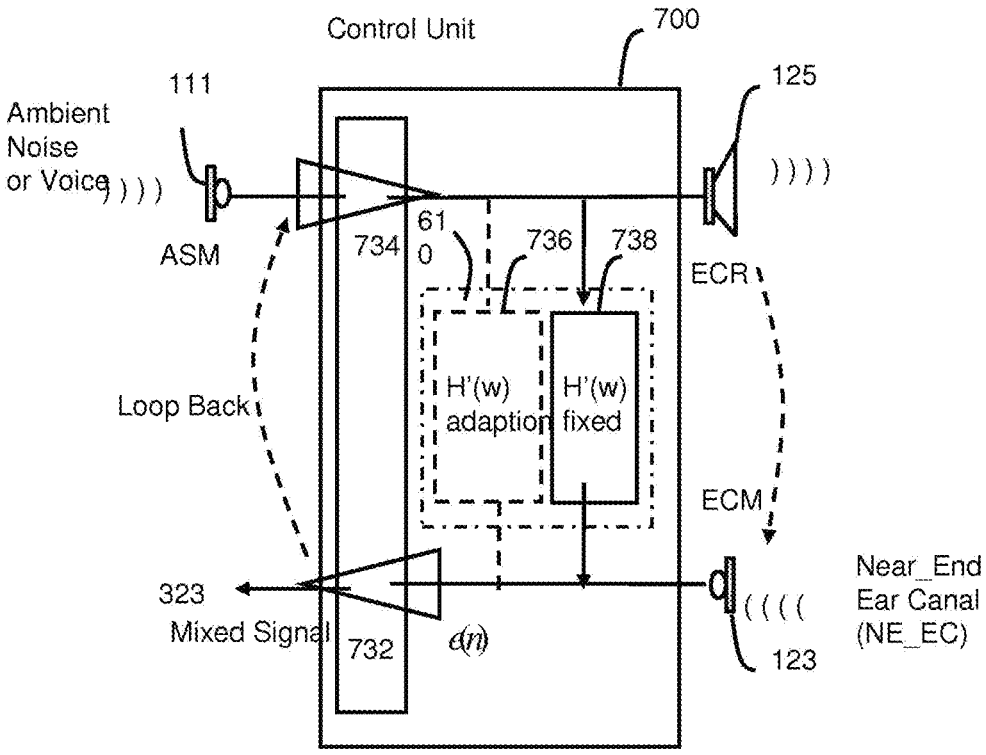

Control Unit
700
111
Ambient
Noise
or Voice
ASM
125
ECR
734
61
0
736
738
H'(w)
adaption
H'(w)
fixed
Loop Back
ECM
323
Mixed Signal
732
d(n)
Near_End
Ear Canal
(NE_EC)
123

FIG. 7

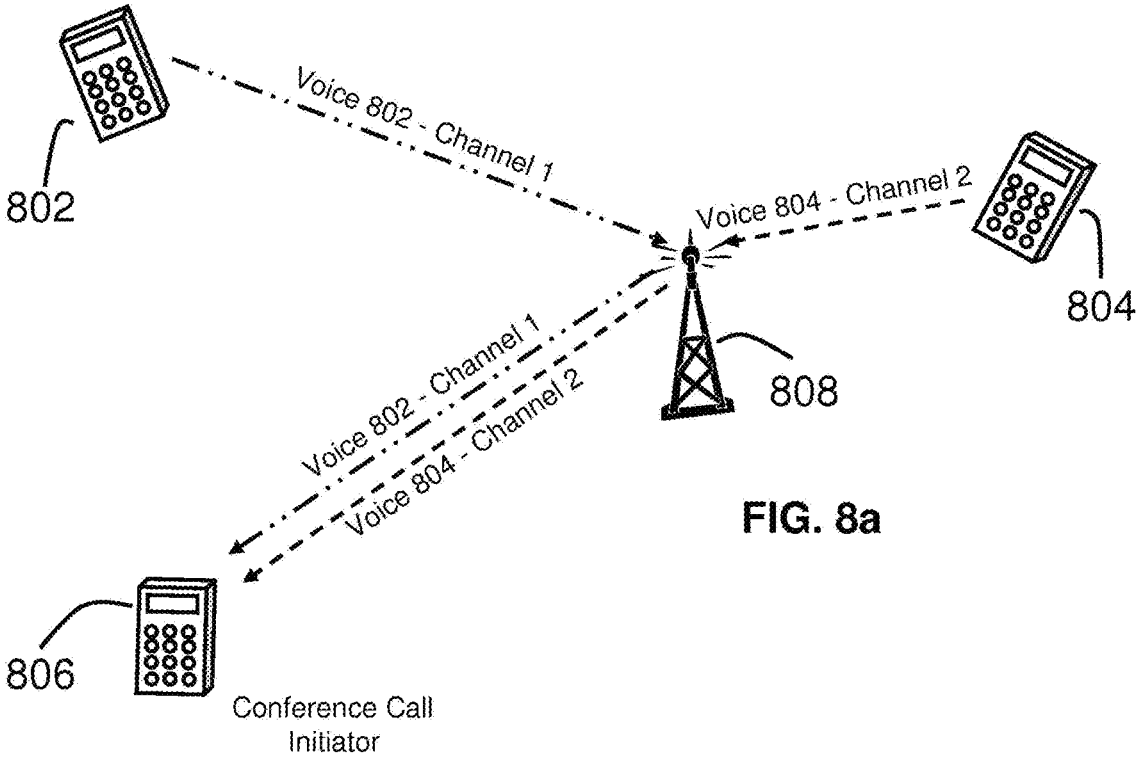

Voice 802 - Channel 1
Voice 804 - Channel 2
802
804
808
Voice 802 - Channel 1
Voice 804 - Channel 2
806
Conference Call Initiator

FIG. 8a

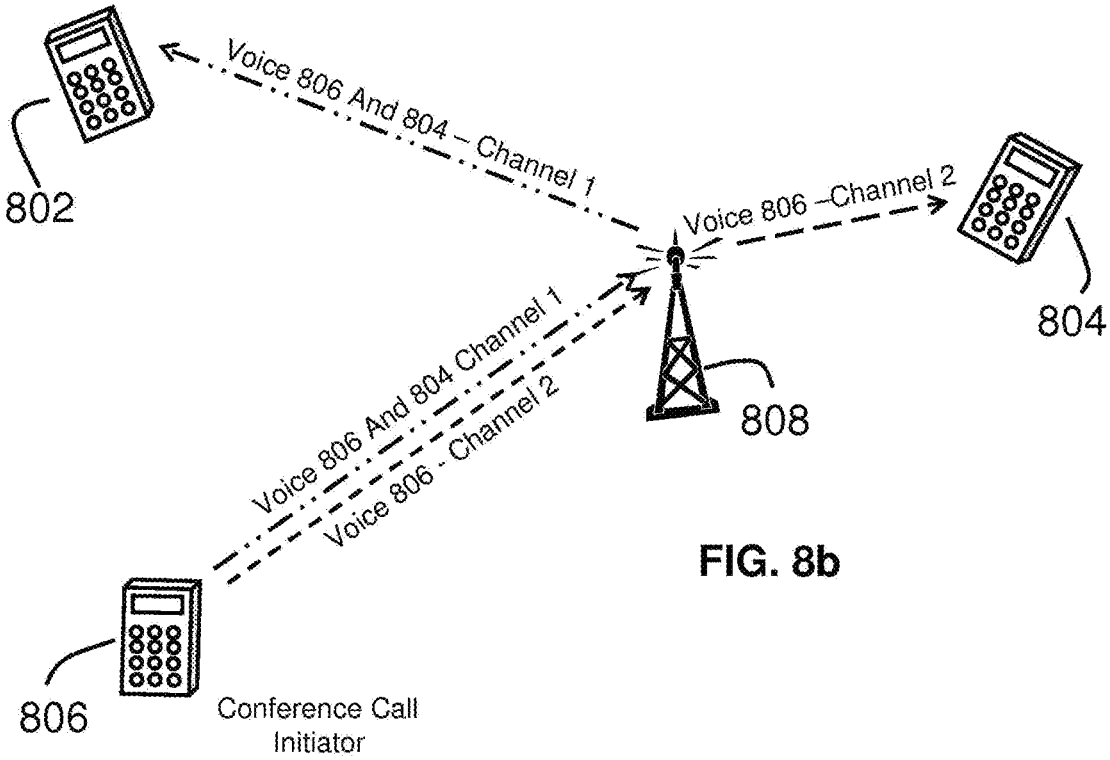

Voice 806 And 804 – Channel 1
Voice 806 –Channel 2
802
804
808
Voice 806 And 804 Channel 1
Voice 806 - Channel 2
806
Conference Call Initiator

FIG. 8b

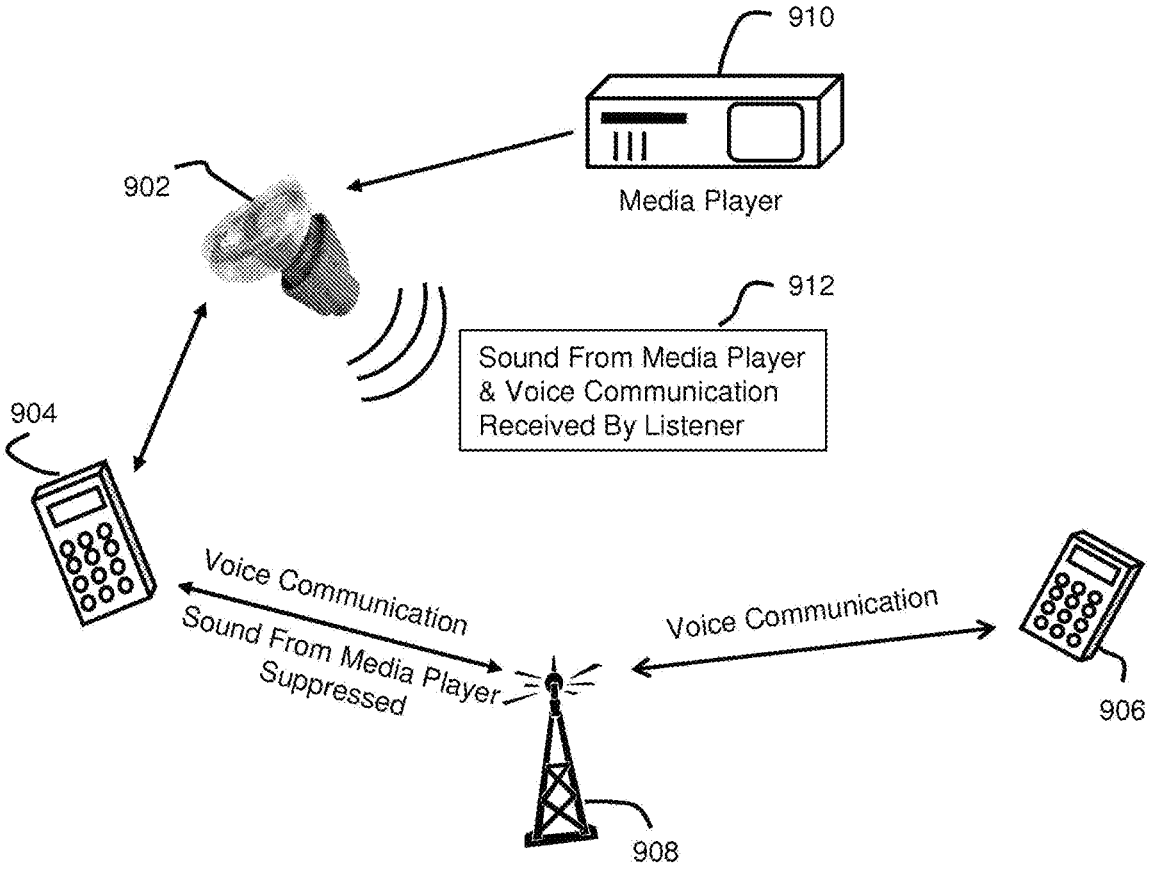

910
Media Player
902
912
Sound From Media Player
& Voice Communication
Received By Listener
904
Voice Communication
Sound From Media Player
Suppressed
Voice Communication
906
908

FIG. 9

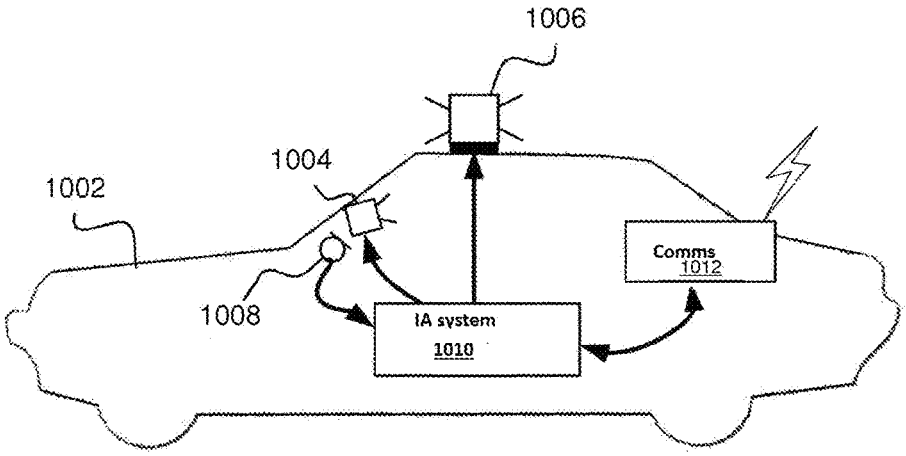

1006
1004
1002
1008
IA system
1010
Comms
1012

FIG. 10

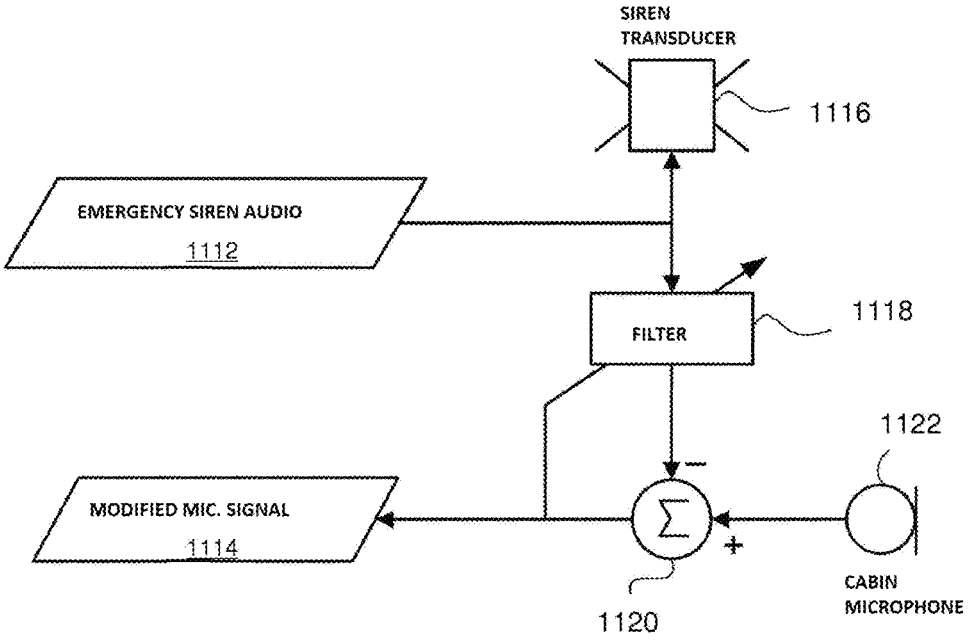

SIREN
TRANSDUCER
1116
EMERGENCY SIREN AUDIO
1112
FILTER
1118
1122
MODIFIED MIC. SIGNAL
1114
1120
CABIN
MICROPHONE

FIG. 11

METHOD AND APPARATUS FOR IN-EAR CANAL SOUND SUPPRESSION

CROSS REFERENCE TO RELATED APPLICATIONS

This application is a Continuation of U.S. patent application Ser. No. 16/258,015, which is a Continuation of U.S. patent application Ser. No. 14/943,001, filed on Nov. 16, 2015, which is a Continuation of U.S. patent application Ser. No. 12/245,316, filed on Oct. 3, 2008, now U.S. Pat. No. 9,191,740, which is a Continuation-In-Part of U.S. patent application Ser. No. 12/115,349, filed on May 5, 2008, now U.S. Pat. No. 8,081,780, which claims the priority benefit of U.S. Provisional Application Ser. No. 60/916,271, filed on May 4, 2007, the entire disclosure of which is incorporated herein by reference. This application is also related to U.S. patent application Ser. No. 12/110,773, filed on Apr. 28, 2008, now U.S. Pat. No. 8,577,062, which claims the priority benefit of U.S. Provisional Application Ser. No. 60/914,318, filed Apr. 27, 2007, the entire disclosure of which is incorporated herein by reference. This application is also related to U.S. patent application Ser. No. 12/170,171, filed on Jul. 9, 2008, now U.S. Pat. No. 8,526,645, the entire disclosure of which is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention pertains to sound reproduction, sound recording, audio communications and hearing protection using earphone devices designed to provide variable acoustical isolation from ambient sounds while being able to audition both environmental and desired audio stimuli. Particularly, the present invention describes a method and device for suppressing echo in an ear-canal when capturing a user's voice when using an ambient sound microphone and an ear canal microphone.

BACKGROUND OF THE INVENTION

People use headsets or earpieces primarily for voice communications and music listening enjoyment. A headset or earpiece generally includes a microphone and a speaker for allowing the user to speak and listen. An ambient sound microphone mounted on the earpiece can capture ambient sounds in the environment; sounds that can include the user's voice. An ear canal microphone mounted internally on the earpiece can capture voice within the ear canal; sounds generated when the user is speaking.

An earpiece that provides sufficient occlusion can utilize both the ambient sound microphone and the ear canal microphone to enhance the user's voice. An ear canal receiver mounted internal to the ear canal can loopback sound captured at the ambient sound microphone or the ear canal microphone to allow the user to listen to captured sound. If the earpiece is however not properly sealed within the ear canal, the ambient sounds can leak through into the ear canal and create an echo feedback condition with the ear canal microphone and ear canal receiver. In such cases, the feedback loop can generate an annoying "howling" sound that degrades the quality of the voice communication and listening experience.

SUMMARY

In a first embodiment, a method of listening to music or other media content during a full duplex communication event, the method comprising the steps of delivering audio content to an ear canal of a first user by way of an Ear Canal Receiver (ECR) to produce an acoustic audio content where the audio content includes music or other media content, capturing in the ear canal of the first user by way of an Ear Canal Microphone (ECM) an electronic signal comprising the acoustic audio content and a spoken voice of the first user in the presence of the audio content delivered to the ear canal, suppressing the audio content in the electronic signal while preserving the spoken voice to produce a modified electronic signal, and sending the modified electronic signal to at least one other user so that the audio content is sufficiently inaudible and the spoken voice is audible during the full duplex communication.

In a second embodiment, a method of conferencing, the method comprising the steps of initiating a conference call with two or more people, selecting to suppress the voice communication of at least one person on the conference call where a modified electronic signal is generated with the selected at least one person voice communication being inaudible, and sending the modified electronic signal to at least one other person on the conference call.

In a third embodiment, a method of listening to audio content comprising the steps of listening to audio content from a transducer coupled to a communication device, engaging in a full duplex voice communication with the communication device where the voice communication and the audio content is output by the transducer, and suppressing the audio content from a transmitted signal from the communication device such that participants receiving the transmitted signal hear a spoken voice of a user of the communication device but the audio content is inaudible.

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 block diagram for an acoustic management module in accordance with an exemplary embodiment;

FIG. **4** is a schematic for the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal as a function of a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **5** is a more detailed schematic of the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **6** is a block diagram of a system for generating modified electronic signals in accordance with an exemplary embodiment;

FIG. **7** is a schematic of a control unit for controlling adaptation of a first set and second set of filter coefficients of an echo suppressor for in-ear canal echo suppression in accordance with an exemplary embodiment;

FIGS. **8***a*-**8***b* are diagrams illustrating a conference call initiated by a user of a communication device in accordance with an exemplary embodiment;

FIG. **9** illustrates that a user can be listening to audio content while in voice communication with a remote communication device in accordance with an exemplary embodiment;

FIG. **10** is an illustration of a system for suppressing a siren or acoustic signal from a communication signal in accordance with an exemplary embodiment; and

FIG. **11** is a block diagram of the invisible audio system of FIG. **10** for suppressing the siren or acoustic signal in accordance with an exemplary embodiment.

DETAILED DESCRIPTION OF THE DRAWINGS

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.

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.

Various embodiments herein provide a method and device for automatically mixing audio signals produced by a pair of microphone signals that monitor a first ambient sound field and a second ear canal sound field, to create a third new mixed signal. An Ambient Sound Microphone (ASM) and an Ear Canal Microphone (ECM) can be housed in an earpiece that forms a seal in the ear of a user. The third mixed signal can be auditioned by the user with an Ear Canal Receiver (ECR) mounted in the earpiece, which creates a sound pressure in the occluded ear canal of the user. A voice activity detector can determine when the user is speaking and control an echo suppressor to suppress associated feedback in the ECR.

When the user engages in a voice communication, the echo suppressor can suppress feedback of the spoken voice from the ECR. The echo suppressor can contain two sets of filter coefficients; a first set that adapts when voice is not present and becomes fixed when voice is present, and a second set that adapts when the first set is fixed. The voice activity detector can discriminate between audible content, such as music, that the user is listening to, and spoken voice generated by the user when engaged in voice communication. The third mixed signal contains primarily the spoken voice captured at the ASM and ECM without echo, and can be transmitted to a remote voice communications system, such as a mobile phone, personal media player, recording device, walkie-talkie radio, etc. Before the ASM and ECM signals are mixed, they can be echo suppressed and subjected to different filters and at optional additional gains. This permits a single earpiece to provide full-duplex voice communication with proper or improper acoustic sealing.

The characteristic responses of the ASM and ECM filters can differ based on characteristics of the background noise and the voice activity level. In some exemplary embodiments, the filter response can depend on the measured Background Noise Level (BNL). A gain of a filtered ASM and a filtered ECM signal can also depend on the BNL. The (BNL) can be calculated using either or both the conditioned ASM and/or ECM signal(s). The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and can be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level (i.e. the high and low frequencies are attenuated before the level of the microphone signals are calculated).

At least one exemplary embodiment of the invention is directed to an earpiece for voice operated control. Reference is made to FIG. **1** in which an earpiece device, generally indicated as earpiece **100**, is constructed and operates 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 **131**. 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 **131** 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 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 frequency 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 **131**. This seal is also a basis for a 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 or partially 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 the earpiece **100**. In one arrangement, the ASM **111** can be housed in the assembly **113** to monitor 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**.

The earpiece **100** can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece **100** in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

The earpiece **100** can measure ambient sounds in the environment 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.

The earpiece **100** can generate an Ear Canal Transfer Function (ECTF) to model the ear canal **131** using ECR **125** and ECM **123**, as well as an Outer Ear Canal Transfer function (OETF) using ASM **111**. For instance, the ECR **125** can deliver an impulse within the ear canal and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal. The earpiece **100** can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece **100** to safely administer and monitor sound exposure to the ear.

Referring to FIG. **2**, a block diagram **200** of the earpiece **100** in accordance with an exemplary embodiment is shown. As illustrated, the earpiece **100** can include the processor **121** 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 **121** 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 processor **121** can also include a clock to record a time stamp.

As illustrated, the earpiece **100** can include an acoustic management module **201** to mix sounds captured at the ASM **111** and ECM **123** to produce a mixed signal. The processor **121** can then provide the mixed signal to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor or communication device. The acoustic management module **201** can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the functionality of the acoustic management module **201** can be provided by way of software, such as program code, assembly language, or machine language.

The memory **208** can also store program instructions for execution on the processor **121** as well as captured audio processing data and filter coefficient data. The memory **208** can be off-chip and external to the processor **121**, and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor **121**. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor **121** to provide high speed data access. The storage memory **208** can be non-volatile memory such as SRAM to store captured or compressed audio data.

The earpiece **100** can include an audio interface **212** operatively coupled to the processor **121** and acoustic management module **201** to receive audio content, for example from a media player, cell phone, or any other communication device, and deliver the audio content to the processor **121**. The processor **121** responsive to detecting spoken voice from the acoustic management module **201** can adjust the audio content delivered to the ear canal. For instance, the processor **121** (or acoustic management module **201**) can lower a volume of the audio content responsive to detecting a spoken voice. The processor **121** by way of the ECM **123** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the acoustic management module **201**.

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 location receiver **232** can utilize common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece **100**.

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 **121** 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 block diagram of the acoustic management module **201** in accordance with an exemplary embodiment. Briefly, the acoustic management module **201** facilitates monitoring, recording and transmission of user-generated voice (speech) to a voice communication system. User-generated sound is detected with the ASM **111** that monitors a sound field near the entrance to a user's ear, and with the ECM **123** that monitors a sound field in the user's occluded ear canal. A new mixed signal **323** is created by filtering and mixing the ASM and ECM microphone signals. The filtering and mixing process is automatically controlled depending on the background noise level of the ambient sound field to enhance intelligibility of the new mixed signal **323**. For instance, when the background noise level is high, the acoustic management module **201** automatically increases the level of the ECM **123** signal relative to the level of the ASM **111** to create the new mixed signal **323**. When the background noise level is low, the acoustic management module **201** automatically decreases the level of the ECM **123** signal relative to the level of the ASM **111** to create the new mixed signal **323**.

As illustrated, the ASM **111** is configured to capture ambient sound and produce an electronic ambient signal **426**, the ECR **125** is configured to pass, process, or play acoustic audio content **402** (e.g., audio content **321**, mixed signal **323**) to the ear canal, and the ECM **123** is configured to capture internal sound in the ear canal and produce an

electronic internal signal **410**. The acoustic management module **201** is configured to measure a background noise signal from the electronic ambient signal **426** or the electronic internal signal **410**, and mix the electronic ambient signal **426** with the electronic internal signal **410** in a ratio dependent on the background noise signal to produce the mixed signal **323**. The acoustic management module **201** filters the electronic ambient signal **426** and the electronic internal **410** signal based on a characteristic of the background noise signal using filter coefficients stored in memory or filter coefficients generated algorithmically.

In practice, the acoustic management module **201** mixes sounds captured at the ASM **111** and the ECM **123** to produce the mixed signal **323** based on characteristics of the background noise in the environment and a voice activity level. The characteristics can be a background noise level, a spectral profile, or an envelope fluctuation. The acoustic management module **201** manages echo feedback conditions affecting the voice activity level when the ASM **111**, the ECM **123**, and the ECR **125** are used together in a single earpiece for full-duplex communication, when the user is speaking to generate spoken voice (captured by the ASM **111** and ECM **123**) and simultaneously listening to audio content (delivered by ECR **125**).

In noisy ambient environments, the voice captured at the ASM **111** includes the background noise from the environment, whereas, the internal voice created in the ear canal **131** captured by the ECM **123** has less noise artifacts, since the noise is blocked due to the occlusion of the earpiece **100** in the ear. It should be noted that the background noise can enter the ear canal if the earpiece **100** is not completely sealed. In this case, when speaking, the user's voice can leak through and cause an echo feedback condition that the acoustic management module **201** mitigates.

FIG. **4** is a schematic **300** of the acoustic management module **201** illustrating a mixing of the electronic ambient signal **426** with the electronic internal signal **410** as a function of a background noise level (BNL) and a voice activity level (VAL) in accordance with an exemplary embodiment. As illustrated, the acoustic management module **201** includes an Automatic Gain Control (AGC) **302** to measure background noise characteristics. The acoustic management module **201** also includes a Voice Activity Detector (VAD) **306**. The VAD **306** can analyze either or both the electronic ambient signal **426** and the electronic internal signal **410** to estimate the VAL. As an example, the VAL can be a numeric range such as 0 to 10 indicating a degree of voicing. For instance, a voiced signal can be predominately periodic due to the periodic vibrations of the vocal cords. A highly voiced signal (e.g., vowel) can be associated with a high level, and a non-voiced signal (e.g., fricative, plosive, consonant) can be associated with a lower level.

The acoustic management module **201** includes a first gain (G1) **304** applied to the AGC processed electronic ambient signal **426**. A second gain (G2) **308** is applied to the VAD processed electronic internal signal **410**. The acoustic management module **201** applies the first gain (G1) **304** and the second gain (G2) **308** as a function of the background noise level and the voice activity level to produce the mixed signal **323**, where

$$G1=f(\mathrm{BNL})+f(\mathrm{VAL}) \text{ and } G2=f(\mathrm{BNL})+f(\mathrm{VAL})$$

As illustrated, the mixed signal is the sum **310** of the G1 scaled electronic ambient signal and the G2 scaled electronic internal signal. The mixed signal **323** can then be transmitted to a second communication device (e.g. second cell phone, voice recorder, etc.) to receive the enhanced voice signal. The acoustic management module **201** can also play the mixed signal **323** back to the ECR for loopback listening. The loopback allows the user to hear himself or herself when speaking, as though the earpiece **100** and associated occlusion effect were absent. The loopback can also be mixed with the audio content **321** based on the background noise level, the VAL, and audio content level. The acoustic management module **201** can also account for an acoustic attenuation level of the earpiece, and account for the audio content level reproduced by the ECR when measuring background noise characteristics. Echo conditions created as a result of the loopback can be mitigated to ensure that the voice activity level is accurate.

FIG. **5** is a more detailed schematic of the acoustic management module **201** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment. In particular, the gain blocks for G1 and G2 of FIG. **4** are a function of the BNL and the VAL and are shown in greater detail. As illustrated, the AGC produces a BNL that can be used to set a first gain **322** for the processed electronic ambient signal **311** and a second gain **324** for the processed electronic internal signal **312**. For instance, when the BNL is low (<70 dBA), gain **322** is set higher relative to gain **324** so as to amplify the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. When the BNL is high (>85 dBA), gain **322** is set lower relative to gain **324** so as to attenuate the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. The mixing can be performed in accordance with the relation:

Mixed signal=(1−β)*electronic ambient signal+(β)*electronic internal signal where=(1−β) is an external gain, (β) is an internal gain, and the mixing is performed with $0<\beta<1$.

As illustrated, the VAD produces a VAL that can be used to set a third gain **326** for the processed electronic ambient signal **311** and a fourth gain **328** for the processed electronic internal signal **312**. For instance, when the VAL is low (e.g., 0-3), gain **326** and gain **328** are set low so as to attenuate the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is not detected. When the VAL is high (e.g., 7-10), gain **326** and gain **328** are set high so as to amplify the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is detected.

The gain scaled processed electronic ambient signal **311** and the gain scaled processed electronic internal signal **312** are then summed at adder **320** to produce the mixed signal **323**. The mixed signal **323**, as indicated previously, can be transmitted to another communication device, or as loopback to allow the user to hear his or her self.

FIG. **6** is an exemplary schematic of an operational unit **600** of the acoustic management module for generating modified electronic signals in accordance with an embodiment. The operational unit **600** may contain more or less than the number of components shown in the schematic. The operational unit **600** can include an adaptive filter **610**, an adaptive filter **612**, and a voice decision logic **620**.

Adaptive filters **610** and **612** can be a Least Mean Squares (LMS) or Normalized Least Mean Squares (NLMS) adaptive filter that models an ear canal transfer function (ECTF) between the ECR **125** and the ECM **123**. The adaptive filter **610** generates the modified electronic signal, e(n) **412**, which is provided as an input to the voice decision logic **620**; e(n) is also termed the error signal e(n) of the adaptive filter **610**. In an echo cancellation mode, the error signal e(n) **412** is used to update the filter H(w) to model the ECTF of an

echo path. The error signal e(n) **412** closely approximates the user's spoken voice signal u(n) **607** when the echo suppressor **610** accurately models the ECTF.

Alternately, a first reference signal can be provided to adaptive filter **610** and a second reference signal can be provided to adaptive filter **612**. For example, a reference signal can be music, media content, or a voice communication signal. The error signal e(n) **412** for adaptive filter **610** or e(n) **618** for adaptive filter **612** is used to update their respective filters H(w) to model the ECTF in the signal path. The error signal e(n) **412** for adaptive filter **610** or e(n) **618** for adaptive filter **612** will closely approximate the corresponding acoustic reference signal as output by ECR **125** and received ECM **123** when adaptive filters **610** and **612** accurately model the ECTF.

In the configuration shown the adaptive filter **610** (and similarly adaptive filter **612**) minimizes the error between the filtered signal, $\tilde{y}(n)$, and the electronic internal signal, z(n), in an effort to obtain a transfer function H' which is a best approximation to the H(w) (i.e., ECTF). H(w) represents the transfer function of the ear canal and models the echo response. (z(n)=u(n)+y(n)+v(n), where u(n) is the spoken voice or the reference signal that is an internal sound **607**, y(n) is the acoustic signal output by ECR **125**, and v(n) is background noise (if present, for instance due to improper sealing)).

In the echo cancellation mode, the adaptive filter **610** monitors the mixed signal **323** delivered to the ECR **125** and produces an echo estimate $\tilde{y}(n)$ of an echo y(n) **609** based on the captured electronic internal signal **410** and the mixed signal **323**. The adaptive filter **610**, upon learning the ECTF by an adaptive process, can then suppress the echo y(n) **609** of the acoustic audio content **603** (e.g., output mixed signal **323**) in the electronic internal signal z(n) **410**. It subtracts the echo estimate $\tilde{y}(n)$ from the electronic internal signal **410** to produce the modified electronic internal signal e(n) **412**.

It should be noted that more than two adaptive filters could be used to generate multiple modified electronic signals for using with the earpiece or other devices coupled or paired to the earpiece. The circuitry and process disclosed herein is not limited to an earpiece and can be practiced in other communication devices such as a cell phone, smart phone, PDA, laptop computer, radio communication systems, and conferencing systems to name but a few. In a reference suppression mode, a signal or multiple signals are suppressed from at least one device.

Referring to FIG. **8***a*, a conference call is initiated by the user of a communication device **806** in accordance with an exemplary embodiment. Communication device **806** is coupled via a network **808** to communication devices **802** and **804**. In at least one exemplary embodiment, communication device **806** opens a channel **1** to communicate with device **802** and a channel **2** to communicate with device **804**. Thus, device **806** receives voice communication from both devices **802** and **804**.

The reference suppression mode allows the initiator of the conference call to selectively suppress the voice communication of one or more participants and to selectively determine which of the participants each person gets to hear. As shown, two participants are engaged in the call. In at least one exemplary embodiment, the conference call initiator pre-selects which caller is suppressed. Referring to FIG. **8***b*, the conference call initiator has selected that the user of device **804** will not hear the voice of the user of device **802**. Thus, the voice communication signal from device **802** is the reference signal for suppression. For example, the users of devices **804** and **806** are in discussion of a business deal. The conference call initiator wants to better understand the implications of the business terms as the dialogue progresses. The user of device **802** is a business associate of the conference call initiator and provides input or comment that can be relevant to the negotiation. The voice communication of the user of device **802** is suppressed by communication device **806** such that only the initiator of the conference call is heard by the user of device **804**. The suppressed communication signal is sent to the user of device **804** through channel **2**. Conversely, the communication signal sent on channel **1** is not modified allowing the business associate to hear the full conversation while commenting to the conference call initiator.

Referring to FIG. **9**, a user can be listening to audio content while in voice communication with a user of device **906** in accordance with an exemplary embodiment. The user of communication device **906** does not hear the audio content in the voice communication. An earpiece **902** is coupled for receiving audio content from a media player **910**. As shown, earpiece **902** receives the media content directly from media player **910** via a wired or wireless connection. Alternately, earpiece **902** can receive the audio content through communication device **904**. The audio content can be music, media content, or other audio content that the user of earpiece **902** wants to hear.

Communication device **904** is paired with earpiece **902**. In at least one exemplary embodiment, the audio content provided by media player **910** is uninterrupted by a call from the user of communication device **906**. In at least one exemplary embodiment, the volume of the audio content can be automatically adjusted to a predetermined level (defined by the user) that would allow voice communication with the user of device **906** via network **908**. For example, the volume could be lowered for the user to hear that a call is incoming and further adjust the volume if the call is taken. In the reference suppression mode **912**, the audio content is the reference signal. Earpiece **902** suppresses the audio content from a communication signal being sent to the user of device **906**. Alternately, device **904** can suppress the audio content from the transmitted communication signal if it has access to the reference signal using the method described hereinbelow. Thus, the user of device **904** can continue to listen to a ball game, hear a news bulletin, or enjoy music while in communication with others and the audio content is not transmitted to those in communication with device **904**.

Referring back to FIG. **6**, a switch **624** includes an input **614** coupled to a source 1, an input **616** coupled to a source 2, an output **626**, an output **628**, an output **630** and control signals **632**. Source 1 and source 2 are signals such as music, media content, voice communication, or other audio content. Control signals configure switch **624** for providing source 1, source 2, or a combination of source 1 and source 2 at each output **626**, **628**, and **630**.

In the first example disclosed above, a user of the system initiates a conference call with two other people. Source 1 is a voice communication from a first caller. Source 2 is a voice communication from a second caller. The conference call initiator sets up the call on the system so that the first caller does not hear the voice of the second caller. The system configures switch **624** such that the first and second caller voice communication is mixed with an electronic ambient signal **426** and is provided as mixed signal **323** to ECR **125**. ECR **125** provides the acoustic audio content **603** that includes the voices of the first and second callers. Thus, the call initiator hears both the callers. Switch **624** is configured such that output **628** is coupled to source 2 which is the second caller voice communication. The second caller voice

communication is the reference signal for adaptive filter **610**. The adaptive filter **610**, upon learning the ECTF by an adaptive process can suppress the second caller voice of the acoustic audio content **603** (e.g., output mixed signal **323**) in the electronic internal signal **410** (z(n)). It subtracts the second caller estimate ỹ(n) from the electronic internal signal **410** to produce the modified electronic internal signal e(n) **412**. Thus, the modified electronic signal 1 will include the voice of the initiator of the conference call but not the second caller. The modified electronic signal 1 is transmitted to the first caller.

The system is set up for the second caller to hear the entire conversation. Output **630** of switch **624** couples to the adaptive filter **612**. No reference signal is provided to adaptive filter **612** so nothing in electronic internal signal **410** is suppressed. Alternately, adaptive filter **612** can be disabled. Modified electronic signal 2 is electronic internal signal **410** that includes all of the voice communication. The modified electronic signal 2 is transmitted to the second caller.

In the second example disclosed above, a user is listening to audio content such as music when receiving a call. The user can listen to the audio content while taking the call without the caller hearing the audio content. A second modified electronic signal 2 is not required so adaptive filter **612** is disabled. Source 1 couples to a media player or other audio source. For illustrative purposes, source 1 is music from a media player. Source 2 is the voice communication signal from a remote caller. Control signals **632** enable switch **624** for providing source 1 and source 2 at the output **626**. The music signal and the voice communication signal at output **626** are mixed with the electronic ambient signal **426** forming mixed signal **323**. Mixed signal **323** is provided to ECR **125**. ECR **125** outputs acoustic audio content **603** which includes the music and the voice communication from the remote caller.

Switch **624** is configured such that output **628** is coupled to source 1 which is the music signal. The music signal is the reference signal for adaptive filter **610**. The adaptive filter **610**, upon learning the ECTF by an adaptive process can suppress the music signal of the acoustic audio content **603** (e.g., output mixed signal **323**) in the electronic internal signal **410** (z(n)). It subtracts the music signal estimate ỹ(n) from the electronic internal signal **410** to produce the modified electronic internal signal e(n) **412**. Thus, the modified electronic signal 1 will include the voice of the initiator of the conference call but not the music. The modified electronic signal 1 is transmitted to the first caller.

The voice decision logic **620** analyzes the modified electronic signal **412** e(n) and the electronic ambient signal **426** to produce a voice activity level **622**, α. The voice activity level α identifies a probability that the user is speaking, for example, when the user is using the earpiece for two way voice communication. The voice activity level **622** can also indicate a degree of voicing (e.g., periodicity, amplitude), When the user is speaking, voice is captured externally from acoustic ambient signal **424** by the ASM **111** in the ambient environment and also by the ECM **123** in the ear canal. The voice decision logic **620** provides the voice activity level α to the acoustic management module **201** as an input parameter for mixing the ASM **111** and ECM **123** signals. Briefly referring back to FIG. **4**, the acoustic management module **201** performs the mixing as a function of the voice activity level α and the background noise level (see G=f(BNL)+f(VAL)).

For instance, at low background noise levels and low voice activity levels, the acoustic management module **201** amplifies the electronic ambient signal **426** from the ASM **111** relative to the electronic internal signal **410** from the ECM **123** in producing the mixed signal **323**. At medium background noise levels and medium voice activity levels, the acoustic management module **201** attenuates low frequencies in the electronic ambient signal **426** and attenuates high frequencies in the electronic internal signal **410**. At high background noise levels and high voice activity levels, the acoustic management module **201** amplifies the electronic internal signal **410** from the ECM **123** relative to the electronic ambient signal **426** from the ASM **111** in producing the mixed signal. The acoustic management module **201** can additionally apply frequency specific filters based on the characteristics of the background noise.

FIG. **7** is a schematic of a control unit **700** for controlling adaptation of a first set (**736**) and a second set (**738**) of filter coefficients of the adaptive filter **610** for in-ear canal echo suppression in accordance with an exemplary embodiment. The description hereinbelow also applies to adaptive filter **612**. Briefly, the control unit **700** illustrates a freezing (fixing) of weights in upon detection of spoken voice. The adaptive filter **610** resumes weight adaptation when e(n) is low, and freezes weights when e(n) is high signifying presence of spoken voice.

When the user is not speaking, the ECR **125** can pass through ambient sound captured at the ASM **111**, thereby allowing the user to hear environmental ambient sounds. In an echo suppression mode, the adaptive filter **610** models an ECTF and suppresses an echo of the mixed signal **323** that is looped back to the ECR **125** by way of the ASM **111** (see dotted line Loop Back path). When the user is not speaking, the suppressor continually adapts to model the ECTF. When the ECTF is properly modeled, the adaptive filter **610** produces a modified internal electronic signal e(n) that is low in amplitude level (i.e, low in error). The suppressor adapts the weights to keep the error signal low. When the user speaks, the suppressor however initially produces a high-level e(n) (e.g., the error signal increases). This happens since the speaker's voice is uncorrelated with the audio signal played out the ECR **125**, which disrupts the ECTF modeling ability of adaptive filter **610**.

The control unit **700** upon detecting a rise in e(n), freezes the weights of the adaptive filter **610** to produce a fixed filter H'(w) fixed **738**. Upon detecting the rise in e(n) the control unit adjusts the gain **734** for the ASM signal and the gain **732** for the mixed signal **323** that is looped back to the ECR **125**. The mixed signal **323** fed back to the ECR **125** permits the user to hear themself speak. Although the weights are frozen when the user is speaking, a second filter H'(w) **736** continually adapts the weights for generating a second e(n) that is used to determine presence of spoken voice. That is, the control unit **700** monitors the second error signal e(n) produced by the second filter **736** for monitoring a presence of the spoken voice.

The first error signal e(n) (in a parallel path) generated by the first filter **738** is used as the mixed signal **323**. The first error signal contains primarily the spoken voice since the ECTF model has been fixed due to the weights. That is, the second (adaptive) filter is used to monitor a presence of spoken voice, and the first (fixed) filter is used to generate the mixed signal **323**.

Upon detecting a fall of e(n), the control unit restores the gains **734** and **732** and unfreezes the weights of the suppressor, and the first filter H'(w) **738** returns to being an adaptive filter. The second filter H'(w) **736** remains on stand-by until spoken voice is detected, and at which point, the first filter H'(w) **738** goes fixed, and the second filter

H'(w) **736** begins adaptation for producing the e(n) signal that is monitored for voice activity. Notably, the control unit **700** monitors e(n) from the first filter **738** or the second filter **736** for changes in amplitude to determine when spoken voice is detected based on the state of voice activity.

FIG. **10** is an illustration of a system for suppressing a siren or acoustic signal from a communication signal in accordance with an exemplary embodiment. In general, an emergency vehicle such as a police car, military vehicle, fire truck, or ambulance is a noisy environment. Typically, the vehicle has a loud siren or horn that is played periodically in an emergency situation to alert people that the vehicle is approaching. The siren is of sufficient loudness that it penetrates the cabin of the vehicle where it is picked up by a communication device being used. The siren can make the intelligibility of the speech difficult to understand on the receiving end of the communication. For example, an ambulance medical technician is rapidly telling a hospital the condition of a patient while the siren is blaring in the background. Removing the siren from the communication would allow the speech to be heard with clarity and allow concise recognition of the information being provided.

In at least one exemplary embodiment, a system for a vehicle **1002** comprises a transducer **1006** for emitting an acoustic signal (e.g. siren), a cabin loudspeaker **1004**, an "invisible audio" (IA) system **1010**, a communication system **1012**, and a microphone **1008**. The siren acoustic component detected by cabin microphone **1008** is removed by the "Invisible Audio" (IA) system **1010** before the voice communication signal is transmitted to a remote party via communication system **1012**. Removing the siren or acoustic signal increases the intelligibility of the transmitted voice communication signal to a remote party, e.g. an emergency call center. In another embodiment, warning signals that can be reproduced with cabin loudspeaker **1004**, such as voice audio signals from other calling parties or vehicle warning systems, can also be removed (or attenuated) by the IA system **1010**.

FIG. **11** is a block diagram of the invisible audio system of FIG. **10** for suppressing the siren or acoustic signal in accordance with an exemplary embodiment. An emergency siren audio signal **1112** can be generated either algorithmically using a digital signal processing system, or can be acquired from computer readable memory, and is emitted using acoustic siren transducer **1116** (e.g. a loudspeaker mounted on the outside of the emergency vehicle). The emergency siren audio signal **1112** is further filtered using filter **1118** and subtracted from the cabin microphone signal **1122** using signal subtraction unit **1120**, to generate a modified microphone signal **1114**. In at least one exemplary embodiment, filter **1118** is a fixed FIR-type digital filter, where the impulse response approximates the acoustic impulse response between the siren transducer **1116** and cabin microphone **1122**. In another exemplary embodiment, filter **1118** is an adaptive filter, where the filter coefficients are updated over time to approximate the acoustic impulse response between the siren transducer **1116** and cabin microphone **1122**. The adaptive filter can be updated according to one of many well-known adaptive filter algorithms, such as the normalized least-mean-square algorithm (NLMS) or recursive least-mean-square algorithm. The modified microphone signal **1114** can then be further transmitted to a remote party, such as an emergency caller control center with better intelligibility and clarity.

Where applicable, the present embodiments of the invention can be realized in hardware, software or a combination of hardware and software. Any kind of computer system or other apparatus adapted for carrying out the methods described herein are suitable. A typical combination of hardware and software can be a mobile communications device with a computer program that, when being loaded and executed, can control the mobile communications device such that it carries out the methods described herein. Portions of the present method and system may also be embedded in a computer program product, which comprises all the features enabling the implementation of the methods described herein and which when loaded in a computer system, is able to carry out these methods.

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 acoustic management control suitable for use in an earpiece, the method comprising the steps of:

measuring an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal;

measuring an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal;

generating a background noise level (BNL) from at least one of the electronic ambient signal or the electronic internal signal or a combination of both;

generating a voice activity level (VAL);

generating a gain, wherein the gain is a function of the background noise level and the voice activity level;

generating a modified electronic ambient signal by applying the gain to the electronic ambient signal;

mixing the modified electronic ambient signal with an audio signal to produce a mixed audio signal; and

sending the mixed audio signal to a speaker.

2. The method according to claim **1** further including the step of:

selecting an amount of the ambient sound to pass through into an ear canal, where the amount is user selected.

3. The method according to claim **2**, where a user selects the amount using a software application on a device.

4. The method according to claim **3**, where the device is at least one of a phone, a watch or a computer.

5. The method according to claim **4**, where the user selects the amount using a slider in a GUI of the software application.

6. The method according to claim **2**, where a user selects the amount by touching a portion of a communication device.

7. The method according to claim **2**, where a user selects the amount by vocalizing a voice command to enter an ambient or transparency mode.

8. A communication device, comprising:

a memory that stores instructions; and

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

capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal;

capturing an internal sound signal from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal;

measuring a background noise signal from at least one of the electronic ambient signal or the electronic internal signal or both;

generating a background noise level (BNL) from the background noise signal;

generating a voice activity level using the internal sound signal;

generating a gain, wherein the gain is a function of the background noise level and the voice activity level;

generating a modified electronic ambient signal by applying the gain to the electronic ambient signal;

mixing the modified electronic ambient signal with an audio signal at a user selected ratio to produce a mixed audio signal; and

sending the mixed audio signal to a speaker.

9. The device according to claim **8**, where the operation further comprises:

selecting the ratio by user interaction with a software application on a second device.

10. The device according to claim **9**, where the second device is at least one of a phone, a watch or a computer.

11. The method according to claim **9**, where the user selects the ratio using a slider in a GUI of the software application.

12. The device according to claim **8**, where the user selects a ratio by touching a portion of the communication device.

13. The device according to claim **8**, where the user selects the ratio by vocalizing a voice command to enter an ambient or transparency mode.

14. An earphone comprising:

a sealing section configured to form an acoustic seal when the earphone is used;

an ambient sound microphone (ASM) that produces an electronic ambient signal;

an ear canal microphone (ECM) that produces an electronic internal signal;

a memory that stores instructions; and

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

receiving the electronic ambient signal;

receiving the electronic internal signal;

calculating a background noise signal from the electronic ambient signal or the electronic internal signal or a combination of both;

receiving an audio signal;

generating a noise reduction signal;

determining if the earphone is acoustically sealed;

producing an adjusted audio signal using the noise reduction signal;

receiving a mixing level;

generating a mixed audio signal by mixing the adjusted audio signal with the electronic ambient signal using the mixing level; and

sending the mixed audio signal to a speaker.

15. The earphone according to claim **14**, where the mixing level is set by a user from a low level of mixing to a high level of mixing, where a low level of mixing is where less than half of the ambient signal by amplitude is mixed with the adjusted audio signal, and where a high level of mixing is where more than half of the ambient signal by amplitude is mixed with the adjusted audio signal.

* * * * *

Exhibit E

US011659315B2

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

(10) **Patent No.: US 11,659,315 B2**
(45) **Date of Patent: May 23, 2023**

(54) **METHODS AND MECHANISMS FOR INFLATION**

(71) Applicant: **Staton Techiya LLC**, Lighthouse Point, FL (US)

(72) Inventors: **Moises Perez**, Delray Beach, FL (US); **Steven W. Goldstein**, Delray Beach, 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 33 days.

(21) Appl. No.: **17/211,814**

(22) Filed: **Mar. 24, 2021**

(65) **Prior Publication Data**

US 2021/0211801 A1 Jul. 8, 2021

Related U.S. Application Data

(63) Continuation of application No. 16/266,829, filed on Feb. 4, 2019, now Pat. No. 11,006,199, which is a continuation of application No. 15/968,231, filed on May 1, 2018, now Pat. No. 10,200,775, which is a continuation of application No. 14/109,954, filed on Dec. 17, 2013, now Pat. No. 10,009,677.

(Continued)

(51) **Int. Cl.**

H04R 25/00	(2006.01)
H04R 1/10	(2006.01)
H04R 5/033	(2006.01)

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

CPC ***H04R 1/1041*** (2013.01); ***H04R 5/033*** (2013.01); *H04R 1/1016* (2013.01);

(Continued)

(58) **Field of Classification Search**

CPC H04R 1/1041; H04R 5/033; H04R 1/1016; H04R 25/554; H04R 2420/07; H04R 2430/01; H04R 2460/17

(Continued)

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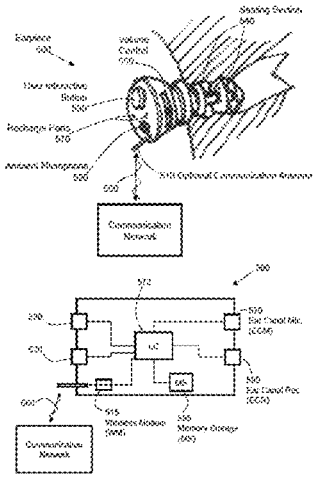
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Primary Examiner — Phylesha Dabney
(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti

(57) **ABSTRACT**

A device includes a balloon inflation module having an involuted balloon housed within a lumen, an integrated reservoir in fluid communication with the involuted balloon where the balloon inflation module is configured to selectively displace a volume of fluid from the integrated reservoir into the involuted balloon and from the involuted balloon into the integrated reservoir. The device also includes a port configured to receive an accessory module. In one embodiment, the accessory module is a communication module having a logic circuit coupled to an ambient microphone configured to pick up ambient audio content, an ear canal microphone configured to pick up audio in the proximity of an ear canal, and an ear canal receiver configured to provide audio content in the proximity of the ear canal. In some embodiments, the integrated reservoir is formed within portions of a push button spring-loaded pump assembly. Other embodiments are disclosed.

13 Claims, 19 Drawing Sheets



Related U.S. Application Data

(60) Provisional application No. 61/737,932, filed on Dec. 17, 2012.

(52) **U.S. Cl.**
CPC *H04R 25/554* (2013.01); *H04R 2420/07* (2013.01); *H04R 2430/01* (2013.01); *H04R 2460/17* (2013.01); *Y10T 137/3584* (2015.04)

(58) **Field of Classification Search**
USPC 381/380
See application file for complete search history.

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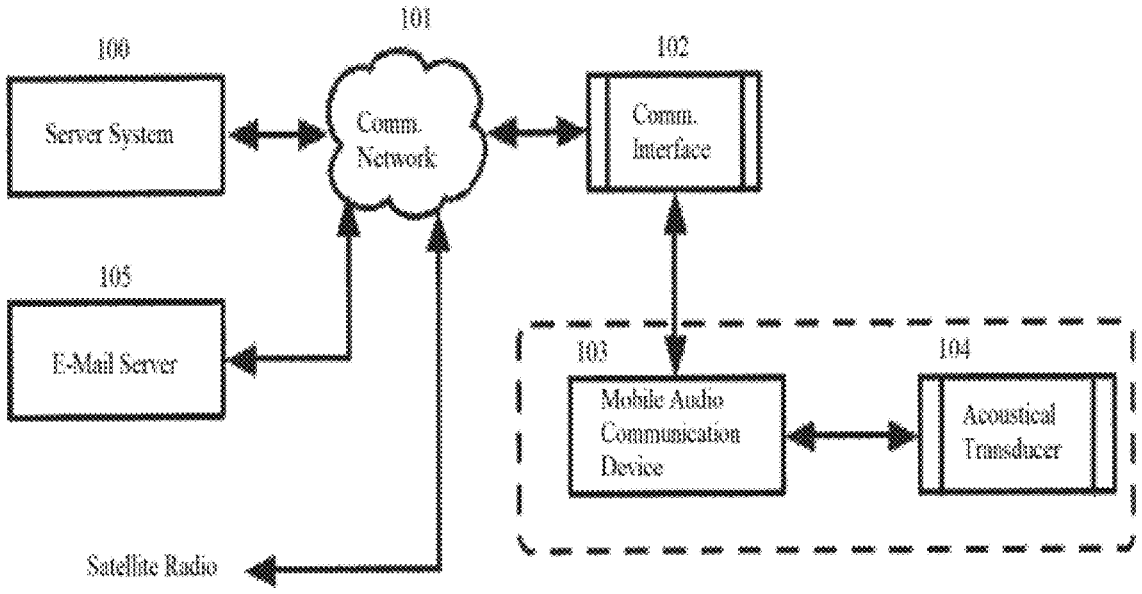

100
Server System
101
Comm. Network
102
Comm. Interface
105
E-Mail Server
103
Mobile Audio Communication Device
104
Acoustical Transducer
Satellite Radio

Fig. 1

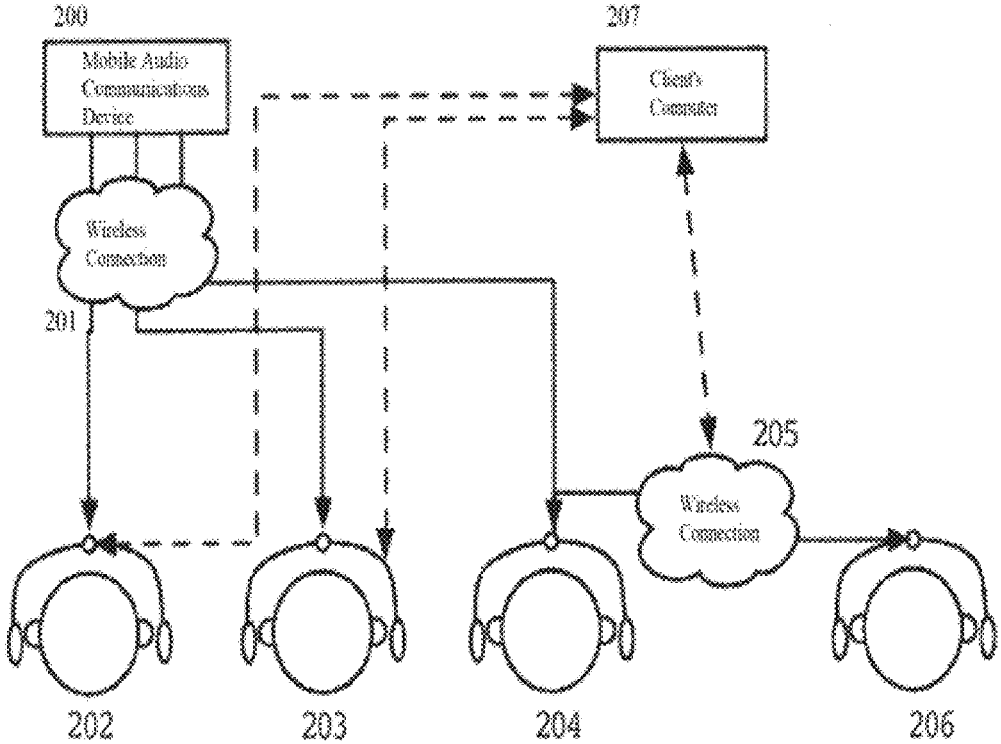

200
Mobile Audio Communication Device
207
Client's Computer
Wireless Connection
201
205
Wireless Connection
202
203
204
206

Fig. 2

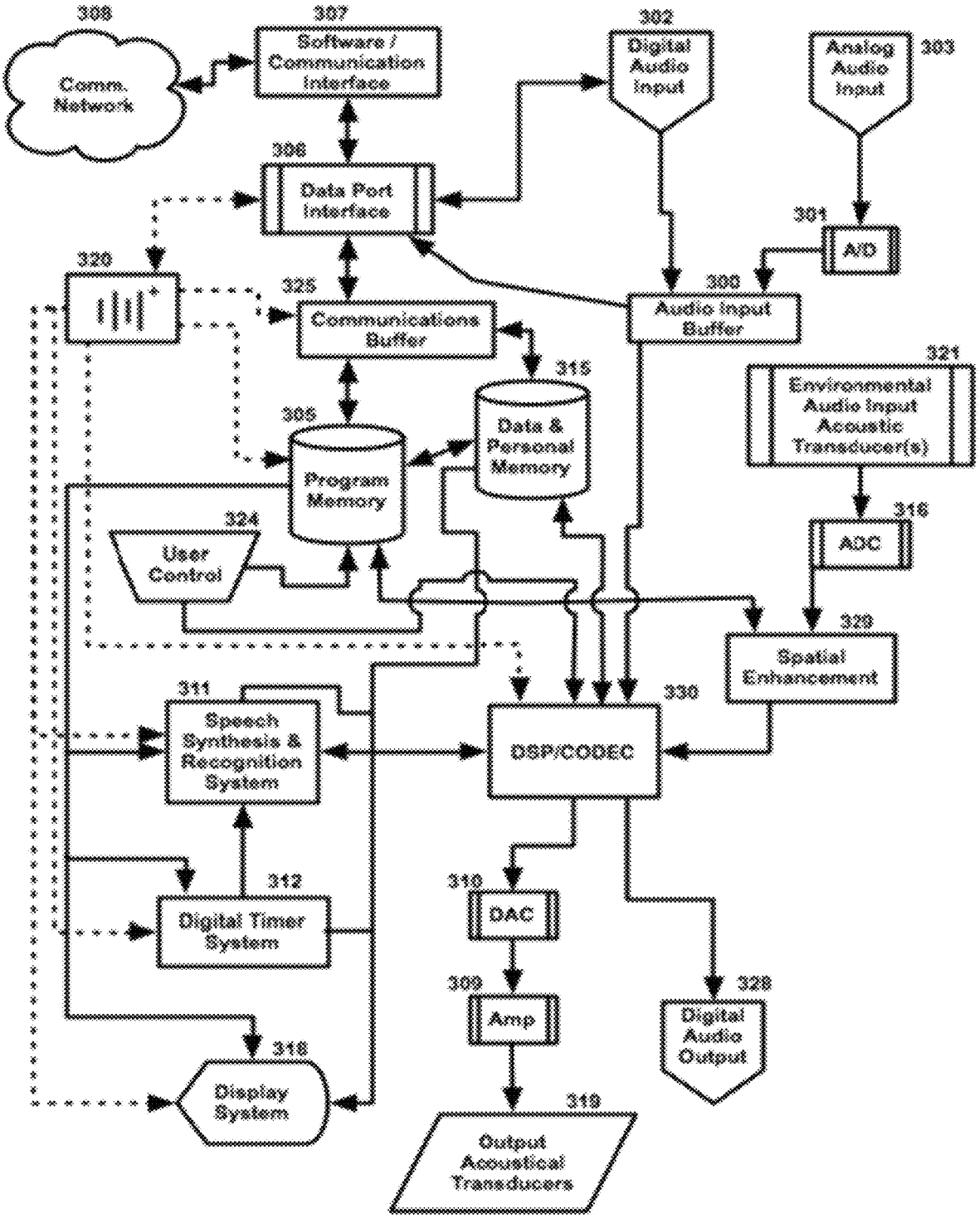

308
Comm. Network
307
Software / Communication Interface
302
Digital Audio Input
303
Analog Audio Input
306
Data Port Interface
301
A/D
320
325
Communications Buffer
304
Audio Input Buffer
315
321
Environmental Audio Input Acoustic Transducer(s)
305
Data & Personal Memory
Program Memory
316
ADC
324
User Control
329
Spatial Enhancement
311
330
Speech Synthesis & Recognition System
DSP/CODEC
312
Digital Timer System
310
DAC
309
Amp
328
Digital Audio Output
318
Display System
319
Output Acoustical Transducers

Fig. 3

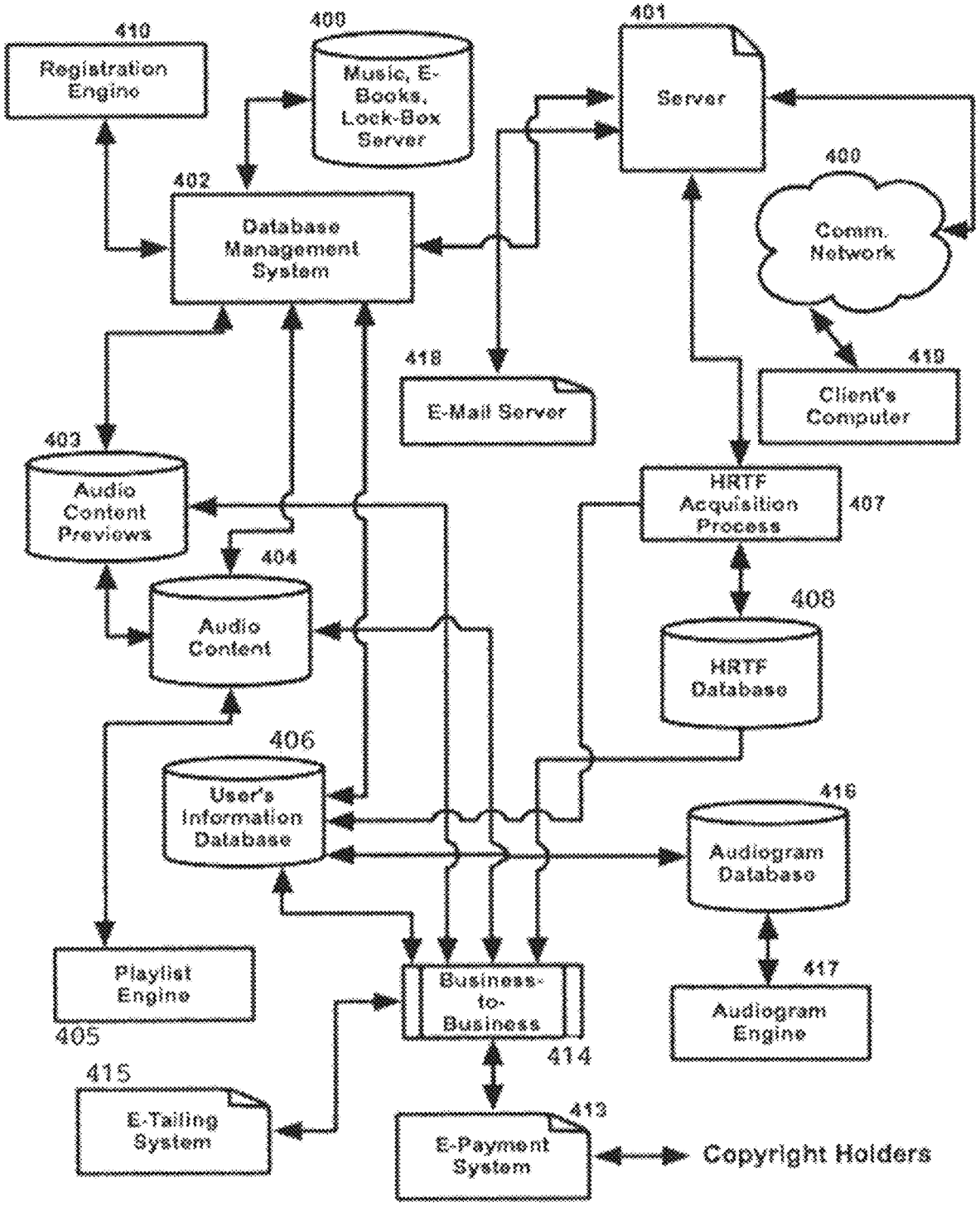

410
Registration Engine
400
Music, E-Books, Lock-Box Server
401
Server
402
Database Management System
400
Comm. Network
418
E-Mail Server
419
Client's Computer
403
Audio Content Previews
404
Audio Content
HRTF Acquisition Process
407
408
HRTF Database
406
User's Information Database
416
Audiogram Database
Playlist Engine
405
Business-to-Business
414
417
Audiogram Engine
415
E-Tailing System
413
E-Payment System
Copyright Holders

Fig. 4

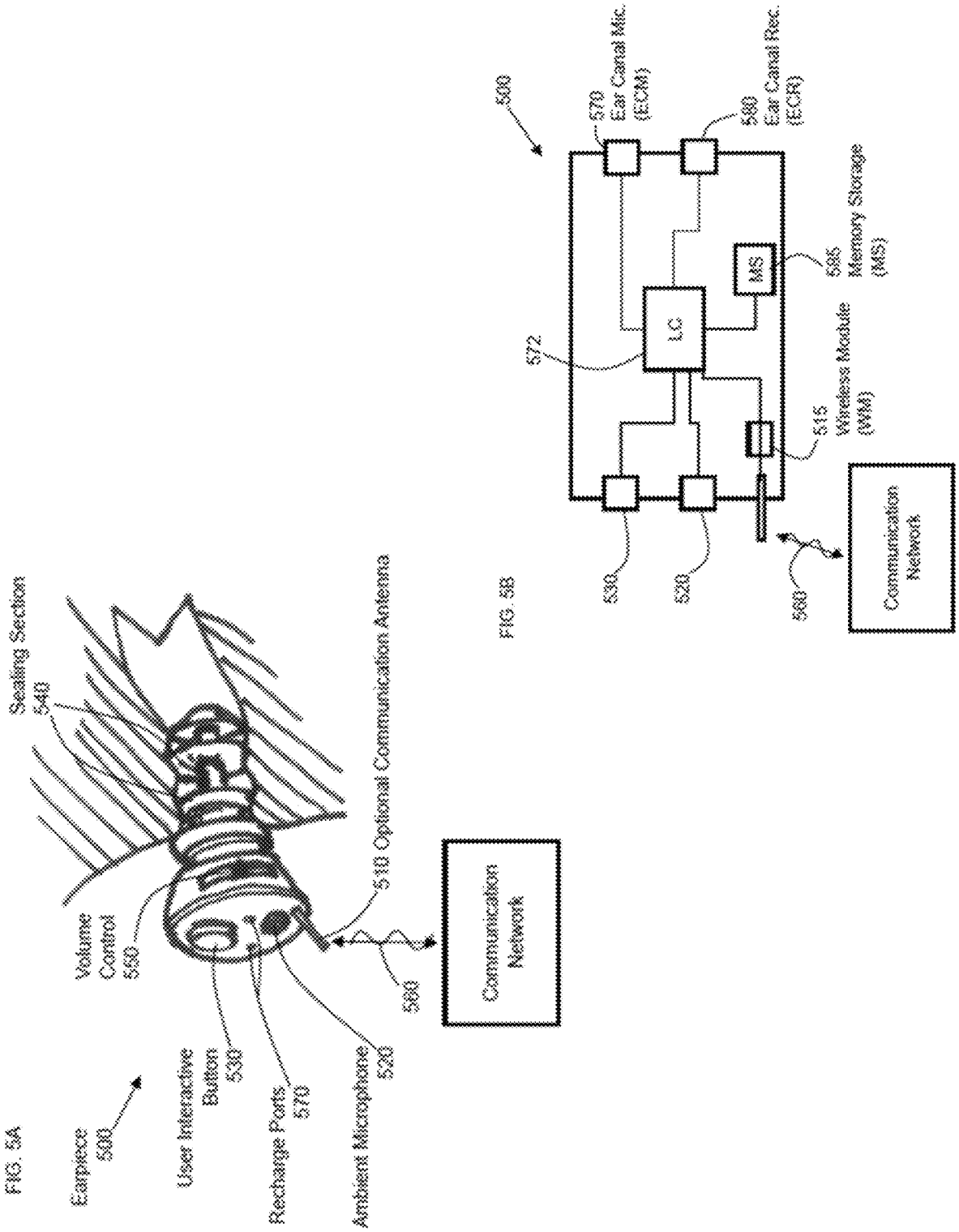
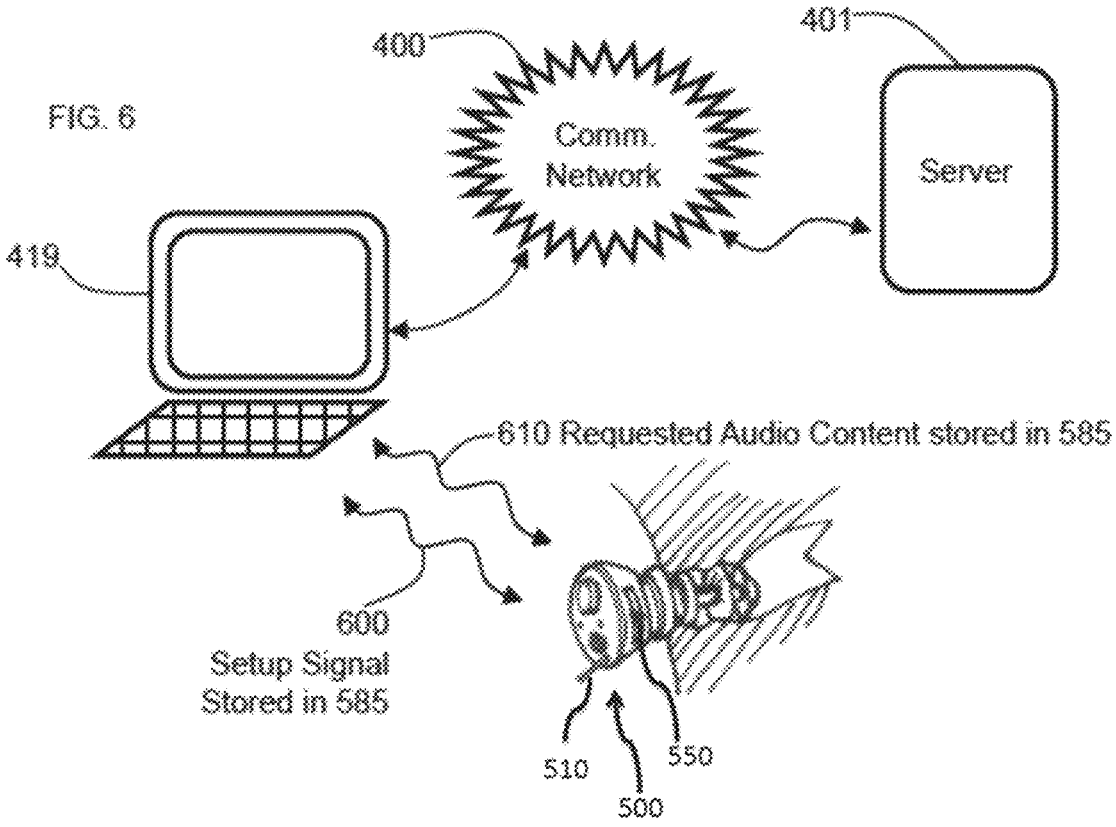

Sealing Section
540
Volume Control
550
User Interactive Button
530
Recharge Ports
570
Ambient Microphone
520
510 Optional Communication Antenna
560
Communication Network
Earpiece
500
FIG. 5A
500
570 Ear Canal Mic. (ECM)
580 Ear Canal Rec. (ECR)
585 Memory Storage (MS)
MS
LC
572
515 Wireless Module (WM)
530
520
560
Communication Network
FIG. 5B

FIG. 6


400
401
Comm. Network
Server
419
610 Requested Audio Content stored in 585
600
Setup Signal Stored in 585
510
550
500

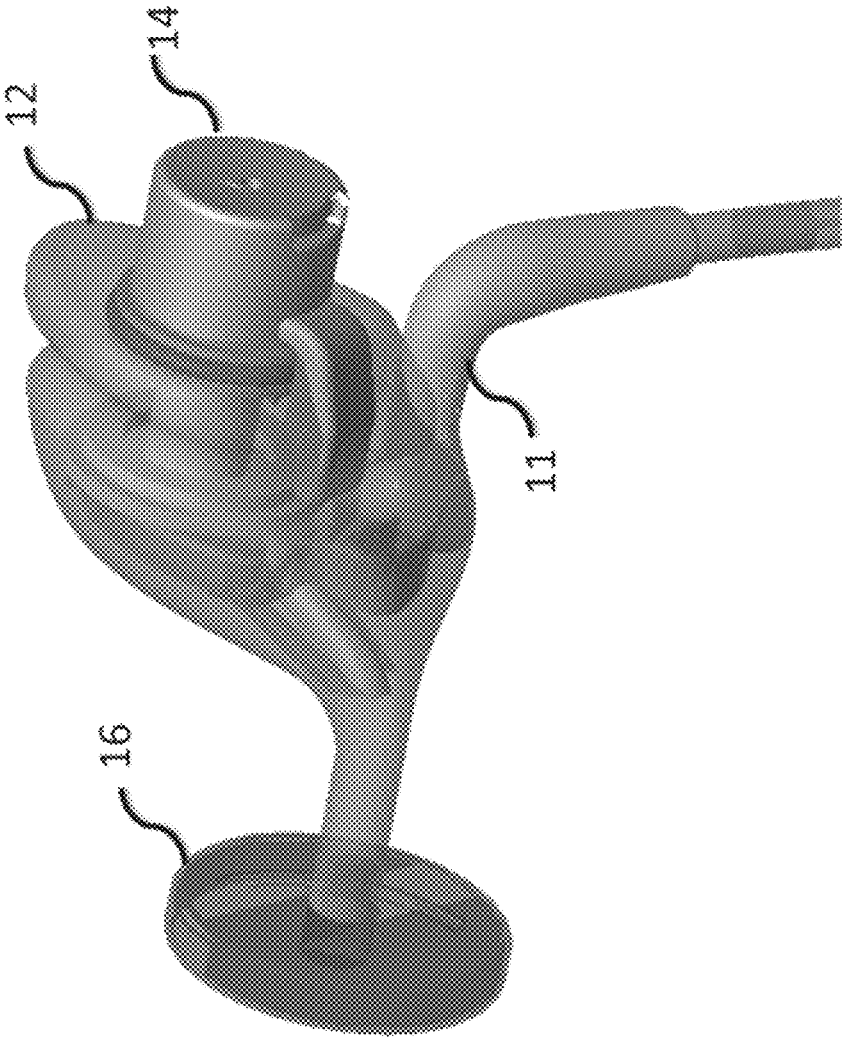


FIG. 7

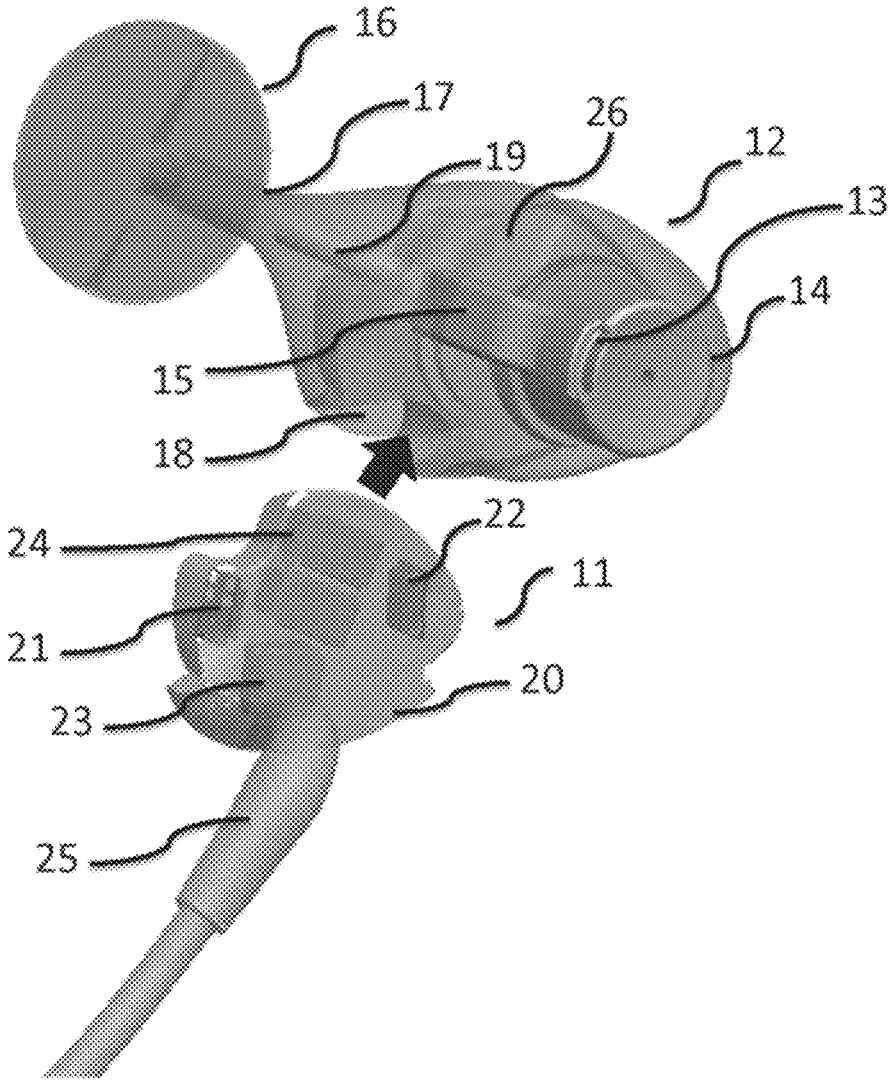


FIG. 8

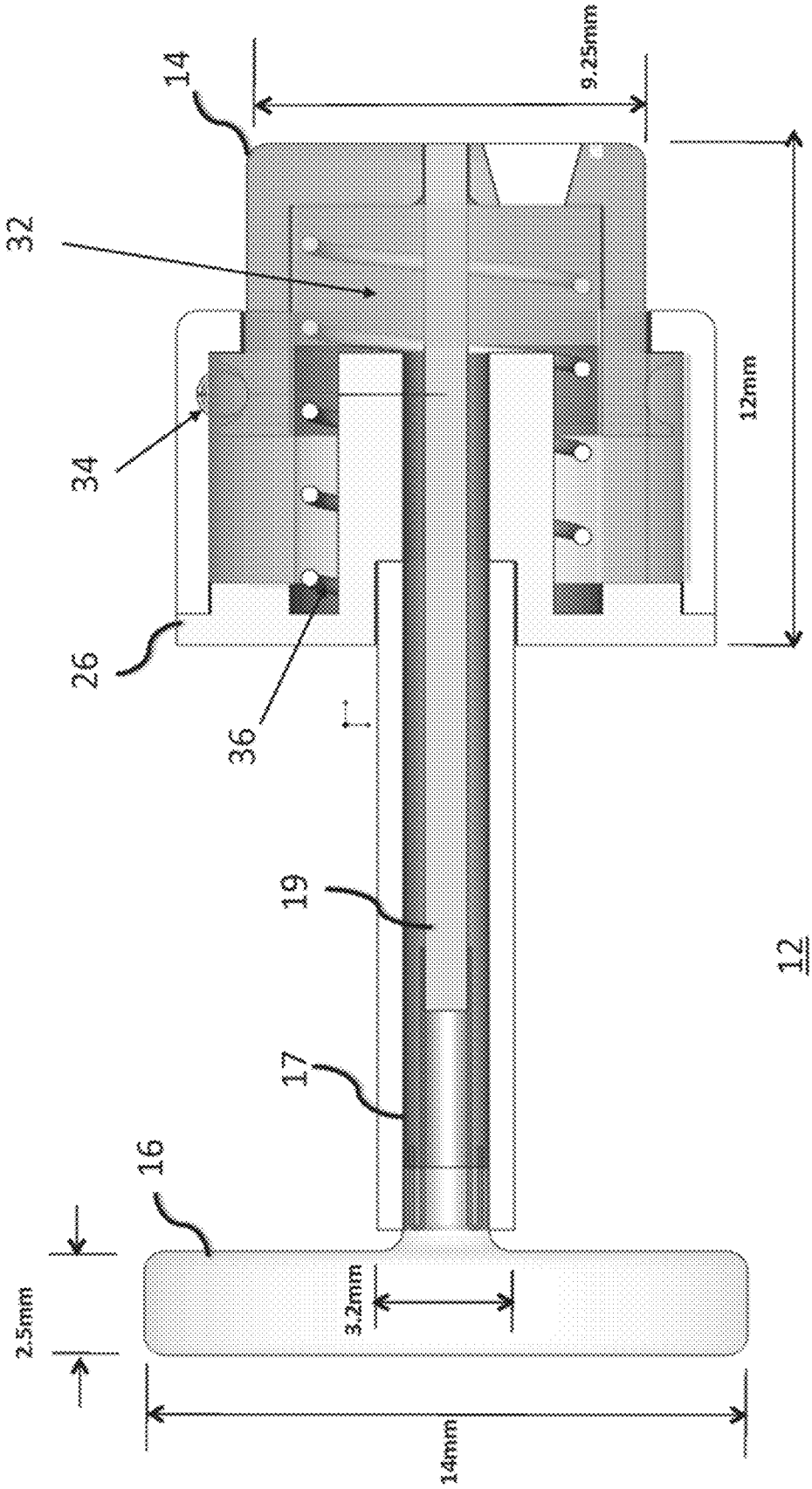


FIG. 9

FIG. 10

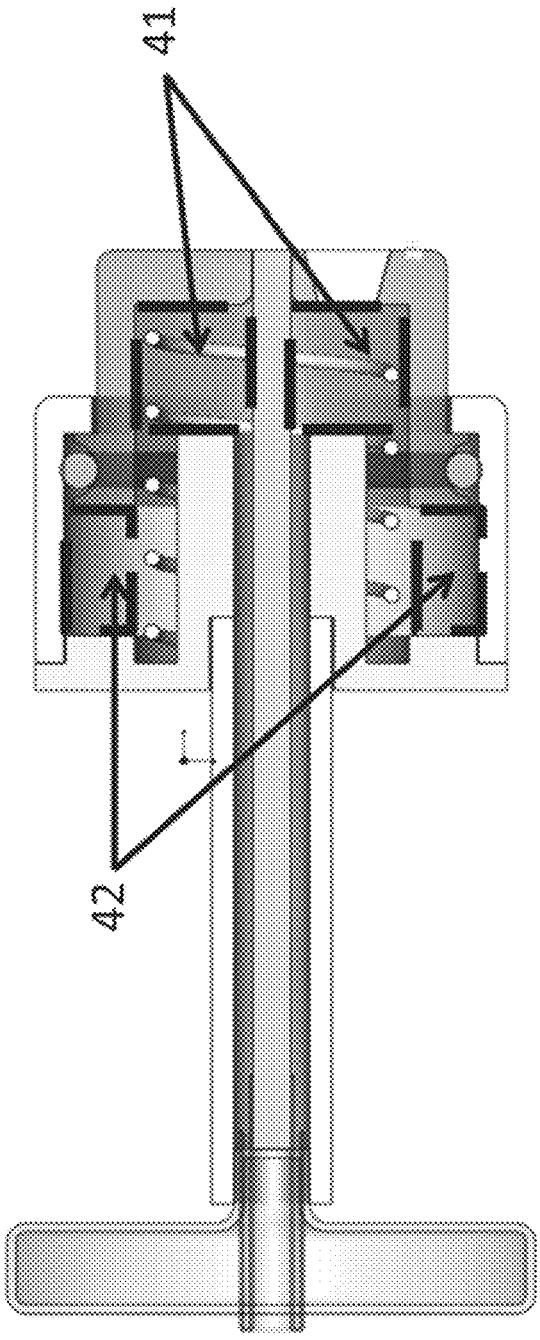
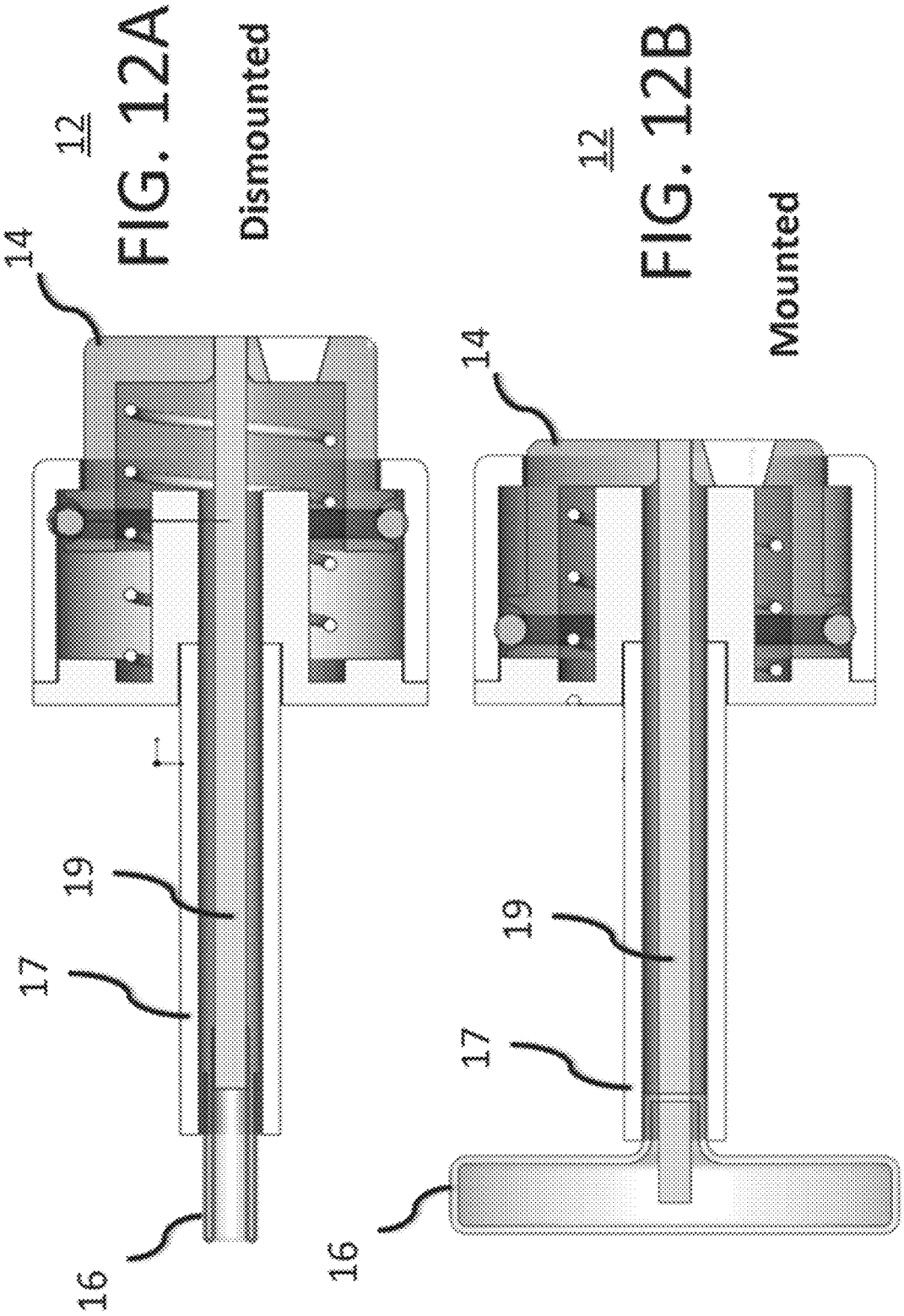
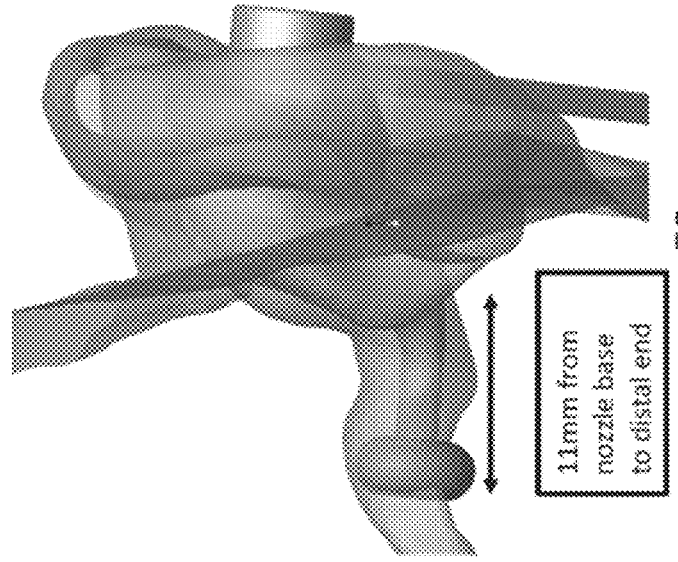


FIG. 11

45

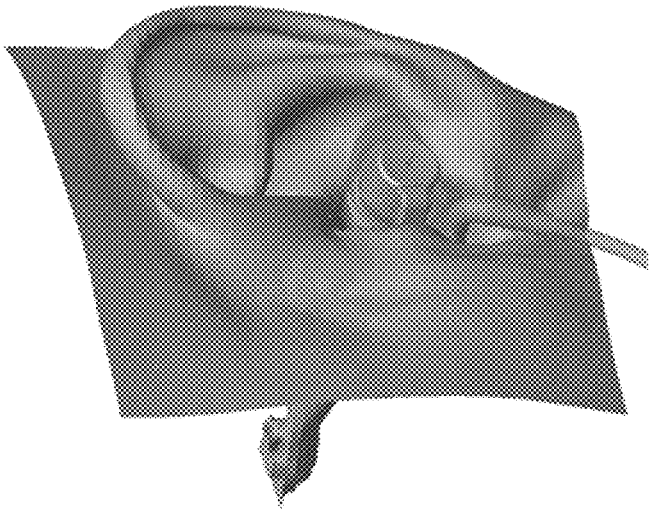
Component	Volume (cc)	Volume (mm^3)
Button Cavity (V1)	0.142	142
Body Cavity (V2)	0.188	188
TOTAL DISPLACEMENT	0.33	330


12
FIG. 12A
Dismounted
14
19
17
16
12
FIG. 12B
Mounted
14
19
17
16



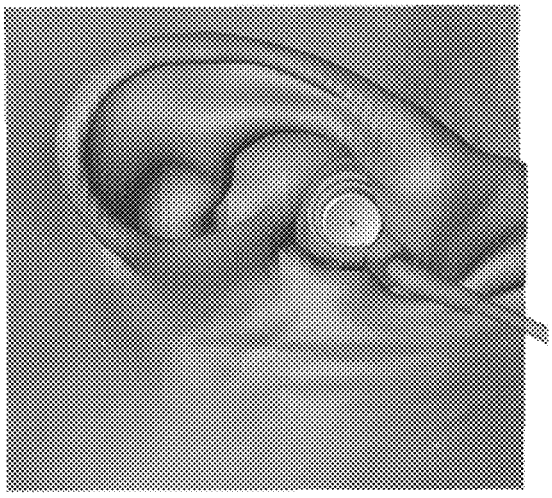
52

FIG. 13C



51

FIG. 13B



50

FIG. 13A

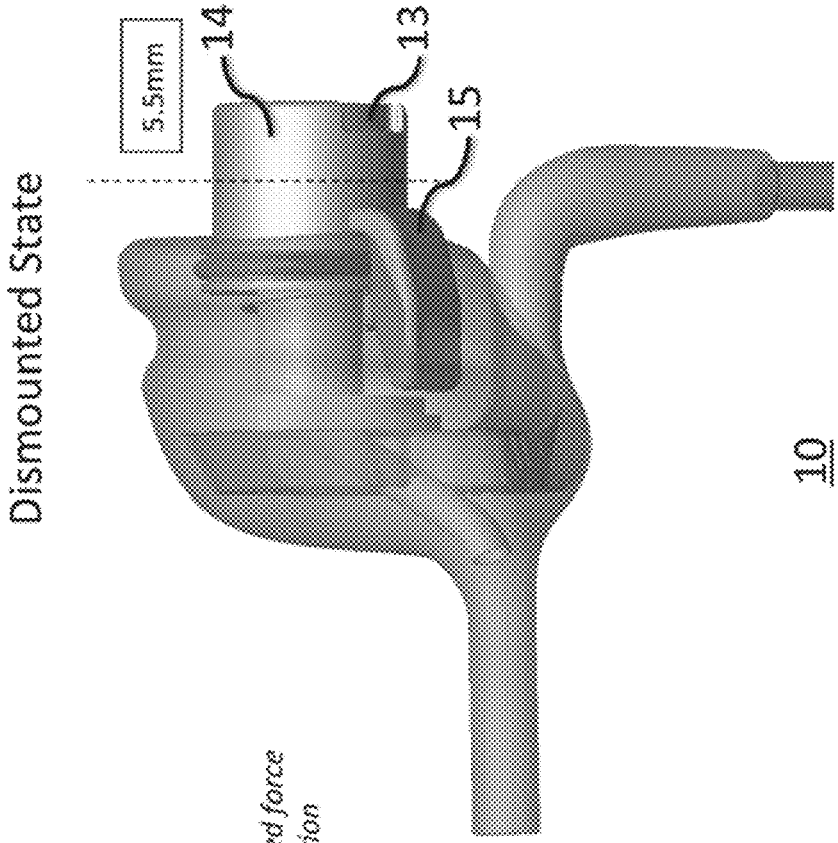


FIG. 14B

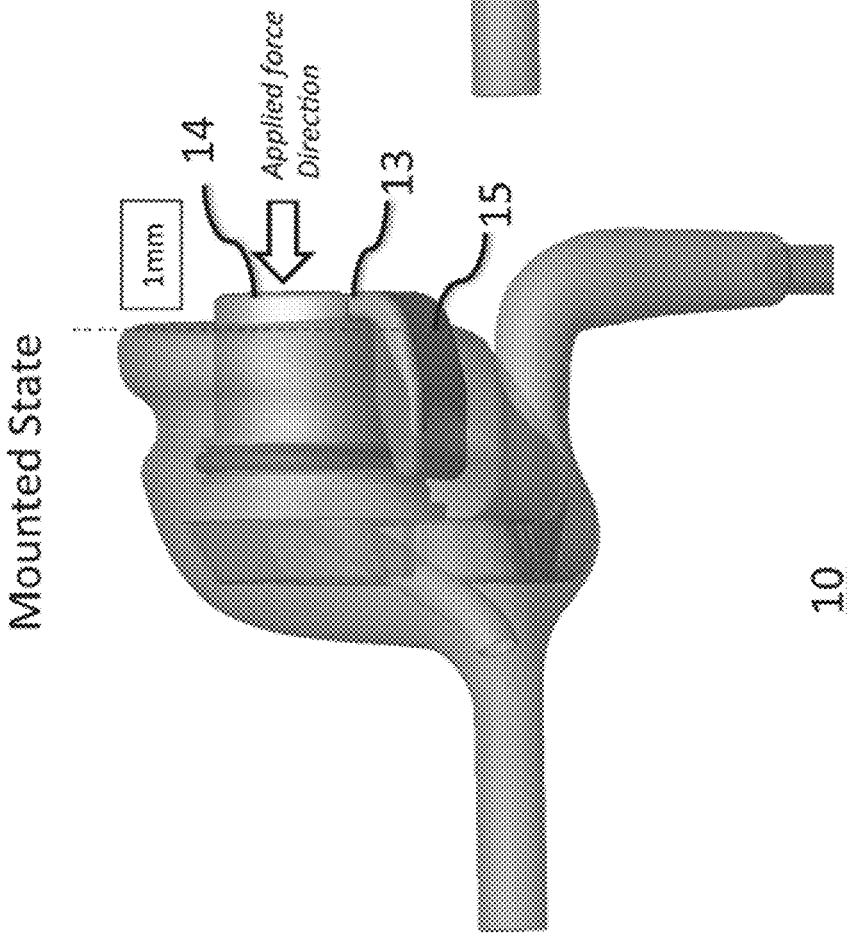


FIG. 14A

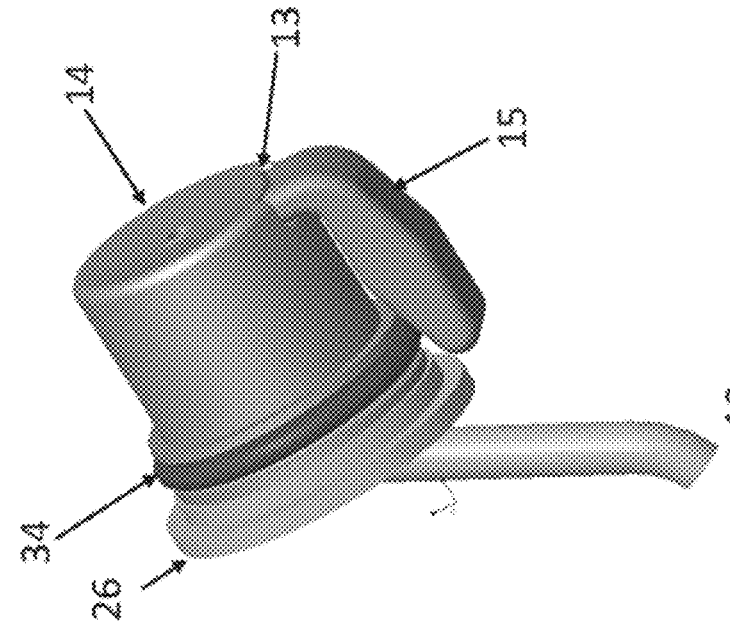


FIG. 15B
Mounted State

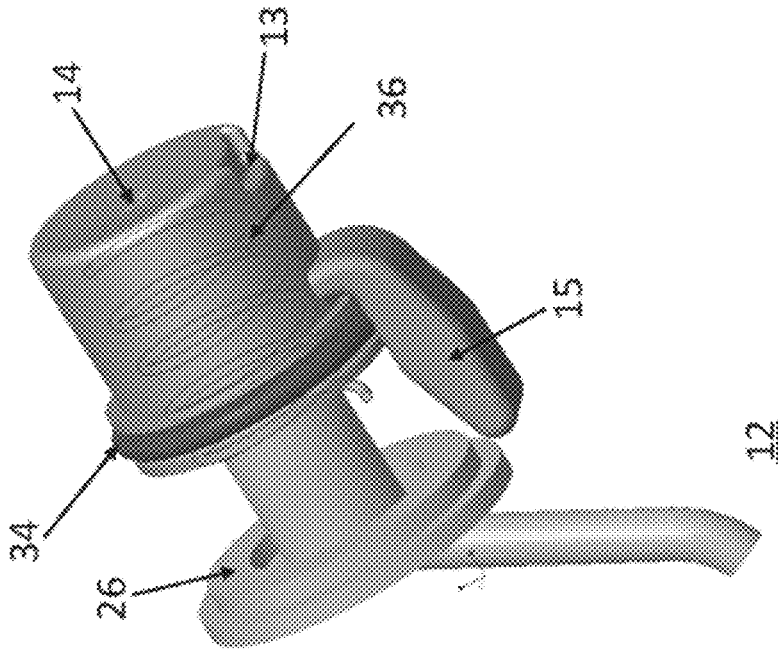


FIG. 15A
Dismounted State

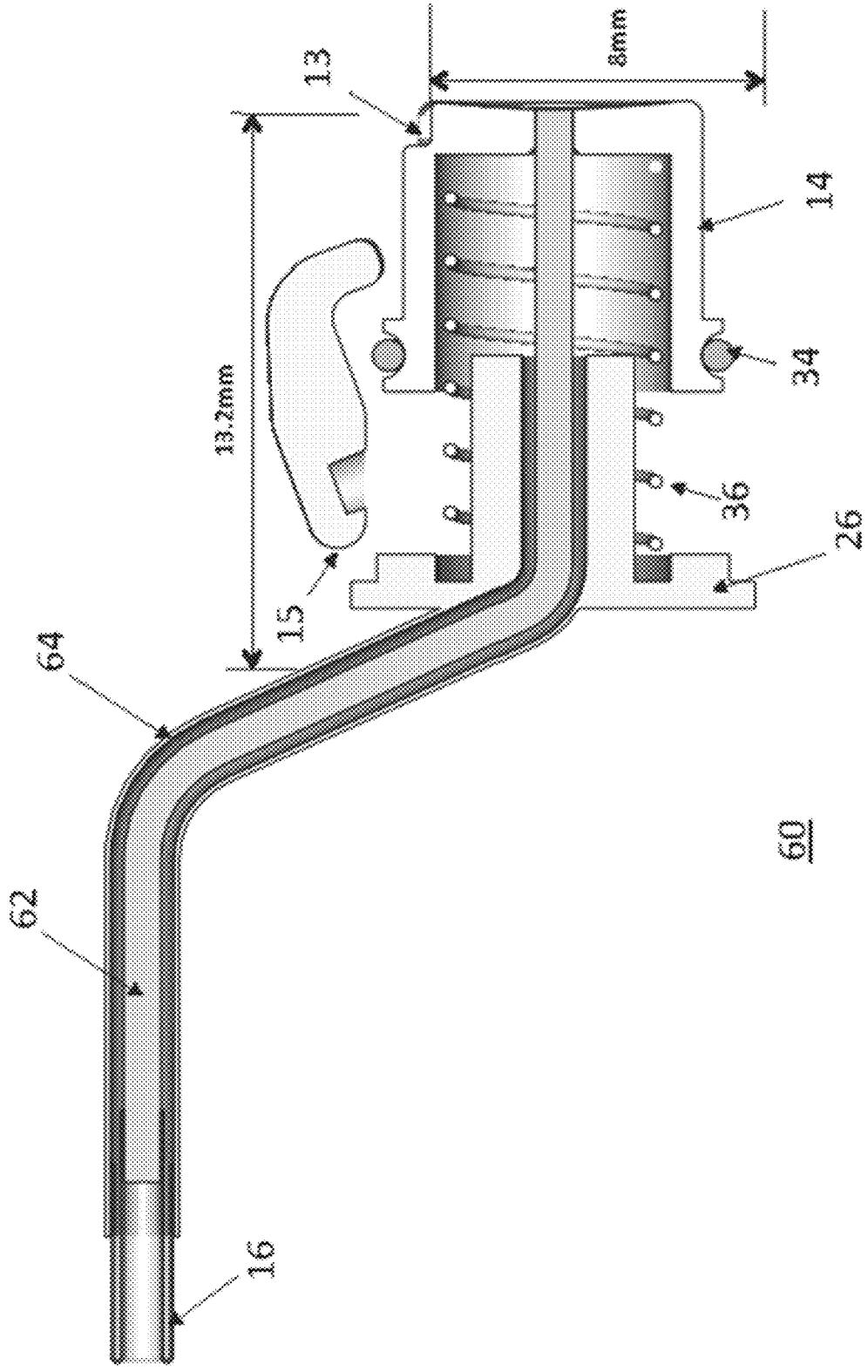


FIG. 16

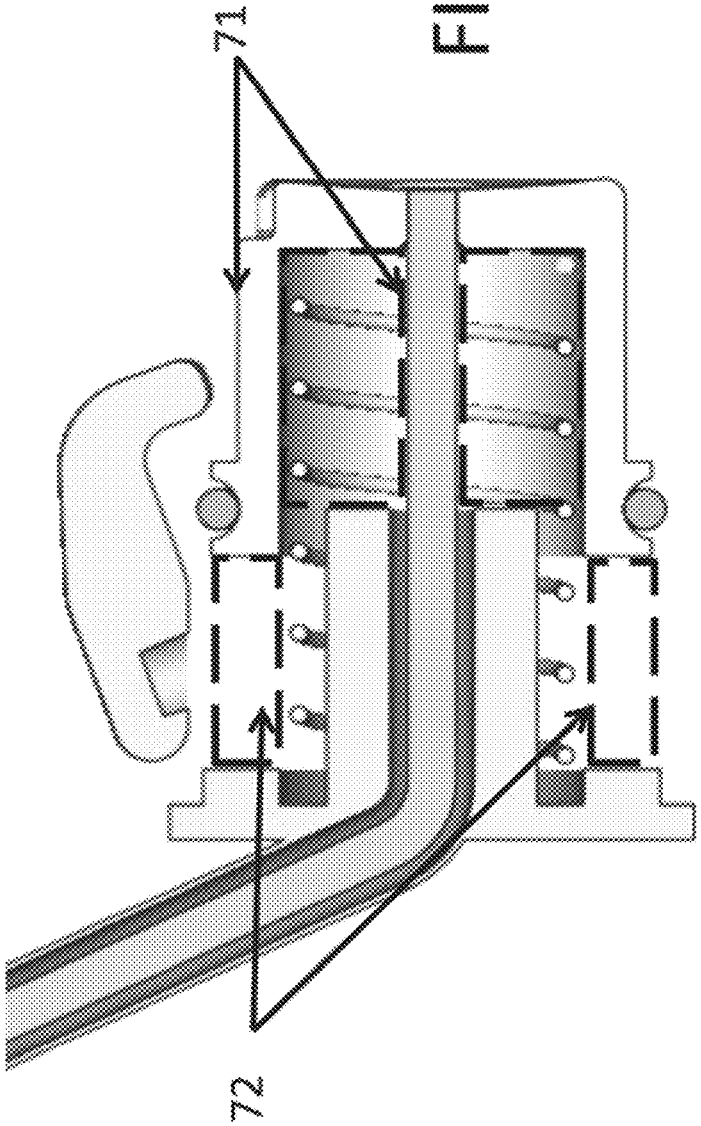


FIG. 17

75

Component	Volume (cc)	Volume (mm^3)
Button Cavity (V1)	.152	152
Body Cavity (V2)	.165	165
TOTAL DISPLACEMENT	0.317	317

FIG. 18

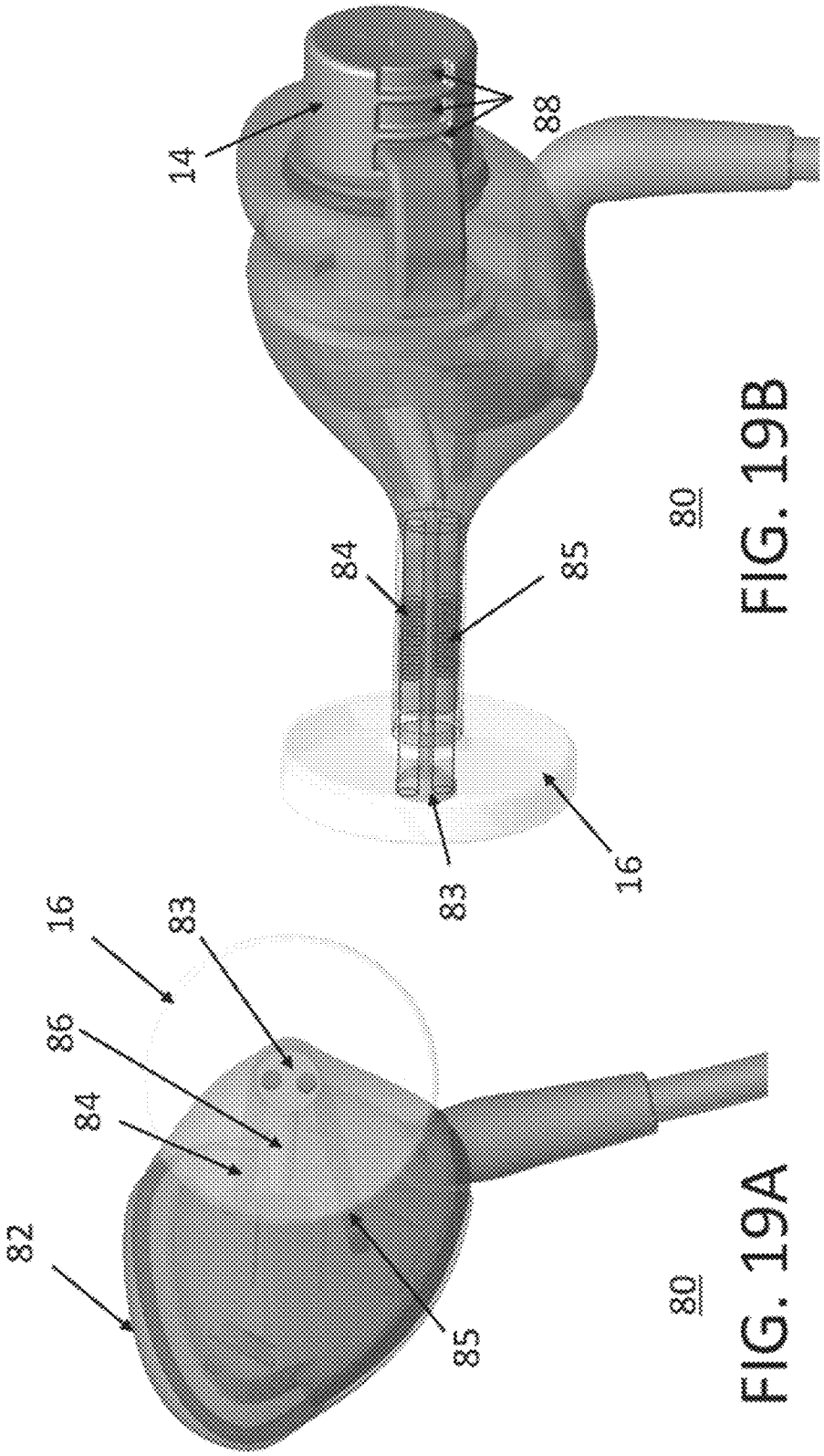


FIG. 19B

FIG. 19A

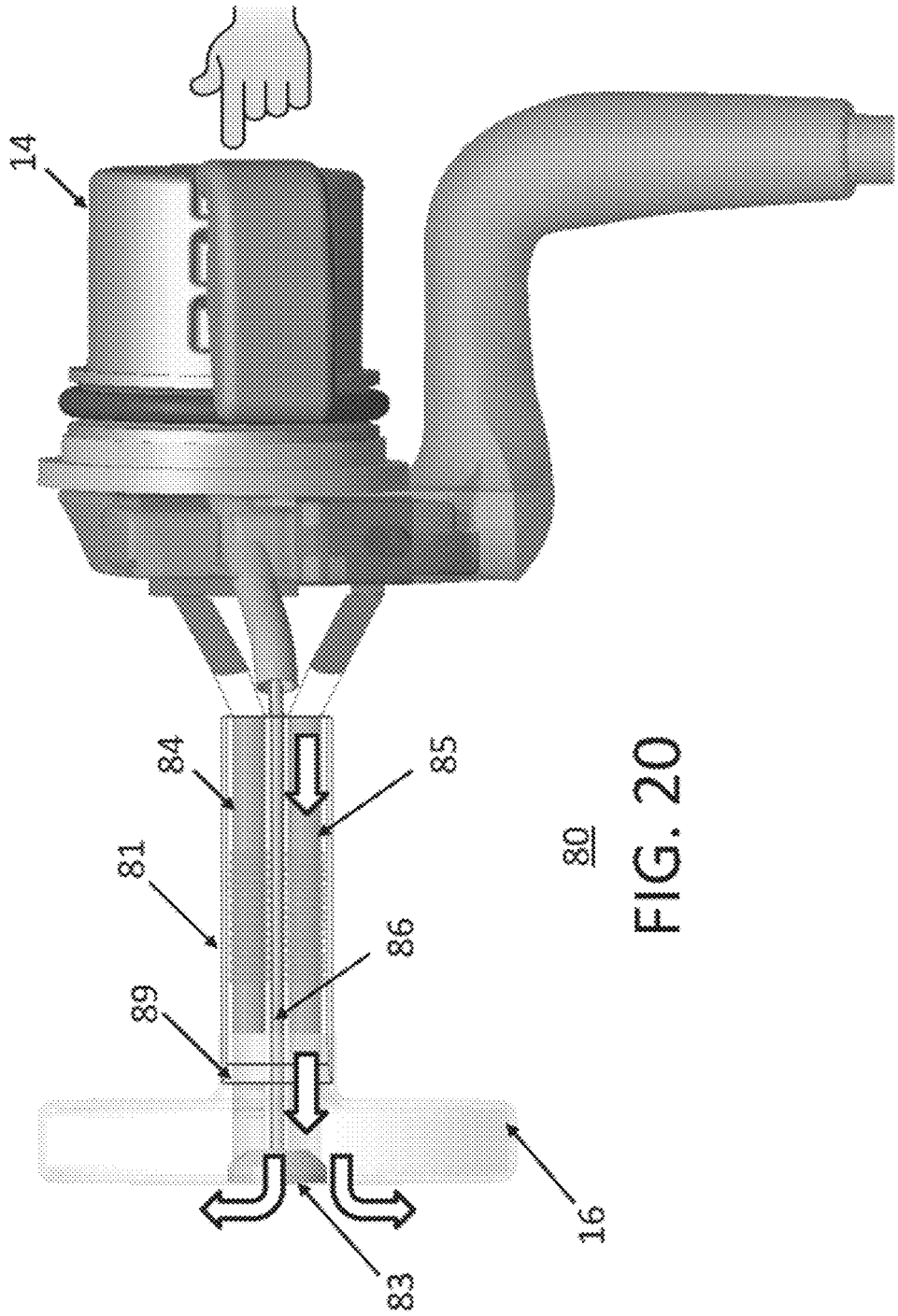


FIG. 20

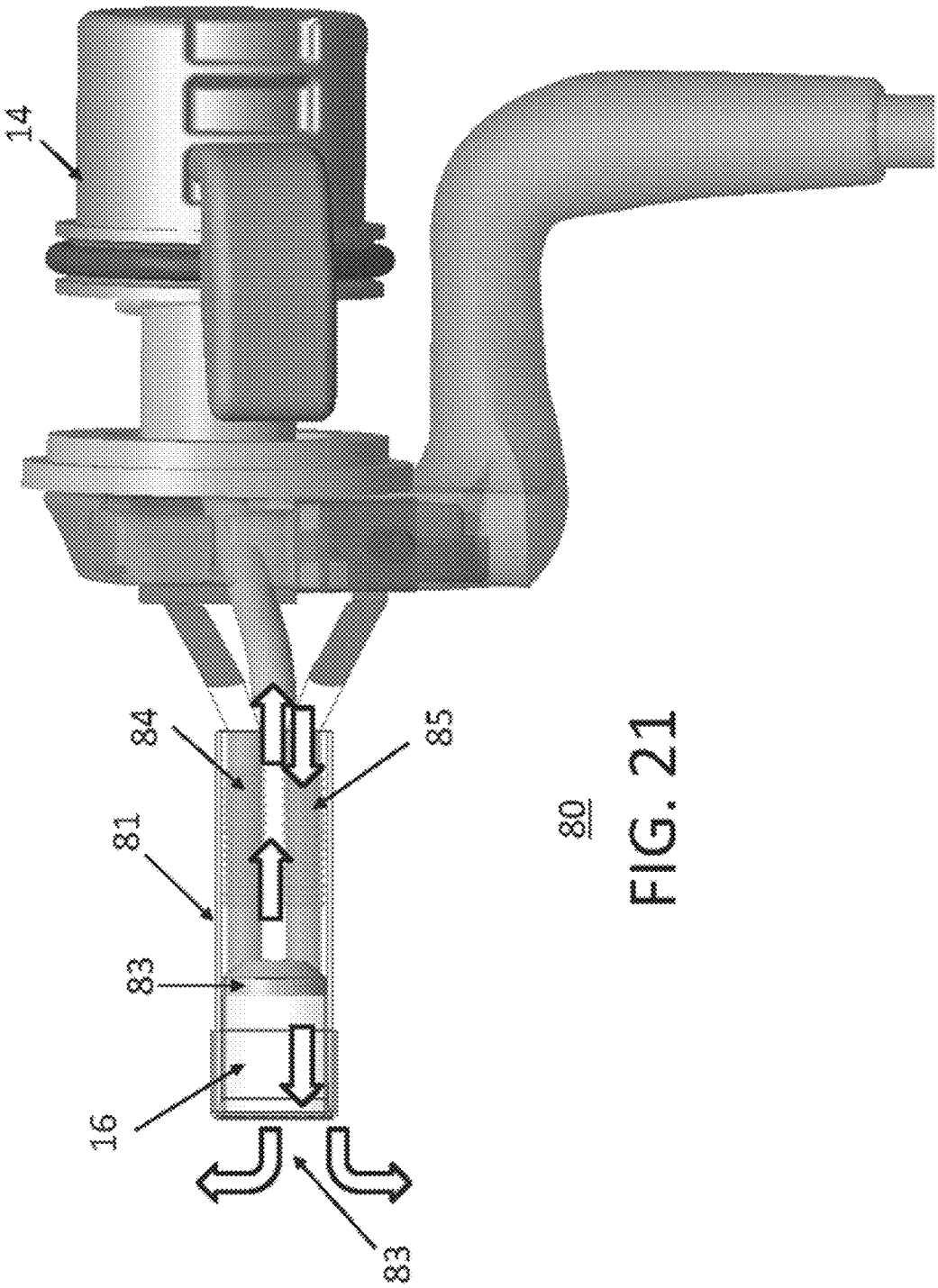


FIG. 21

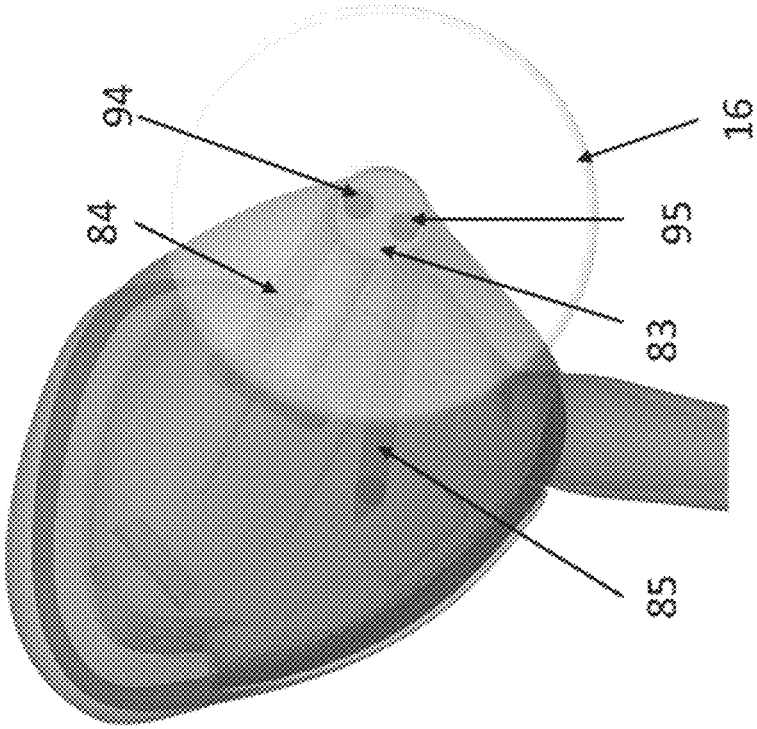


FIG. 23

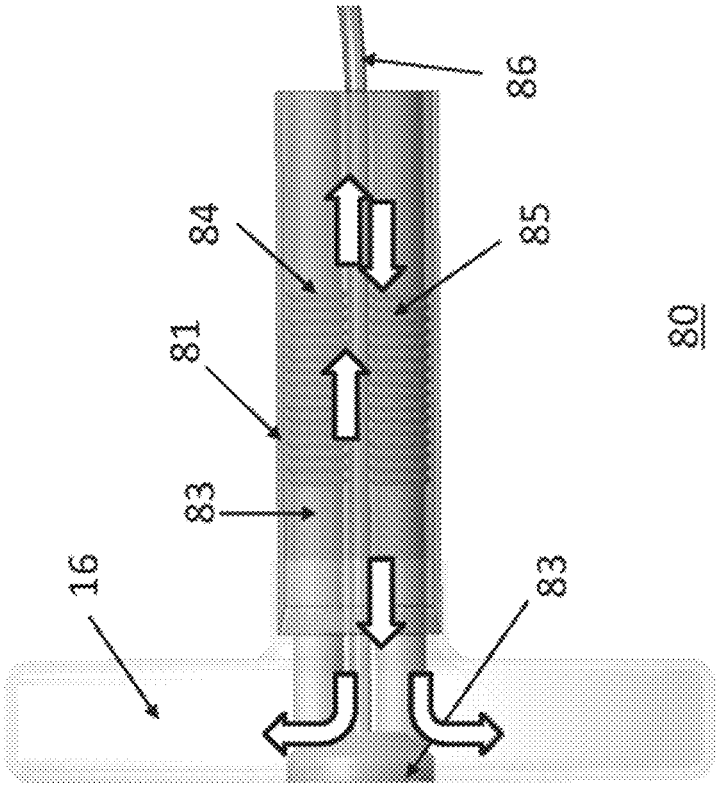


FIG. 22

METHODS AND MECHANISMS FOR INFLATION

CROSS-REFERENCE TO RELATED APPLICATIONS

This application is a continuation of and claims the priority benefit of U.S. patent application Ser. No. 16/266,829, filed Feb. 4, 2019 which is a continuation of and claims the priority benefit of U.S. patent application Ser. No. 15/968,231, filed May 1, 2018, which is a continuation of and claims the priority benefit of U.S. patent application Ser. No. 14/109,954, filed Dec. 17, 2013, now U.S. Pat. No. 10,009,677, which claims the priority benefit of U.S. U.S. Provisional Application No. 61/737,932 filed Dec. 17, 2012, all of which are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

The invention relates in general to methods and devices for inflation, and in particular, though not exclusively, for inflation of a balloon in a device that operates cooperatively with an accessory device.

BACKGROUND OF THE INVENTION

Present audio content playing devices are separated from the headphone system that normally contains the speakers (also referred to as receivers). The reason for this has typically been that audio content has been stored on disks that require a separate playing system. However, even with the advent of storing audio content on non-disk RANI (Random Access Memory) storage systems, the audio content player has been separated from the earpiece system (e.g., plug in headphones or earbuds). Combining the capacity for audio download and playing in an earpiece system is not obvious over related art since the user interaction system (e.g., play button, keyboard system) does not readily appear compatible with the size of an earpiece device and the difficulty of user interaction.

Additionally, no system currently exists for registration and download of audio content into an earpiece. Furthermore, the structures or mechanisms of existing earpieces or earbuds fail to adequately compensate for ambient noise.

BRIEF DESCRIPTION OF THE DRAWINGS

Embodiments of the present invention will become apparent from the following detailed description, taken in conjunction with the drawings in which:

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network;

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity);

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment;

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer;

FIG. **5**A illustrates an earpiece that can store and download audio content in accordance with at least one exemplary embodiment;

FIG. **5**B illustrates a block diagram of the earpiece of FIG. **5**A;

FIG. **6** illustrates a user interface for setting the parameters of an earpiece operating as a Personal Audio Assistant;

FIG. **7** illustrates a device having a balloon inflation module and an accessory module combined in a form of a smart earpiece in accordance with the embodiments;

FIG. **8** illustrates an exploded view of the device of FIG. **7** before the accessory module mates with the balloon inflation module in accordance with the embodiments;

FIG. **9** is a side view of the balloon inflation module of FIG. **7**;

FIG. **10** is another side view of the balloon inflation module of FIG. **7** illustrating volume displacement from cavities in a button assembly of the balloon inflation module towards a balloon;

FIG. **11** is a chart illustrating an example of volume displacements from the cavities in the button assembly of FIG. **10**;

FIG. **12**A is another side view of the balloon inflation module of FIG. **7** illustrating a balloon partially stored within a lumen when a button is in a dismounted position in accordance with the embodiments;

FIG. **12**B is a side view of the balloon inflation module of FIG. **7** illustrating the balloon being inflated when the button is in a mounted position in accordance with the embodiments;

FIGS. **13**A, **13**B, and **13**C illustrates various views of the device of FIG. **7** placed within a human ear in accordance with the embodiments;

FIG. **14**A illustrates an alternative view of the device of FIG. **7** in a mounted state in accordance with the embodiments;

FIG. **14**B illustrates an alternative view of the device of FIG. **7** in a dismounted state in accordance with the embodiments;

FIG. **15**A illustrates an internal view of a portion of the components of the device of FIG. **7** in a dismounted state in accordance with the embodiments;

FIG. **15**B illustrates an internal view of a portion of the device of FIG. **7** in a mounted state in accordance with the embodiments;

FIG. **16** is a side view of another embodiment of the device of FIG. **7** in accordance with the embodiments;

FIG. **17** is side view further illustrating the displacement volumes within a button assembly of a balloon inflation module in accordance with the embodiments;

FIG. **18** is a chart illustrating an example of volume displacements from the cavities in the button assembly of FIG. **17**;

FIG. **19**A is a frontal view of a device having a balloon inflation module and an accessory module in accordance with the embodiments;

FIG. **19**B is a perspective view of the device of FIG. **19**A in accordance with the embodiments;

FIG. **20** is a side view of the device of figure illustrating an inflated balloon when a button is in a mounted position in accordance with the embodiments;

FIG. **21** is a side view of the device of figure illustrating an involuted retracted balloon when a button is in a dismounted position in accordance with the embodiments;

FIG. **22** is a close-up view of a portion of the device of FIG. **20**; and

FIG. **23** is a frontal view of the device of FIG. **20** in accordance with the embodiments.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS OF THE PRESENT INVENTION

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

Processes, methods, materials and devices known by one of ordinary skill in the relevant arts can not be discussed in detail but are intended to be part of the enabling discussion where appropriate for example the generation and use of transfer functions.

Notice that similar reference numerals and letters refer to similar items in the following figures.

Note that herein when referring to correcting or corrections of an error (e.g., noise), a reduction of the error and/or a correction of the error is intended.

SUMMARY OF EXEMPLARY EMBODIMENTS

At least one exemplary embodiment is directed to a system for Personalized Services delivered to a Personal Audio Assistant incorporated within an earpiece (e.g., earbuds, headphones). Personalized Services include content such as music files (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech, speech-to-text processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction, academic instruction, time and date information, speech-to-speech delivery, instructions from a GPS system and others. A Personal Audio Assistant can be an audio playback platform for providing the user with Personalized Services.

At least one exemplary embodiment is directed to a Personal Audio Assistant system that is included as part of an earpiece (e.g., Headphone system). The Personal Audio Assistant is capable of digital audio playback, mitigating the need to carry a personal music player. Furthermore, a subscription-based service provides audio content to the user through the Personal Audio Assistant. The type of audio content, which is automatically provided to the user, is based on the user's preferences, which are obtained through a registration process.

The audio content, which is seamlessly downloaded to the Personal Audio Assistant in the background, is managed from a Server system and is only available on the Personal Audio Assistant for a predetermined period of time or for a fixed number of playback counts. However, the user can purchase any music file or electronic book directly from the Personal Audio Assistant with a simple one-click control interface, storing the purchased audio content on the Personal Audio Assistant as well as storing the content permanently in a user storage lock-box location on the Server system.

The system provides for audio content to be new and "fresh" each time the user auditions the content. As such, the content is typically auditioned in a first-in: first-out scenario. In one such example, the user has turned on the Personal Audio Assistant at 8:00 am and by 10:00 am has auditioned 2 hours of content that were created for the user as a manifestation of the user's choices of their preferences of genre, artist, their demographics, day of the week, time of day and purchase history. The system also provides for the elimination of a particular song or playlist in situ.

As the user's Listening History Envelope is updated based on experience, subsequent downloads will only contain content incorporating these revised preferences. The Personal Audio Assistant provides for ample memory, thus permitting hours of uninterrupted playback without the need to download additional content from the server. When in need, the Personal Audio Assistant automatically interrogates various communication platforms as it searches for connections. Once a connection is made, the Listener History Envelope file is uploaded to the server, and a new set of personalized playlist content is downloaded to the Personal Audio Assistant. Accordingly, as the Personal Audio Assistant content is auditioned and thus depleted, the communications system provides for constant replenishment.

In another embodiment, the Personal Audio Assistant also provides for a new set of business solutions to be offered to the music industry. As the personalized audio content is only available for audition for a limited period of time, and can not be sent to the user again from for weeks to months, the user's purchasing behavior can be demonstrated as spontaneous. The basic model of "Try before you buy" is the expected outcome. In another iteration, the distributor of the music can choose to offer discounts, which can be time-sensitive or quantity-sensitive in nature, in effect promoting greater purchase activity from the user.

In another iteration, while in audition a user can wish to place the desired content in a hold status. The hold status forms the basis of a "wish list," thus allowing the user to hold for future consideration audio content while it is being auditioned. This content resides in the memory of the Personal Audio Assistant for a defined period, and is automatically erased, or the user can do so manually. The selected content will also appear on the user's computer via a URL address; here it resides on the server ready for audition or purchase and download.

The system is designed to operate as simply as possible. Using a single button, which has multiple contacts, the interface allows the user to purchase, delete, skip to next, and add to a wish list and even control a listening level.

In another iteration, the user can download their own music to the Personal Audio Assistant for audition. The Personal Audio Assistant system is capable of text-to-speech processing and can interface with personal scheduling software to provide auditory schedule reminders for the user. Auditory reminders relating to the user's medication schedule are also generated by the system.

At least one exemplary embodiment includes input Acoustic Transducers (microphones) for capturing user's speech as well as Environmental Audio. In further embodiments, stereo input Acoustic Transducers capture Environmental Audio, and, mixing it with the audio signal path, present the ambient sound field to the user, mitigating the need to remove the Headphone apparatus for normal conversation.

Additional exemplary embodiments are directed to various scenarios for the delivery and consumption of audio content. The Personal Audio Assistant can store and play back audio content in compressed digital audio formats. In one embodiment, the storage memory of the Personal Audio Assistant is completely closed to the end-user and controlled from the Server. This allows for audio content to be distributed on a temporary basis, as part of a subscription service. In another iteration of the present invention, the storage memory of the Personal Audio Assistant is not completely closed to the end-user, allowing the user to transfer audio

content to the Personal Audio Assistant from any capable device such as a Personal Computer or a Personal Music Player.

In at least one exemplary embodiment the Personal Audio Assistant automatically scans for other Bluetooth®-enabled audio playback systems and notifies the user that additional devices are available. These additional devices can include a Bluetooth® video system, television system, personal video player, video camera, cell phone, another Personal Audio Assistant and others.

In another iteration, the Personal Audio Assistant can be directly connected to a Terrestrial Radio receiver or have such a receiver built in to the system.

In another exemplary embodiment, a technique known as Sonification can be used to convey statistical or other numerical information to a headphone. For example, the user would be able to receive information about the growth or decline of a particular stock, groups of stocks or even sectors of the markets though the Personal Audio Assistant. Many different components can be altered to change the user's perception of the sound, and in turn, their perception of the underlying information being portrayed. An increase or decrease in some level of share price or trading levels can be presented to the user. A stock market price can be portrayed by an increase in the frequency of a sine tone as the stock price rose, and a decline in frequency as it fell. To allow the user to determine that more than one stock was being portrayed, different timbres and spatial locations might be used for the different stocks, or they can be played to the user from different points in space, for example, through different sides of their headphones. The user can act upon this auditory information and use the controls built-in to the headphone to either purchase or sell a particular stock position.

Furthermore, specific sonification techniques and preferences can be presented to the user as "themes" from which the user can select. For example, one theme might auralize the current trading price of one stock with an ambient sine tone in the left ear, the price of another stock in the right ear, their respective trade volumes as perceived elevation using personalized head-related transfer function binauralization, and the current global index or other market indicator as the combined perceptual loudness of both tones. Such a scheme affords ambient auditory display in this example of five dimensions of financial data without compromising the user's ability to converse or work on other tasks. In another embodiment, the system affords users the ability to customize themes to their liking and to rapidly switch among them using simple speech commands. Additionally, the user can search the web from voice commands and receive results via a text to speech synthesizer.

In yet another exemplary embodiment the Personal Audio Assistant (PAA) functions as a dictation device for medical professionals for dictating clinical information to a patient's medical record or write prescriptions for medication or devices. Conversely, the PAA can function as text-to-speech allowing the clinician to audition information from a medical record, rather than reading. The PAA can save considerable time preparing clinician interaction with a patient.

In another iteration, the Personal Audio Assistant can function as a tool to locate other users of Personal Audio Assistant who share common interests, or who are searching for particular attributes of other users. Whereas the first user has stored specific personal information in the Public Data memory of the Personal Audio Assistant, an example of which might be related to schools attended, marital status, profession etc, or the first user can be in search of another user with these attributes and whereas a second user of a Personal Audio Assistant comes within communication range of the first user, the individual Personal Audio Assistants communicate with each other, access the personal information stored in each of their respective Public Data memories to ascertain if these users have common interests. If a match occurs, each unit can contain both audible and visual indicators announcing that a match has been made and thus each user can start dialog either physically or electronically via the environmental microphones.

Examples of Terminology

Note that the following non-limiting examples of terminology are solely intended to aid in understanding various exemplary embodiments and are not intended to be restrictive of the meaning of terms nor all inclusive.

Acoustic Isolation Cushion: An "Acoustic Isolation Cushion" shall be defined as a circum-aural or intra-aural device that provides acoustic isolation from Environmental Noise. Acoustic Isolation Cushions can be included as part of a Headphones system, allowing the output of the acoustical transducers to reach the ear unimpeded, but still providing acoustic isolation from Environmental Noise.

Acoustic Transducer: An "Acoustic Transducer" shall be defined as a device that converts sound pressure level variations into electronic voltages or vice versa. Acoustic Transducers include microphones, loudspeakers, Headphones, and other devices.

Audio Playback: "Audio Playback" shall be defined as the auditory stimuli generated when Playback Hardware reproduces audio content (music, speech, etc) for a listener or a group of listeners listening to Headphones.

Audition: "Audition" shall be defined as the process of detecting sound stimulus using the human auditory system. This includes the physical, psychophysical, psychoacoustic, and cognitive processes associated with the perception of acoustic stimuli.

Client: A "Client" shall be defined as a system that communicates with a Server, usually over a communications network, and directly interfaces with a user. Examples of Client systems include personal computers and mobile phones.

Communications Port: A Communication Port shall be defined as an interface port supporting bidirectional transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others) via a communications network (e.g., the Internet, cellular networks).

Control Data: "Control Data" shall be defined as information that dictates the operating parameters for a system or a set of systems.

Earcon: An Earcon shall be defined as a Personalized Audio signal that informs the User of a pending event typically inserted in advance of the upcoming audio content.

Ear Mold Style: "Ear Mold Style" shall be defined as a description of the form factor for an intra-aural device (e.g., hearing aids). Ear Mold Styles include completely in the canal (CIC), in the canal (ITC), in the ear (ITE), and behind the ear (BTE).

Environmental Audio: "Environmental Audio" shall be defined as auditory stimuli of interest to the user in the environment where the user is present. Environmental Audio includes speech and music in the environment.

Environmental Noise: "Environmental Noise" shall be defined as the auditory stimuli inherent to a particular environment where the user is present and which the user does not wish to audition. The drone of highway traffic is a

common example of Environmental Noise. Note that Environmental Noise and Audio Playback are two distinct types of auditory stimuli. Environmental Noise does not typically include Music or other audio content.

E-Tailing System: An "E-tailing System" shall be defined as a web-based solution through which a user can search, preview and acquire some available product or service. Short for "electronic retailing," E-tailing is the offering of retail goods or services on the Internet. Used in Internet discussions as early as 1995, the term E-tailing seems an almost inevitable addition to e-mail, e-business, and e-commerce. E-tailing is synonymous with business-to-consumer (B2C) transactions. Accordingly, the user can be required to register by submitting personal information, and the user can be required to provide payment in the form of Currency or other consideration in exchange for the product or service. Optionally, a sponsor can bear the cost of compensating the E-tailer, while the user would receive the product or service.

Generic HRTF: A "Generic HRTF" shall be defined as a set of HRTF data that is intended for use by any Member. A Generic HRTF can provide a generalized model of the parts of the human anatomy relevant to audition and localization, or simply a model of the anatomy of an individual other than the Member. The application of Generic HRTF data to Audio Content provides the least convincing Spatial Image for the Member, relative to Semi-Personalized and Personalized HRTF data. Generic HRTF data is generally retrieved from publicly available databases such as the CIPIC HRTF database.

Headphones: "Headphones" (also known as earphones, earbuds, stereophones, headsets, Canalphones, or the slang term "cans") are a pair of transducers that receive an electrical signal from a media player, communication receivers and transceivers, and use speakers placed in close proximity to the ears (hence the name earphone) to convert the signal into audible sound waves. Headphones are intended as personal listening devices that are placed either circum-aural or intra-aural according to one of the Ear Mold Styles, as well as other devices that meet the above definition such as advanced eyewear that includes Acoustical Transducers (i.e. Dataview). Headphones can also include stereo input Acoustic Transducers (microphones) included as part of the Ear Mold Style form factor.

HRTF: "HRTF" is an acronym for head-related transfer function—a set of data that describes the acoustical reflection characteristics of an individual's anatomy relevant to audition. Although in practice they are distinct (but directly related), this definition of HRTF encompasses the head-related impulse response (EMIR) or any other set of data that describes some aspects of an individual's anatomy relevant to audition.

Informed Consent: "Informed Consent" shall be defined as a legal condition whereby a person can provide formal consent based upon an appreciation and understanding of the facts and implications associated with a specific action. For minors or individuals without complete possession of their faculties, Informed Consent includes the formal consent of a parent or guardian.

Listening History Envelope: "Listening History Envelope" shall be defined as a record of a user's listening habits over time. The envelope includes system data, time system was turned off, time the system is presenting content, when the system doesn't audition, system transducers, when the user auditions content, time stamp of content being auditioned, content which is: skipped, deleted, played multiple times, saved in the Wish List, and time between listening sessions.

Music: "Music" shall be defined as a form of expression in the medium of time using the structures of tones and silence to create complex forms in time through construction of patterns and combinations of natural stimuli, principally sound. Music can also be referred to as audio media or audio content.

Playback Hardware: Any device used to play previously recorded or live streaming audio. Playback Hardware includes Headphones, loudspeakers, personal music players, mobile phones, and other devices.

Personal Audio Assistant: A "Personal Audio Assistant" shall be defined as a portable system capable of interfacing with a communications network, directly or through an intermediate, to transmit and receive audio signals and other data.

Personal Computer: "Personal Computer" shall be defined as any piece of hardware that is an open system capable of compiling, linking, and executing a programming language (such as CIC++, Java™, etc.).

Personal Music Player: "Personal Music Player" shall be defined as any portable device that implements perceptual audio decoder technology but is a closed system in that users are not generally allowed or able to write software for the device.

Personalized HRTF: A "Personalized HRTF" shall be defined as a set of HRTF data that is measured for a specific Member and unique to that Member. The application of Personalized HRTF data to Audio Content creates, by far, the most convincing Spatial Image for the Member (Begault et. al. 2001, D. Zotkin, R. Duraiswami, and L. Davis 2002).

Personalized Services: "Personalized Services" shall be defined as services customized to better meet the needs of an individual. Personalized Services include media content (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction [real-time foreign language translation], academic instruction, time and date information, and others.

Public Data: "Public Data" shall be defined as data which contains specific and personal information about the registered user of the Personal Audio Assistant. The registered user chooses which portions of their complete Registration Process data they wish to include in this subset. This data becomes distributed to other users who have compliant devices thus allowing other users to know specific details of the registered user.

Registration Process: "Registration Process" includes the acquisition of the user's preference via a web page. Typically, the process would include the items to be captured: Age, demographics, email, gender, Relative Audiogram, Personal Preferences, banking information, credit card information, wake-up and sleep times, music preferences by genre, artist, preferences for writers and authors, desire to receive advertising, turn-on listening level, equalization, email preferences, parental control setup as well as other user-controlled settings.

Relative Audiogram: A "Relative Audiogram" shall be defined as a measured set of data describing a specific individual's hearing threshold level as a function of frequency. A Relative Audiogram is only an approximate Audiogram, leaving more complete Audiogram analysis to qualified audiologists.

Semi-Personalized HRTF: A "Semi-Personalized HRTF" shall be defined as a set of HRTF data that is selected from a database of known HRTF data as the "best-fit" for a

specific user. Semi-Personalized HRTF data is not necessarily unique to one user; however, interpolation and matching algorithms can be employed to modify HRTF data from the database to improve the accuracy of a Semi-Personalized HRTF. The application of Semi-Personalized HRTF data to Audio Content provides a Spatial Image that is improved compared to that of Generic HRTF data, but less effective than that of Personalized HRTF data. The embodiments within speak to a variety of methods for determining the best-fit HRTF data for a particular Member including anthropometrical measurements extracted from photographs and deduction.

Server: A "Server" shall be defined as a system that controls centrally held data and communicates with Clients.

Sonification: "Sonification" shall be defined as the use of non-speech audio to convey information or to aurally perceptualize non-acoustic data (auralize). Due to a variety of phenomena involving human cognition, certain types of information can be better or more efficiently conveyed using auditory means than, for example, visual means.

Exemplary Embodiments

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network (**101**) via communication interface **102**, which can be operatively connected (via a wired or wireless connection) to a server system (**100**) and/or an e-mail server (**105**). Additionally a radio signal (e.g., satellite radio) can be input into the earpiece **500** (FIG. **5**B) via a communication module (e.g., Bluetooth® wireless module **515**).

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity). For example multiple users (e.g., **202**, **203**, **204**, and **206**) can send signals to each individual earpiece (e.g., **500**) when in range (e.g., via a wireless connection **205**) or to a mobile audio communications device **200** via a wireless connection (**201**) with each earpiece (**500**). Additionally information (e.g., audio content, software download) can be sent via a client's computer **207** to each earpiece, either directly (e.g., **205**), or via **200**. For example audio content can be retrieved on a user's computer and sent to the earpieces that have authorization to use it.

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment. The earpiece can include all or some of the elements illustrated in FIG. **5**B. For example the logic circuit **572** or the operatively connected memory storage device **585**, can include spatial enhancement software **329**, a DSP codec **330**, a speech synthesis and recognition system **311**, and a digital timer system **312**. Additional elements can be connected to the logic circuit **572** as needed, for example a power supply **320**, a software communication interface **307** (e.g., wireless module **515**) (which may be connected to communication network **308**), data port interface **306**, audio input buffers **300** connected to digital audio input **302** and/or analog audio input **303** converted to digital via an ADC **301**, environmental audio input acoustic transducer(s) **321** converted to digital via an ADC **316**, user control **324**, digital audio output **328**, output acoustic transducers **319** (which receive signals converted to analog via a DAC **310** and amplified via amplifier **309**), display system **318**, communication buffers **325**, program memory **305**, data and personal memory **315**, as well as other electronic devices as known by one of ordinary skill in the relevant arts.

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer **419**, via a communication network **400** (e.g., Internet connection) connected to many various database and registration systems as illustrated and labeled in FIG. **4**. For example, server **401**, database management system **402**, audio content preview database **403**, audio content database **404**, playlist engine **405**, user's information database **406**, HRTF acquisition process module **407**, HRTF database **408**, lock-box server **409**, registration engine **410**, e-payment system **413**, business-to-business module **414**, e-tailing system **415**, audiogram database **416**, audiogram engine **417** and/or e-mail server **418**.

FIG. **5**A illustrates an earpiece **500** having sealing section **540** in an ear canal that can store and download audio content **560** in accordance with at least one exemplary embodiment. The earpiece **500**, can include a first user interaction element **530** (e.g., a button), that can be used to turn the earpiece **500** on, or if on then activate an audio play command to start playing saved audio content. The earpiece **500** can also include a second user interaction element **550** (e.g., a slide control) that can be used for example to control the volume. The earpiece **500** can also include recharge ports **570**, that can accept two wires of varying voltage that can be inserted into the recharge ports **570** to recharge any batteries in the earpiece **500**. The earpiece **500** can include an ambient microphone **520** and an optional communication antenna **510**, that if needed can aid in the communication between the earpiece **500** and a communication network.

FIG. **5**B illustrates a block diagram of the earpiece of FIG. **5**A, illustrating the first user interaction element **530**, the ambient microphone (AM) **520**, that can be used to pick up ambient audio content, an ear canal microphone (ECM) **590** that can pick up audio in the ear canal region, an ear canal receiver (ECR) **580** that can direct audio content to the ear drum, all of which can be connected operatively to a logic circuit **572**. A memory storage device **585** can be operatively connected to the logic circuit (LC) **572**, and can store data such as registration, preference, and audio content data. The optional communication antenna **510** can be connected to a communication module (e.g., wireless module **515**), and can receive or transmit information **560** to a communication network.

FIG. **6** illustrates a user interface for setting the parameters stored in the memory storage device **585**. For example a user can use his/her computer **419** to communicate with a server **401** (e.g., via a communication network **400**) to start the user's registration (e.g., with an audio content provider). The registration information can then be transmitted **600** to set the stored parameters in the memory storage device **585** of the earpiece **500**. Additionally, a requested (e.g., bought) audio content can be downloaded **610** into the memory storage device **585** of the earpiece **500**.

At least one exemplary embodiment is directed to an earpiece comprising: an ambient microphone; an ear canal microphone; an ear canal receiver; a sealing section; a logic circuit; a communication module; a memory storage unit; and a user interaction element, where the user interaction element is configured to send a play command to the logic circuit when activated by a user where the logic circuit reads registration parameters stored on the memory storage unit and sends audio content to the ear canal receiver according to the registration parameters.

In at least one exemplary embodiment the audio content is stored in the memory storage unit. The earpiece according to claim **2**, where the communications module is a wireless

communications module. Additionally, the earpiece can include a second user interaction element configured to alter the volume of the audio content that is emitted from the ear canal receiver.

Upon a play command being received by the logic circuit the logic circuit can check registration parameters stored in the memory storage device for example one of the registration parameters can be whether the audio content is a sample audio content or a fully purchased audio content, or the allowed number of times an audio content can be played, and a counter value that keeps track of the number of times the audio content has been played.

The earpiece can send an auditory warning to be emitted by the ear canal receiver when the counter value is greater than or equal to the allowed number of times the audio content can be played, and where the logic circuit does not send the audio content to the ear canal receiver.

Further Exemplary Embodiments

At least one exemplary embodiment is directed to a system for the delivery of Personalized Services to Personal Audio Assistants, the system comprising: a Personal Audio Assistant system for presenting Personalized Services to the user as Audio Playback; a Server system for user registration, Personalized Service management, and communication; a Registration Process for collecting detailed registration information from users, including the information necessary for creating Personalized Services; a communications protocol (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to the Server or connecting the Personal Audio Assistant to other Personal Audio Assistants (peer-to-peer behavior).

In at least one exemplary embodiment a Personal Computer acts as an intermediate, connecting to the Server system over a communications network and connecting to the Personal Audio Assistant over a local connection. At least one exemplary embodiment includes a Personal Hearing Damage Intervention System (e.g., USPTO—60/805,985—Goldstein).

In at least one exemplary embodiment a Personal Audio Assistant system included as part of a Headphone system, the system comprising: a Communications Port supporting a communications protocol enabling communication with the Server system, peer devices, and other capable devices; a non-volatile program memory storage system for storing Control Data, dictating system behavior; a data memory storage system for storing data and audio content; an analog audio input/output and corresponding ADC/DAC; a digital audio input/output and a digital audio signal path; a user control system allowing the user to adjust the level of the audio output and control the behavior of the system; a user control system allowing the user to purchase the content being auditioned in real time; a user control system allowing the user to control, delete, fast forward, output level control, scan, advance, the data stored both stored in memory as well as new streaming data emails and reminders; a display system for presenting information to the user(s) visually using any method familiar to those skilled in the art (LED, LCD, or other); a display system for presenting information to the user(s) (e.g., using Earcons and other sound files); a speech synthesis system for converting text-to-speech and generating speech signals; a speech recognition system for converting speech to-text to respond and send emails and to interface with the control language as to provide navigational commands; a digital timer system; a power supply system in the form of a battery; a unique identification number for each Personal Audio Assistant; Input Acoustic Transducers; an Output Acoustic Transducer; an Audio amplification system; Acoustic Isolation Cushions conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE; see definitions) and other elements common to Headphone systems; a digital signal processor (DSP) system; and a CODEC processor capable of improving the perceptual sound quality of the content to be auditioned while governed by delivering the correct SPL dose.

In at least one exemplary embodiment the system is independent of a Headphone array or can be included and imbedded as part of a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the various processing needed to derive the intended functions are distributed among any combination of a Server system, a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the Personal Audio Assistant system can exchange audio signals with a mobile phone via the Communications Port, allowing the Personal Audio Assistant to function as a mobile phone accessory.

In at least one exemplary embodiment a communications buffer is included. For example when a network connection is available, the communications buffer uploads stored content (e.g., Listening Habits Envelope) and stores incoming transmissions (e.g., music, electronic books, and updates to the firmware or operating system) from the Communications Port; The contents of the communications buffer are then transmitted whenever a network connection becomes available. At least one exemplary embodiment includes a perceptual audio codec decoding technology in the DSP, enabling the storage and playback of compressed digital audio formats (e.g., MP3, AAC, FLAC, etc.). At least one exemplary embodiment is compliant and compatible with DRM, FairPlay® and other forms of digital content governance.

At least one exemplary embodiment includes a user control system for selecting and playing back audio content stored in memory that operates using any combination of the following methods: a button or tactile interface which upon auditioning a song can be pressed to order content; a button, tactile and/or voice controlled interface which, when pressed once, commanded to, activates playback of short audio clips or audio thumbnails of the audio content stored in memory; When the button is pressed again during audio thumbnail playback, the current audio content selection is played in its entirety; The behavior of this interface is similar to the "scan" button interface common in FM/AM radio devices; a button, tactile and/or voice controlled interface that, when pressed or commanded to, skips to the next piece of audio content, which is selected randomly from all available audio content that has a play count equal to or less than the play count of the piece of audio content currently playing; The behavior of this interface is similar to the "shuffle" behavior found in some personal music players; an interface for

browsing audio content storage devices familiar to those skilled in the art; and a process to allow for increased data memory storage capacity for storing audio content.

In at least one exemplary embodiment the contents of the data memory are encrypted and controlled by the Server system only, prohibiting the end-user from loading unauthorized audio content into the data memory. Further the contents of the data memory can be manipulated by the end-user, allowing the user to transfer audio content to the Personal Audio Assistant system from any device capable of interfacing with the communications port; For example, audio content can be transferred to the system from a Personal Music Player or a Personal Computer. According to at least one exemplary embodiment, audio content (or other media content) updates are retrieved from the Server system any time a connection is detected by the communications port. Furthermore, an exemplary embodiment can include an acoustical and/or visual indicator informing the user when a transfer of data is activated.

In at least one exemplary embodiment radio wave transmissions are used to implement some communications protocol and the communications port acts as a radio receiver. Additionally the Personal Audio Assistant can include: interfaces with some personal scheduling software through the communications port; a speech synthesis system which generates speech-signal reminders corresponding to information from the scheduling software, where the digital timer system triggers the presentation of the speech-signal reminders at the appropriate time.

Additionally the Personal Audio Assistant can interface with an email platform through the communications port; The speech synthesis system converts the email in text to speech and provides email to the user in aural presentation format. The system further comprising: a process in the Registration engine allowing the user to optimize their personalization process of incoming emails by associating a specific Earcon with the importance of the incoming email. As such, normal priority email contains an introduction sound announcing to the user the level of importance the sender associated with their email; a speech recognition system for converting speech-to-text which interfaces with the control language as to provide navigational commands allowing the user to respond and send emails.

In at least one exemplary embodiment the communications port system makes use of some wireless communications protocol (802.11, Bluetooth®, A2DP, or other) to transmit and receive digital audio data for playback, the system further comprising: an audio codec to encode and decode digital audio transmissions; a wireless communications system (802.11, Bluetooth®, A2DP, etc.) for transmitting and receiving data (digital audio transmissions, Control Data, etc.); a method for pairing two or more Personal Audio Assistants through a wireless communications protocol to provide a secure exchange of audio content, data such as the user's Public Data; an audio warning signal or visual display system output that notifies the user anytime a compatible transmission becomes available; and a user control system enabling the user to switch between available compatible transmissions.

In at least one exemplary embodiment the system enables listeners to share digital audio transmissions, the system further comprising: a method for scanning for available digital audio transmissions within range; a user control interface for specifying digital audio transmission behavior; a method for employing the system as a relay to other compliant devices; re-broadcasting digital audio transmissions to increase wireless range. In at least one exemplary embodiment multiple systems are capable of sharing the contents of their program and data memory using the wireless communications protocol.

In at least one exemplary embodiment of the system, the input Acoustic Transducer is used to record audio content to the data memory storage system, the system further comprising: an implementation of some perceptual audio codec technology in the DSP, enabling the storage of compressed audio formats (e.g., MP3, AAC, FLAC, etc); and an increased data memory storage capacity for storing recorded audio content.

In at least one exemplary embodiment, the stereo input Acoustic Transducers are ultimately connected to the audio signal path at the DSP, allowing the user to audition Environmental Audio (e.g., speech or music) and mitigating the need for the user to remove the Headphone apparatus to audition Environmental Audio, the system further comprising: a stereo pair of input Acoustic Transducers placed close to the user's ear canal input, conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE, see definitions); and by mounting the input Acoustic Transducers in a CIC or ITC configuration, spatial-acoustic cues are preserved, creating a spatially-accurate Environmental Audio input signal—essentially a personal binaural recording; a method for acoustically compensating for the non-linear frequency response characteristics of the Acoustical Isolation Cushions of a given Headphone system by applying corresponding inverse filters to the Environmental Audio input signal at the DSP; With this method, the system acts as a linear-frequency-response hearing protection apparatus (e.g., USPTO—60/805,985—Goldstein).

At least one exemplary embodiment includes a system for first attenuating Audio Playback and then mixing the Environmental Audio input signals, at a louder listening level, with the audio signal path using the DSP, where the system is activated by any combination of the following methods: a manual switch to activate/deactivate the system; a speech-detection apparatus to activate the system when speech is detected as the principal component of the Environmental Audio input; and a music-detection apparatus to activate the system when music is detected as the principal component of the Environmental Audio input.

At least one exemplary embodiment can include active noise reduction, echo cancellation and signal conditioning that can be environmentally customized through the registration process to better meet the user's specific needs (i.e., occupation-related noise cancellation); A typical application would be a special set of noise cancellation parameters tuned to the drilling equipment used by a dentist.

In at least one exemplary embodiment the input Acoustic Transducers are instead mounted within circum-aural, intra-aural BTE, or intra-aural ITE molds (see Ear Mold Style), the system further comprising: a spatial audio enhancement system for supplementing the spatial-acoustic cues captured by the stereo pair of input Acoustical Transducers to provide improved spatial perception of Environmental Audio using any combination of the following methods: the application of Generic, Semi-Personalized, or Personalized HRTF data to the Environmental Audio input signal; the application of binaural enhancement algorithms, familiar to those skilled in the art, to the Environmental Audio input signals; the application of a pinna simulation algorithm to the Environmental Audio input signal; and a synthetic pinna apparatus placed just before the stereo input Acoustic Transducers.

At least one exemplary embodiment includes a Server system for the creation, Registration, management, and delivery of Personalized Services, the system comprising: a

communications system for interfacing with public communication networks to exchange data with Personal Audio Assistants, a Client's computer, mobile phones, PDAs or other capable devices; a database and database management system for storing and retrieving information relating to user Registration, Personalized Services, audio content, Control Data, and other data; a Registration interface system for collecting, storing, and applying information provided by users; a method for creating Personalized Services based on user Registration information; an end-user audio content Lock-Box storage system, providing every registered user access to their purchased media content; a business-to-business interface system for acquiring audio content with record labels, copyright holders, and other businesses; an E-tailing system including an electronic transactions system enabling users to purchase content, items offered for sale or pay subscription fees electronically; an E-Payment system compensating the various copyholders upon purchase of content by user; a Playlist engine, which acquires the user's Registration information, Listening History Envelope and then creates audio playlists, which is optimized for the user preferences and further refinements; and an Email server, which distributes communications to the user and others, regarding marketing data, the status of the user weekly SPL dose, and other information.

At least one exemplary embodiment includes machine-learning techniques employed to better optimize the user's preferences relating to audio content and other media content, the system further comprising: a method for tracking the purchase history of each user, relating the purchase history to media content preferences, and using the purchase history to make media content recommendations; a method for examining a user's digital media library, stored on a Personal Computer, Personal Music Player, or Personal Audio Assistant, from the Server system, and relating media content preferences and media content recommendations to the user's digital media library; and a method for examining a user's Listening History Profile.

At least one exemplary embodiment includes a Registration system for collecting a wide variety of information from users, including information necessary for creating Personalized Services, the system comprising: a Server system; an interface system for querying the user to collect registration information including demographics (age, gender), Playback Hardware information, Headphone information, occupational information, home and work locations, medication information, music-related preferences, video-related preferences, and other information; a method for customizing Control Data based on registration information; and a method for creating Personalized Services based on registration information.

In at least one exemplary embodiment a fast HRTF acquisition process is included as part of the Registration process, the system further comprising a method for the fast acquisition of Semi-Personalized HRTF data via a deduction process, the method comprising: a database system containing indexed, clustered HRTF data sets; an auditory test signal with distinctive spatial characteristics, where two or more distinct sound source locations exist; a system for the application of potential HRTF matches to the auditory test signal; and a feedback system, allowing the user to select the best listening experience from a number of candidate listening experiences, based on the spatial quality perceived in the HRTF-processed auditory test signal.

In at least one exemplary embodiment Personalized HRTF data is measured and used instead of Semi-Personalized HRTF data, by any method familiar to those skilled in the art.

In at least one exemplary embodiment the user is provided some Personal Audio Assistant free-of-charge or at a discount, given the user agrees to a subscription service commitment to receive Personalized Services for a certain amount of time.

In at least one exemplary embodiment, as part of the Personalized Services, the user is provided with temporary audio content corresponding to the preferences indicated during the registration process; Further, the user is given the option to purchase the audio content permanently; Otherwise, the audio content is replaced with new audio content from the Server, after a predetermined amount of time or a predetermined number of playback counts, the system comprising: a Personal Audio Assistant with an enhanced user control system, enabling a registered user to purchase media content directly from the Personal Audio Assistant with a button; and a Personal Audio Assistant with an enhanced user control system, enabling a registered user to store a reference to media content that can be purchased by the user at a later time.

In at least one exemplary embodiment, video or gaming content is included as well as audio content, the system further comprising: a Personal Audio Assistant with an enhanced visual display system, capable of playing video and/or gaming content.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives medication reminders in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; Medication reminders are generated by the Server system based on the user's registration information.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives stock market information in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; The stock market information is selected by the Server system based on the user's registration information, the system further comprising: the user having successfully registered their Personal Audio Assistant with a brokerage firm, or other stock trading engines, the user can then purchase or sell a stock by use of a user button or a speech command.

Further in at least one exemplary embodiment, the user is able to request specific media content to be transferred temporarily or permanently to the user's Personal Audio Assistant, the system further comprising: an interface system operating on the Server allowing users to request specific media content by artist, title, genre, format, keyword search, or other methods familiar to those skilled in the art; and a media content search engine system.

In at least one exemplary embodiment a Relative Audiogram compensation filter is applied to audio signal path by the digital signal processor, the system either (e.g., USPTO—60/805,985—Goldstein): (a) Retrieves Relative Audiogram compensation information from a remote Server after a registration process (during transmission, the information can include HIPAA compliant encoding); or (b) calculates a compensation filter from Relative Audiogram information obtained by the system locally. For example, U.S. Pat. No. 6,840,908—Edwards, and U.S. Pat. No. 6,379,314—Horn, discuss methods for the acquisition of an individual's Relative Audiogram.

In at least one exemplary embodiment a Satellite Radio transmitter/receiver (transceiver) is incorporated within the

Headphone proper, allowing the user to at least: receive XM®, Sirius® and other broadcasts for playback over the system; select radio stations for playback over the system via the control system, the control system comprising either a single-click tactile interface or the speech-controlled circuitry; store selected portions of such broadcasts in memory for later recall and playback via the control systems; engage a novel commercial-skip feature for attenuating the playback level of suspected sales commercials broadcasts; and engage a speech-skip feature for attenuating the playback of speech (e.g., news, announcements, etc.).

At least one exemplary embodiment includes a Walkie-Talkie mode, which broadcasts input to the system's built-in microphone, whereby the user's speech can be detected by the input acoustic transducer and remotely broadcast where at least one of the following occurs: the Walkie-Talkie mode receives input via AM/FM broadcasts (as well as digital communications protocols) from a nearby user; the Walkie-Talkie mode allows nearby users to engage in conversation with increased perceptual clarity in noisy environments (e.g., aircraft cockpits), using for example a noise-cancellation system; selectively engage and disengage the Walkie-Talkie mode using the control system; detect other users of the system within a given range; and alert the user of the system when other detected systems contain certain Public Data and contain a predefined Public Message Key (e.g., "If the detected system belongs to a single male between the ages of 25 and 30 and whose favorite sport is tennis, then broadcast the message, 'I like tennis also; would you like to have coffee?'" or "If the detected system belongs to a user who attended Princeton University, then broadcast the message, 'Go Tigers!'").

At least one exemplary embodiment can use other communications to accomplish this service rather than AM/FM; as such the system can incorporate communications transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other protocols) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to other Personal Audio Assistants. At least one exemplary embodiment can selectively control the broadcast of public data and public message keys via the control system.

At least one exemplary embodiment includes a Sonification algorithm within the Headphone, which enables auditory display of digitally received data, including for example financial data, news, GPS data, the system further containing a variety of sonification "themes" selected during the registration process that map requested data (e.g., current average trading price of AAPL stock, the Dow Jones Industrial Index, and the Nasdaq Composite®) to corresponding audio content (e.g., the frequency of a sine tone presented in the left ear, the frequency of a sine tone presented in the right ear, and the global amplitude of both sine tones, respectively).

At least one exemplary embodiment includes an auditory display, which is synthesized by the onboard Digital Signal Processor. In at least one exemplary embodiment the auditory display is created through the digital audio signal processing effects applied to any other acoustic data the system is capable of reproducing (e.g., terrestrial radio, prepurchased audio content in the user's digital library, electronic books, etc.); For example, a sudden listening level increase in the playback level of a song to which the user was listening can be triggered by a predefined alert condition (e.g., Nasdaq Composite® has exceeded 2200 points).

At least one exemplary embodiment includes the ability to create themes using a computer program and uploading a file to the Headphone system.

At least one exemplary embodiment includes a speech recognition system for converting speech to HTML (Voice Browser), whereby the user can access the Internet, provide navigational commands, perform searches and receive results via the Headphones through a text (HTML)-speech synthesize.

Additionally, the Personal Audio Assistant can be totally incorporated with a mobile cell phone, or any portable technology which incorporates any of the following protocols, TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others known to those of ordinary skill in the arts via a communications network (e.g., the Internet, cellular networks), the system further comprising: an Acoustic Transducer constructed as part of the mobile cell phone or a series of Acoustic Transducers, which are constructed as part of mobile cell phone; a commutations path incorporated into the mobile cell phone providing for bi-directional communication with a Headphone array; the incorporation of the mobile cell phone's microphone(s) to act as the Environmental Audio Acoustical Transducer(s); and the incorporation of the mobile cell phone's keyboard or touch sensitive screen to function as a manual input or to complement speech commands and that can act in a way to respond to Personalized Services offered to a user.

FIG. **7** illustrates a device **10** having a balloon inflation module **12** and an accessory module **11**. The accessory module **11** can be any number of devices that can be useful in both medical and non-medical contexts. Although the description is primarily focused on an acoustical or communication device operating with the balloon inflation module **12**, the accessory module **11** can be embodied in various diverse accessories either alone or in different combinations or permutations. For example, the accessory device can be a measuring device that can be used for measuring and/or recording pulse oximetry, blood pressure, temperature, pulse, oxygen saturation, end tidal CO2 level, gradient differentials, or acoustical impedance. In some embodiments, the accessory device **11** is one or more of a camera, a video device, a cutting tool, a laser, a radio frequency device, or a cauterization device or thermal ablation device. In some embodiments, the accessory device **11** can be used for air or gas delivery, chemical or medicine delivery, or for suction.

Accordingly, the balloon inflation module **12** and the accessory module **11** can work cooperatively in numerous contexts that can provide a tool or measuring device in an elegant, small, and protective package that can be used multiple times, yet can be disposable or replaceable. It is also reusable and retractable. The package is protective in terms of the balloon since the balloon comes in and out of an orifice or lumen where the balloon is retractable and involuted back into the orifice or lumen after or during use as needed. The balloon and/or accessory module can be deployed and used in various applications. For example, the device can be used to set or temporarily use a balloon in nasal passages to mitigate sleep apnea or SIDS for young children. In another application, the balloon can be used to set an NG or feeding tube in a nasal passage instead of using tape. In another use case, the device can be used to leverage or move tissue or can be used to manipulate, support or expand tissue.

In yet another use case, the balloon can be used inside a cannula to deliver fluid or a fluid with anesthetics first where the balloon is subsequently used as a dissecting tool to

expand the plane between two tissues, In other words, the balloon inflation module can be used anywhere for endoscopic procedures in a modality for dissecting tissue planes. The balloon inflation module can be used with fluid to distend tissue, to dissect tissue, to create true planes from virtual planes, for introduction of instruments or removal of tissue, or obstructing pathology (e.g., stones, cancer, etc). In yet another embodiment, the balloon can be coated with different substances to provide a rasp-like function to rasp surrounding tissue. For example, the balloon can be coated with diamonds fixed to the balloon. Additionally or alternatively, the surface of the balloon can be textured or non-textured, coated or covered by other surfaces so that there is an interface between the balloon and the tissue being worked on.

In yet other embodiments, the balloon inflation module can be used in conjunction with an endoscope for various procedures in a number of human or animal orifices (sinus, ear, throat, etc.). In some embodiments, the device can provide either single or multiple balloons for insertion and can otherwise provide a delivery system for multiple balloons. In some embodiments, the device can be used with or as part of a stent deployment system providing, for example, the dilatation and manipulation of tissue to place a stent in the appropriate positioning.

The balloon can be modified for various uses and the shape can be constrained for particular body vessels. It can have a unique eccentric shape. The balloon can also be accessorized with a wiper or squeegee at the periphery of the distal end of the lumen of the balloon inflation module to easily remove wax or cerumen or other accumulated materials before the balloon is retracted and involuted. A tapered edge about the periphery of the distal end can be formed to serve as a squeegee.

As noted above, the device is not limited to medical applications. In some embodiments, the device can be used in non-medical applications such as plumbing, examining conduits, oil pipeline inspection or repair. In other words, the device can be a modality to seal or repair leaks in conduits or used as part of system to diagnose and mitigate leaks in conduits. In one example, the retractable involuted balloon can deployed as part of an oil pipe inspection robot at a location of a leak or crack for repair.

Referring again to FIG. **7**, the balloon inflation module **12** can further include a button or button assembly **14** used to displace a volume of fluid within the balloon inflation module towards a balloon **16** shown in an inflated mode in FIG. **7**.

FIG. **8** illustrates an exploded view of the device **10**. The balloon inflation module **12** includes a button **14** having a detent **13**. The module **12** further includes a lumen **17** and the balloon **16**. When the button **14** is depressed or pushed towards a body cap **26**, the button **14** is placed in a mounted position, which can be fixed or locked with a latch **15** that mates with the detent **13**. Fluid is displaced from cavities within the button or button assembly **14** towards the balloon **16** via the lumen **17** or in another embodiment, via another lumen such as the pushrod **19**. The accessory module **11** can couple to the balloon inflation module at port **18**. The balloon inflation module can be part of an inflation management system (IMS) that includes an involuted balloon that is retractable and further includes an integrated reservoir for fluid. The fluid can be liquid, air, or gel for example. The accessory module **11** can be in one embodiment, an acoustic management system (AMS) having software and hardware that controls a user's acoustic experience. Among the components that can be included in the AMS are a first ambient microphone **21**, a second ambient microphone **22**, an optional ear canal microphone **23**, and an ear canal receiver **24**. The microphones **21**, **22**, **23** and the ear canal receiver **24** can be housed within a housing **20**. The AMS can further include a logic circuit coupled to the various microphones and the receiver (speaker). The ambient microphones **21** and **22** can be configured to pick up ambient audio content. The ear canal microphone **23** can be configured to pick up audio in the proximity of an ear canal, and the ear canal receiver **24** configured to provide audio content in the proximity of the ear canal. The AMS can further include a conduit **25** for delivering audio content to the AMS. Further note that the AMS and IMS are field replaceable allowing for customization or service at the time of purchase or once deployed in the field.

FIG. **9** illustrates a side view of the balloon inflation module **12** having a straight lumen that houses the balloon **16**. In addition to the components described above, the balloon inflation module **12** can include within the button assembly **14**, an a-ring **34** and a spring **36** to bias the button assembly **14** in an unmounted position. In some embodiments, the fluid **32** used within the cavities of the balloon inflation module **12** can be thermally and chemically stable and non-flammable. In medical applications, the fluid should also be non-toxic and leave essentially no residue upon evaporation. In some embodiments the balloon **16** can be attached to the outside lumen **17** while in other embodiments the balloon **16** can be attached to the pushrod **19** (or inner lumen).

FIG. **10** illustrate the side view of the balloon inflation module **12** and further illustrate a first volume within the button cavity **41** and a second volume within a body cavity **42**. In one example as shown in the chart **45** of FIG. **11**, the button cavity can have 0.142 cc in volume and the body cavity can have 0.188 cc in volume for a total volume displacement of 0.33 cc.

FIG. **12**A illustrates the balloon inflation module in a dismounted position where the balloon **16** is partially retracted within the lumen **17**. In the dismounted position, the fluid is retained within the cavities of the button assembly **14**. In FIG. **12**B, the balloon inflation module **12** is shown in a mounted position where the fluid previously in the button cavities is displaced toward and within the balloon **16** in order to inflate the balloon as shown. The pushrod **19** can be used to push or deploy the balloon outside the lumen **17**. As noted above, the balloon can be attached in one embodiment to the pushrod **19** itself where the pushrod would act as a lumen or conduit for the fluid towards the balloon. In another embodiment, the balloon can be attached to the outside lumen **17** where the fluid is displaced towards the balloon **16** either through the lumen **17** or the pushrod **19** if the pushrod is a lumen.

FIG. **13**A illustrates an averaged ear model **50** having the device **10** placed within a human ear. FIG. **13**B illustrates an ear model **51** with a similar view as FIG. **13**A, but at an angle. FIG. **13***c* further illustrates a representation of placement of the device in an ear model **52** further illustrating placement within an ear canal.

FIG. **14**A illustrates the device **10** in a mounted state and FIG. **14**B illustrates the device **10** in a dismounted state. In FIG. **14**A, the latch **15** is matted with the detent **13** of button **14** to retain the button in place. The latch **15** can be unlocked from the detent **13** to allow the button to go back to a dismounted state as shown in FIG. **14**B. In one embodiment, the profile height of the button in a mounted state can be 1 mm for example whereas the profile height of the button **14** in a dismounted state can be 5.5 mm.

FIG. **15**A illustrates some of the internal components of the balloon inflation module **12** in a dismounted state where FIG. **15**B illustrates the same components in a mounted state. For use with an ear canal, the conduit leading to the balloon can be angled for better placement within an ear canal. Thus, the body cap is attached with an angled port. In some embodiments, the button can include a concave dimple for haptic localization by the user. In some embodiments, various detents can be used, or an internal thread adjustment can be used to allow an optimal expansion for different ear canal sizes or in other contexts for different canal or orifice sizes. The O-ring **34** shown is used for body sealing. The spring **36** is an internal return spring for retrieval of the involuted balloon. Also note that the latch **15** can be spring loaded.

FIG. **16** illustrates a side view of an alternative button inflation module **60**. The module **60** includes many of the same elements as module **10** of FIG. **7** such as the button **14** with detent **13**, latch **15**, a-ring **34**, spring **36**, and cap **26**. An outer lumen **64** is bent or angled for better placement within an ear canal. The inner lumen **62** is similarly angled to match the lumen **64**. In this embodiment, the inner lumen **62** serves as a pushrod or guide wire. The balloon **16** is bonded to a distal end of the guide wire or lumen **62**. The button **14** is bonded or attached to proximal end of the guide wire or lumen **62**.

FIG. **17** illustrates the side view of the balloon inflation module **70** and further illustrate a first volume within the button cavity **71** and a second volume within a body cavity **72**. In one example as shown in the chart **75** of FIG. **18**, the button cavity can have 0.152 cc in volume and the body cavity can have 0.165 cc in volume for a total volume displacement of 0.317 cc.

FIGS. **19**A and **19**B illustrate a front view and side view respectively of a device **80** with ear canal microphone routing **85**, ear canal receiver routing **84**, and a pushrod or inner lumen **86** within a housing **82**. The lumen can be a multi-lumen tube having routing for the ear canal microphone port, the ear canal receiver port, or other ports. Insert tip or cap **83** with mounted balloon **16** and two through ports for the ECR (top) and ECM (bottom). In some embodiments, the ECR and ECM can be ribbed flexible ports that serve as bellows. Also, the button **14** can have multiple detents **88** for multiple positioning and sizing of the balloon within the ear canal.

FIG. **20** illustrates another side view of the device **80** having the ear canal microphone routing **85**, ear canal receiver routing **84**, and a pushrod or inner lumen **86**. The cap **83** with mounted balloon **16** and two through ports for the ECR (top) and ECM (bottom). The balloon can be adhered to the top of the distal surface of the rod or guide wire **86**. In some embodiments, the ECR and ECM can be ribbed flexible ports that serve as bellows. The balloon plug ***83**) can have two micro-bellow flex ports adhered as well as the retrieval nylon. The button **14** can have multiple detents **88** for multiple positioning and sizing of the balloon within the ear canal.

FIG. **21** illustrates the device **80** with the balloon **16** in a retracted involuted position. During motion the two micro-bellow flex ports contract providing a spring force allowing the balloon plug to effectively withdraw. The balloon plug **83** is retrieved by the nylon guide line (previously shown) causing the balloon to fold in on itself and be stored within the large single lumen tube. Operationally, as the button is released, the spring within the actuating button pushes the button out and pulls the nylon guide line attached.

FIG. **22** is another side view illustrating the balloon in an inflated mode. FIG. **23** illustrates the same device **80** and further illustrating the ECR port **94** and the ear canal microphone port **95**.

In some embodiments, a device includes a balloon inflation module having an involuted balloon housed within a lumen, an integrated reservoir in fluid communication with the involuted balloon configured to selectively displace a volume of fluid from the integrated reservoir into the involuted balloon and from the involuted balloon into the integrated reservoir, and an electronic package selectively coupled to the balloon inflation module. In some embodiments, the balloon inflation module or the electronic module is replaceable.

In some embodiments, the electronic module is an acoustic management module such as a logic circuit coupled to an ambient microphone. In some embodiments the electronic module is a communication module comprising a logic circuit coupled to an ambient microphone configured to pick up ambient audio content, a canal microphone configured to pick up audio in a canal, and a canal receiver. In some embodiments, the balloon inflation module further comprises a multi-lumen structure having the lumen for the involuted balloon, and at least one of a lumen for a canal microphone port or for a canal receiver port. In some embodiments, the balloon inflation module can further include a multi-lumen structure having the lumen for the involuted balloon, a ribbed lumen for a canal microphone port, and a ribbed lumen for a canal receiver port. In some embodiments, the ribbed lumen for the canal microphone port and the ribbed lumen for the canal receiver port forms bellow flex ports that contract and expand. In some embodiments, the balloon inflation module further includes a balloon plug coupled to a distal end of a pushrod and a distal end of the involuted balloon where a lateral retrieval of the pushrod causes the involuted balloon to fold in on itself. In some embodiments, the lumen comprises a sharpened tip at a distal end of the lumen.

In some embodiments, the balloon inflation module further includes a pushrod within the lumen for laterally displacing the involuted balloon towards a distal end of the lumen during inflation. In some embodiments, the pushrod is in fluid communication with the involuted balloon and provides a via between the integrated reservoir and the balloon. In some embodiments, the pushrod is attached to a distal portion of the involuted balloon. In some embodiments, the involuted balloon is attached to an outside portion of a distal end of the lumen.

In some embodiments, the balloon inflation module includes a spring loaded pump that displaces fluid to the involuted balloon in a mounted state and displaces fluid to the integrated reservoir in a dismounted state. In some embodiments, the balloon inflation module further includes a spring loaded pump latch that retains the spring loaded pump in a mounted or locked state when mated with a detent and releases the spring loaded pump in an dismounted or unlocked state.

In some embodiments, a device includes a balloon inflation module having an involuted balloon housed within a lumen, an integrated reservoir in fluid communication with the involuted balloon configured to selectively displace a volume of fluid from the integrated reservoir into the involuted balloon and from the involuted balloon into the integrated reservoir, and a port configured to receive an accessory module. In some embodiments, the accessory module is a communication module comprising a logic circuit coupled to an ambient microphone configured to pick up ambient audio content, an ear canal microphone configured to pick up audio in the proximity of an ear canal, and an ear canal receiver configured to provide audio content in the proximity of the ear canal. In some embodiments, integrated reservoir is formed within portions of a push button spring-loaded pump assembly.

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 such modifications and equivalent structures and functions.

What is claimed:

1. An earpiece comprising:

a first ambient microphone configured to generate a first microphone signal;

a second ambient microphone configured to generate a second microphone signal;

an ear canal microphone configured to generate a third microphone signal;

a speaker;

a memory that stores instructions;

a logic circuit, where the logic circuit is configured to execute the instructions to perform operations comprising:

generating a noise reduction signal using at least one of the first microphone signal or the second microphone signal or the third microphone signal or a combination of microphone signals;

receiving an audio signal;

generating a mixed audio signal by mixing the noise reduction signal with the audio signal; and

sending a modified audio signal to the speaker, where the modified audio signal includes the mixed audio signal.

2. The earpiece according to claim **1**, wherein the operations further comprise:

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

3. The earpiece according to claim **2**, wherein the operations further comprise:

generating the modified audio signal by mixing the ambient sound signal with the mixed audio signal.

4. The earpiece according to claim **1**, wherein the operations further comprise:

receiving an indication of an event.

5. The earpiece according to claim **4**, wherein the event is at least one of a scheduled meeting, an incoming phone call, an email, a voice message, or a text message.

6. The earpiece according to claim **5**, wherein the operations further comprise:

retrieving an Earcon from the memory, where the Earcon matches the event.

7. The earpiece according to claim **5**, wherein the operations further comprise:

sending the Earcon to the speaker.

8. The earpiece according to claim **1**, further comprising:

a sealing section.

9. The earpiece according to claim **8**, wherein the sealing section comprises:

a lumen; and

a balloon.

10. The earpiece according to claim **9**, wherein the lumen is operatively connected to the balloon, and wherein the lumen is configured to pass fluid from a reservoir to the balloon.

11. The earpiece according to claim **10**, further comprising:

interaction element.

12. The earpiece according to claim **11**, wherein the interaction element is a button.

13. The earpiece according to claim **12**, wherein the button is configured so that when pressed fluid is moved through the lumen into the balloon.

* * * * *

Exhibit F

US011683643B2

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Goldstein et al.

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(54) **METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION**

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CPC ***H04R 3/002*** (2013.01); ***H04R 1/1016*** (2013.01); ***H04R 25/02*** (2013.01)

(58) **Field of Classification Search**
CPC H04R 1/1016; H04R 25/02; H04R 3/002; H04R 25/453; H04R 2460/01; H04R 3/02;

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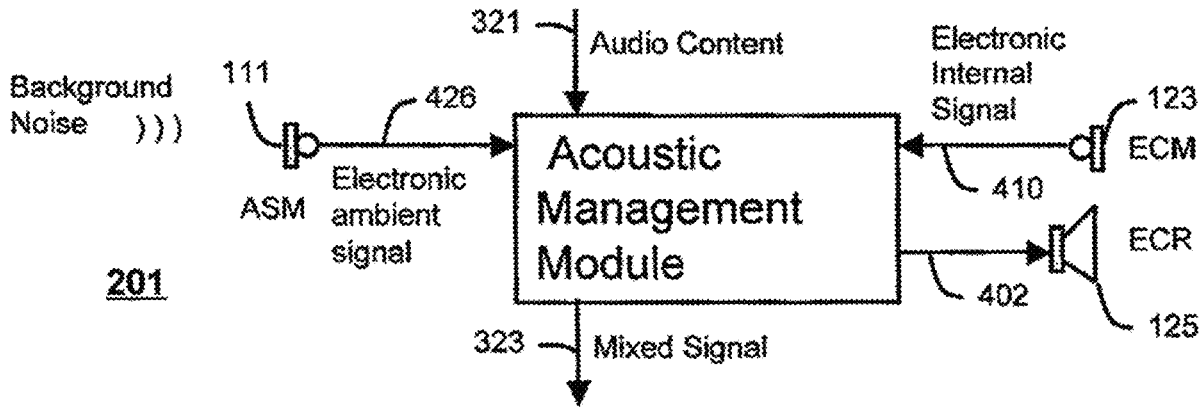
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(57) **ABSTRACT**

An earpiece (**100**) and acoustic management module (**300**) for in-ear canal echo suppression control suitable is provided. The earpiece can include an Ambient Sound Microphone (**111**) to capture ambient sound, an Ear Canal Receiver (**125**) to deliver audio content to an ear canal, an Ear Canal Microphone (**123**) configured to capture internal sound, and a processor (**121**) to generate a voice activity level (**622**) and suppress an echo of spoken voice in the electronic internal signal, and mix an electronic ambient signal with an electronic internal signal in a ratio dependent on the voice activity level and a background noise level to produce a mixed signal (**323**) that is delivered to the ear canal (**131**).

11 Claims, 10 Drawing Sheets



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USPC 381/66, 56–58, 312, 314, 74, 92, 381/94.1–94.5
See application file for complete search history.

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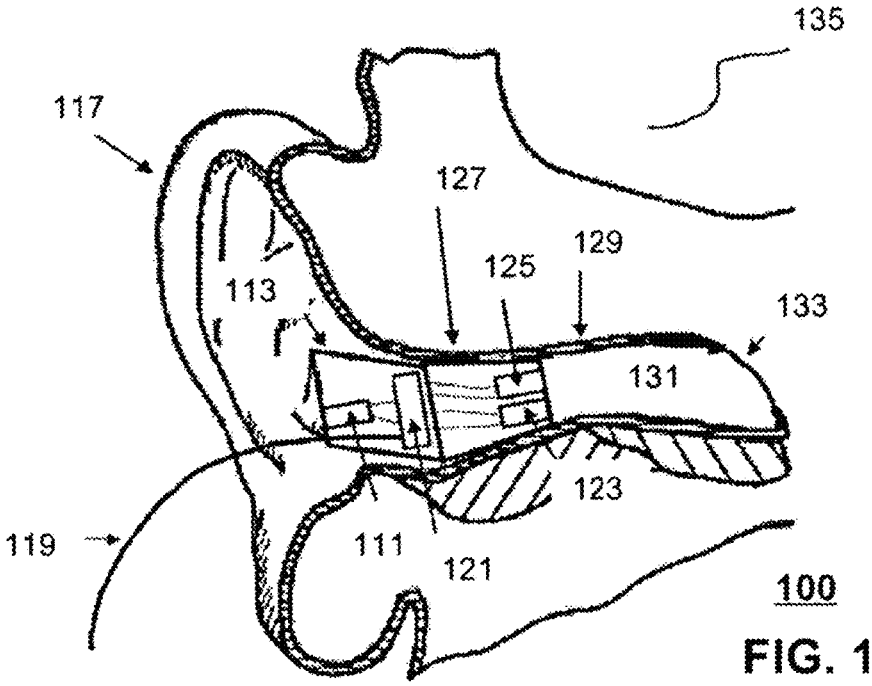

135
117
127
129
125
113
133
131
123
119
111
121
100

FIG. 1

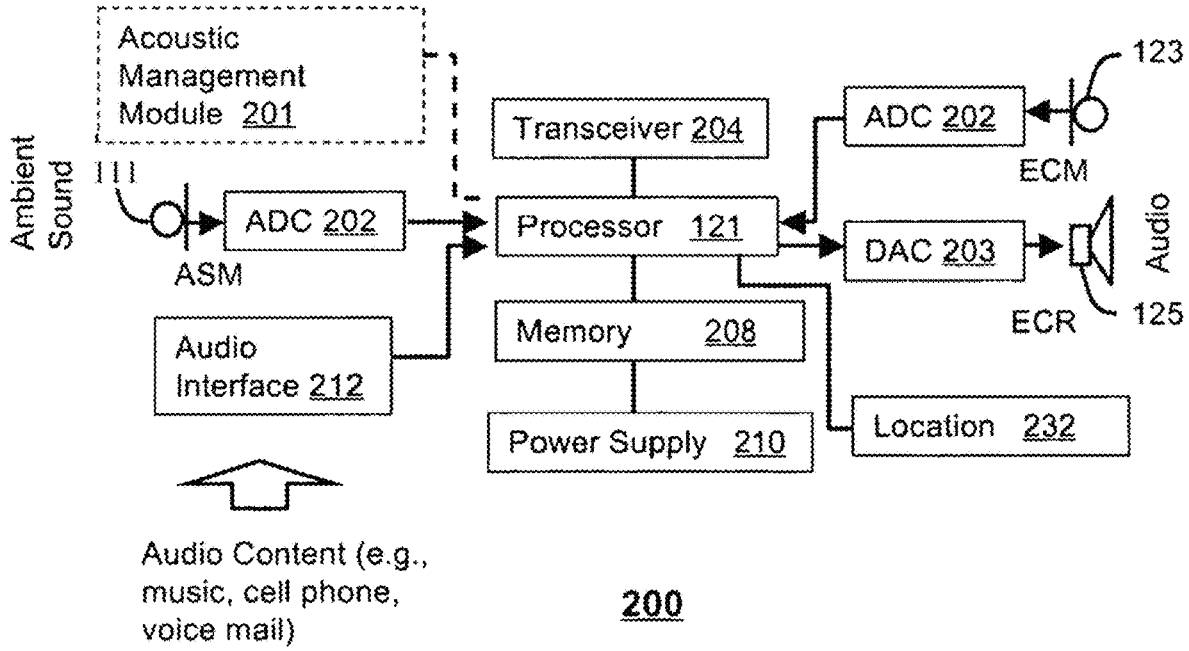

Acoustic Management Module 201
Ambient Sound
111
ADC 202
ASM
Transceiver 204
Processor 121
Memory 208
Power Supply 210
Audio Interface 212
ADC 202
123
ECM
DAC 203
Audio
ECR
125
Location 232
Audio Content (e.g., music, cell phone, voice mail)
200

FIG. 2

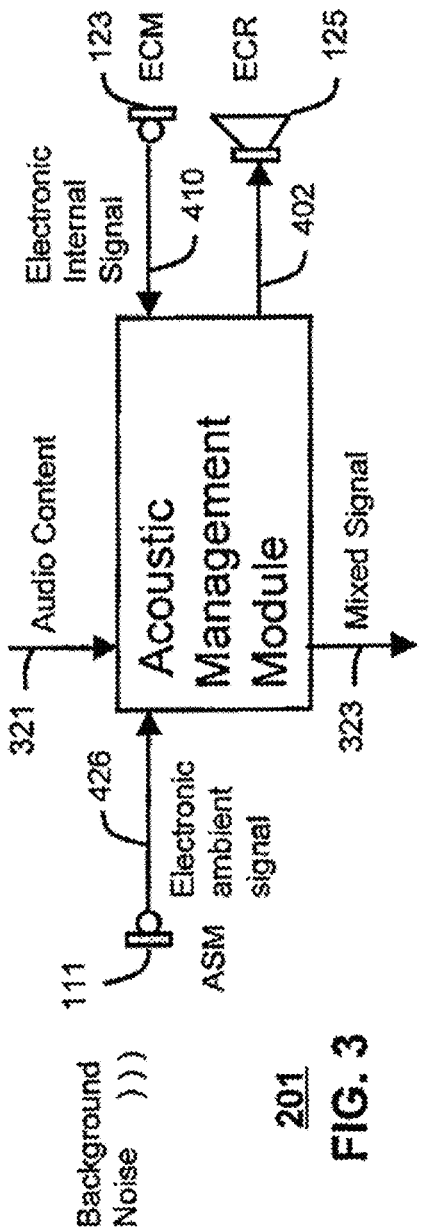

Background Noise
111
ASM
Electronic ambient signal
426
321
Audio Content
Acoustic Management Module
Electronic Internal Signal
410
123
ECM
402
ECR
125
323
Mixed Signal
201

FIG. 3

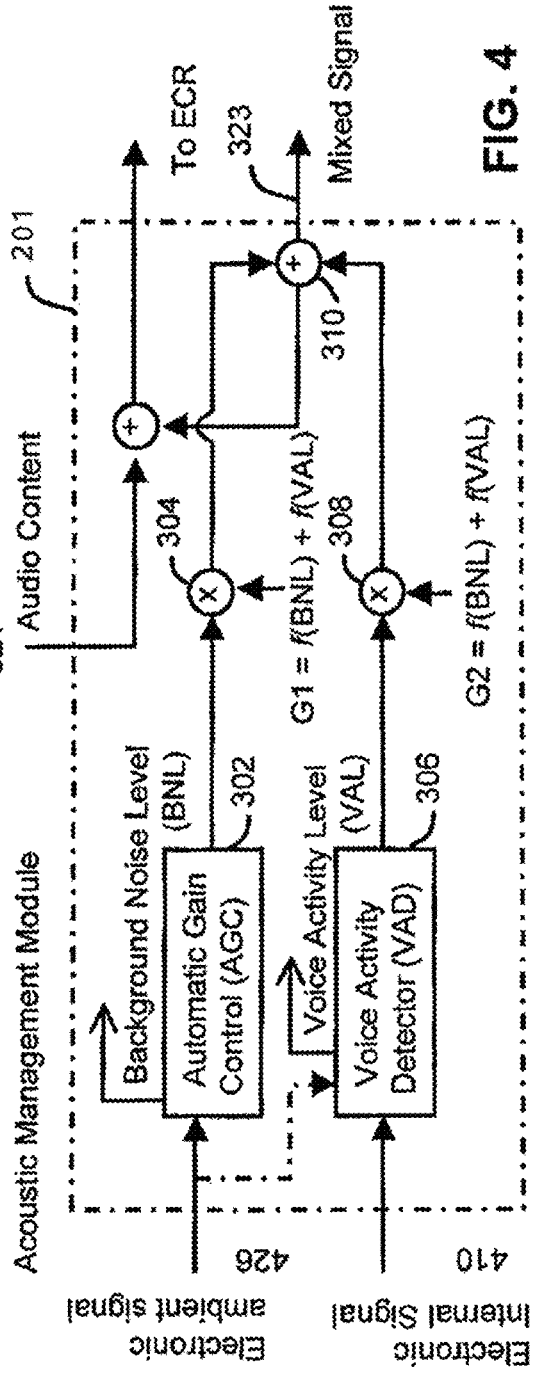

Acoustic Management Module
201
321
Audio Content
Background Noise Level (BNL)
Automatic Gain Control (AGC)
302
Voice Activity Level (VAL)
Voice Activity Detector (VAD)
306
304
308
310
G1 = f(BNL) + f(VAL)
G2 = f(BNL) + f(VAL)
To ECR
323
Mixed Signal
Electronic ambient signal
426
Electronic Internal Signal
410

FIG. 4

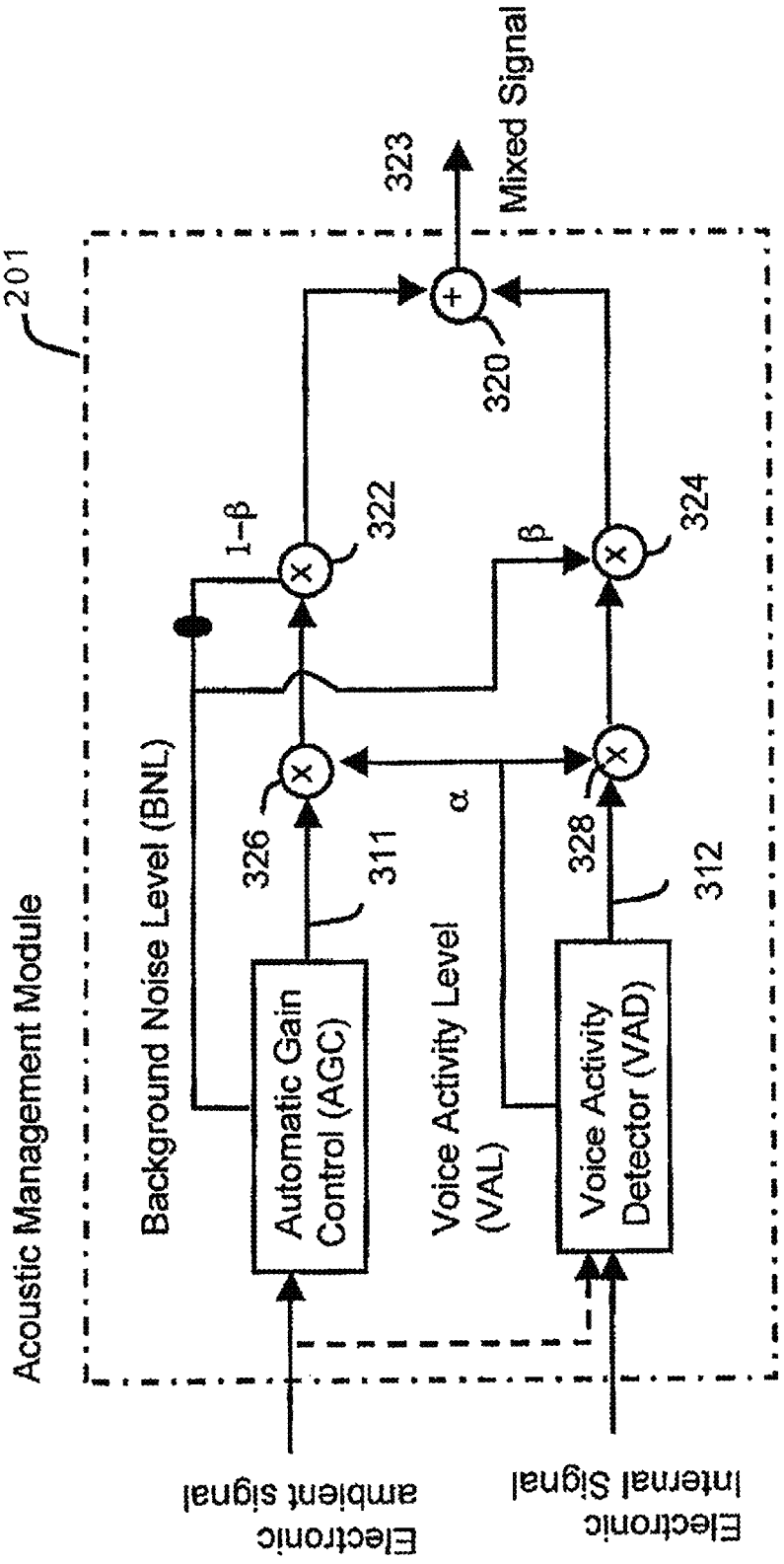
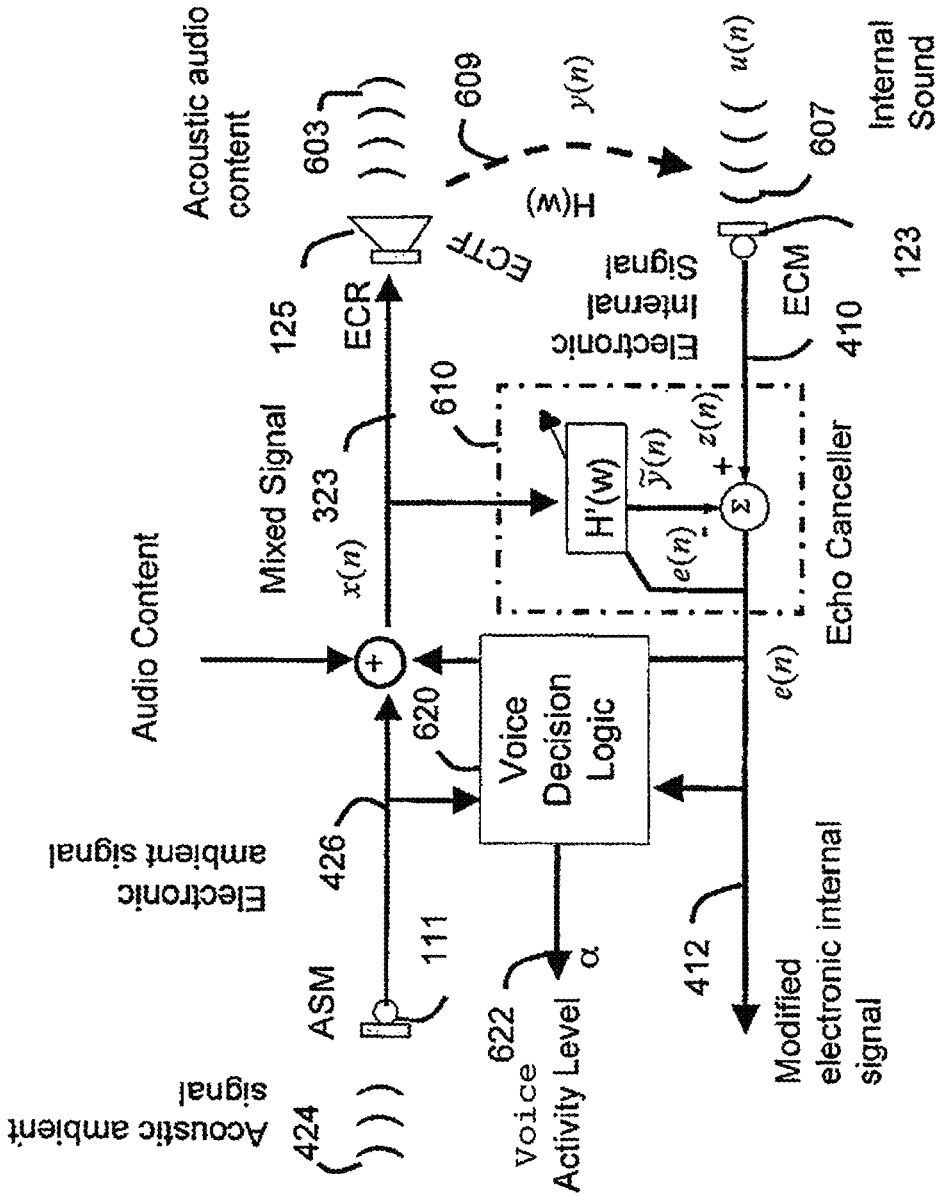

Acoustic Management Module
201
Background Noise Level (BNL)
Voice Activity Level (VAL)
Automatic Gain Control (AGC)
Voice Activity Detector (VAD)
Electronic ambient signal
Electronic Internal Signal
Mixed Signal
323
320
322
324
326
328
311
312
1−β
β
α

FIG. 5


Acoustic audio content
603
609
$y(n)$
H(w)
ECTF
Internal Sound
607
$u(n)$
123
ECR
125
Electronic Internal Signal
ECM
410
Mixed Signal
323
$x(n)$
610
H'(W)
$\tilde{y}(n)$
$z(n)$
$e(n)$
Echo Canceller
Audio Content
620
Voice Decision Logic
$e(n)$
Electronic ambient signal
426
ASM
111
Acoustic ambient signal
424
Voice Activity Level
622
α
412
Modified electronic internal signal

600

FIG. 6

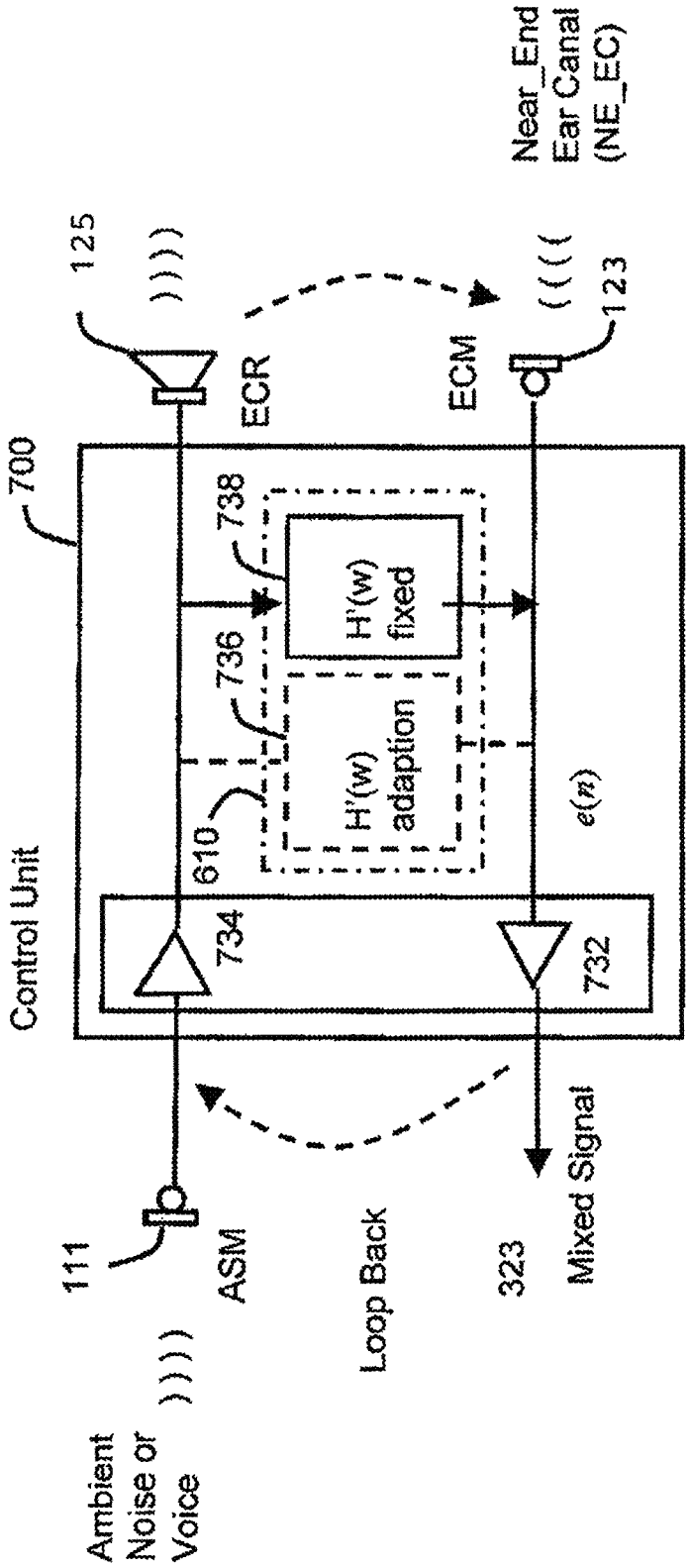

Near_End
Ear Canal
(NE_EC)
125
123
ECR
ECM
700
738
H'(w)
fixed
736
H'(w)
adaption
610
e(n)
Control Unit
734
732
111
ASM
Loop Back
323
Mixed Signal
Ambient
Noise or
Voice

FIG. 7

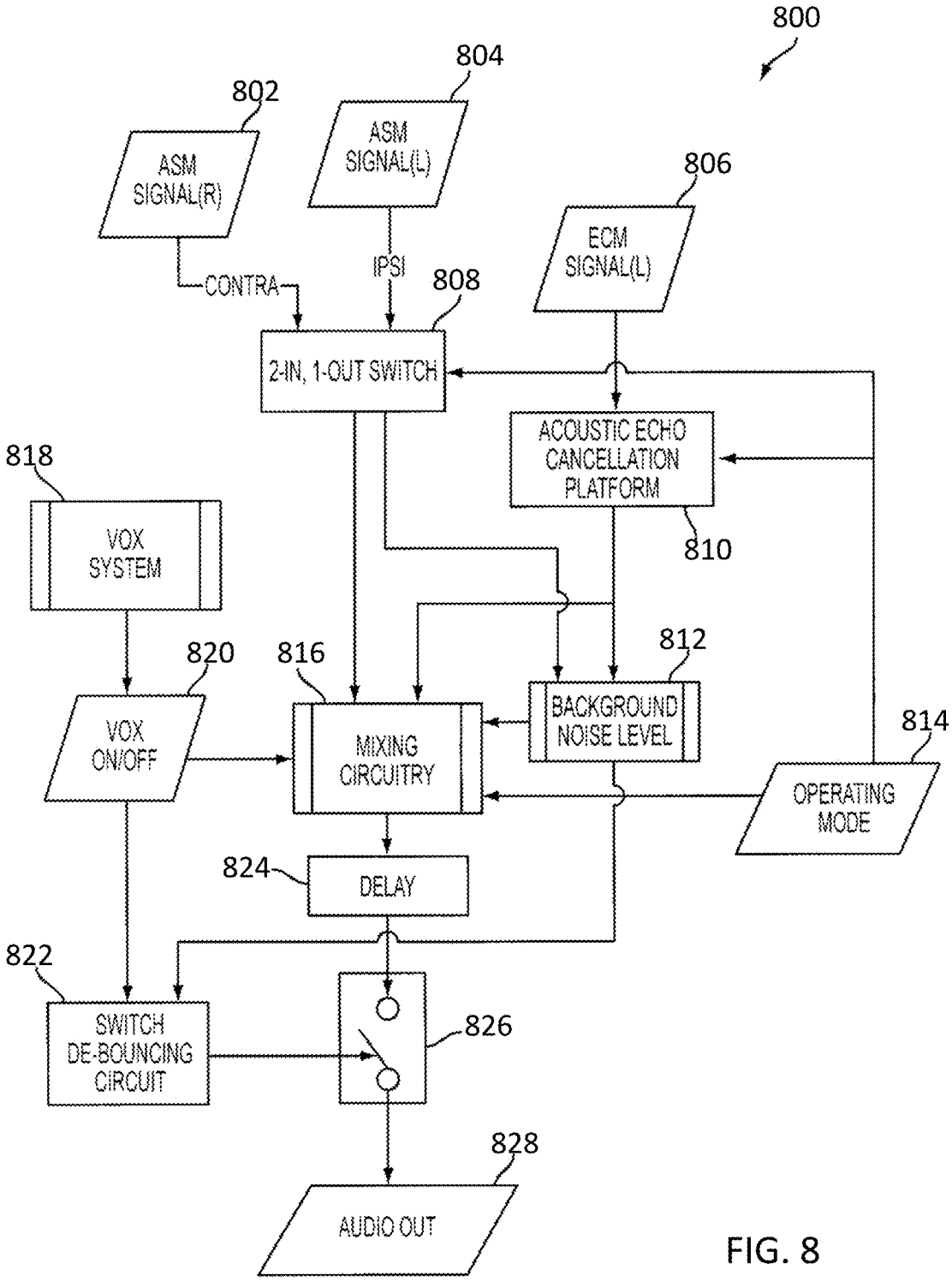

800
804
802
ASM SIGNAL(L)
ASM SIGNAL(R)
806
ECM SIGNAL(L)
IPSI
CONTRA
808
2-IN, 1-OUT SWITCH
ACOUSTIC ECHO CANCELLATION PLATFORM
818
VOX SYSTEM
810
820
816
812
VOX ON/OFF
MIXING CIRCUITRY
BACKGROUND NOISE LEVEL
814
OPERATING MODE
824
DELAY
822
SWITCH DE-BOUNCING CIRCUIT
826
828
AUDIO OUT

FIG. 8

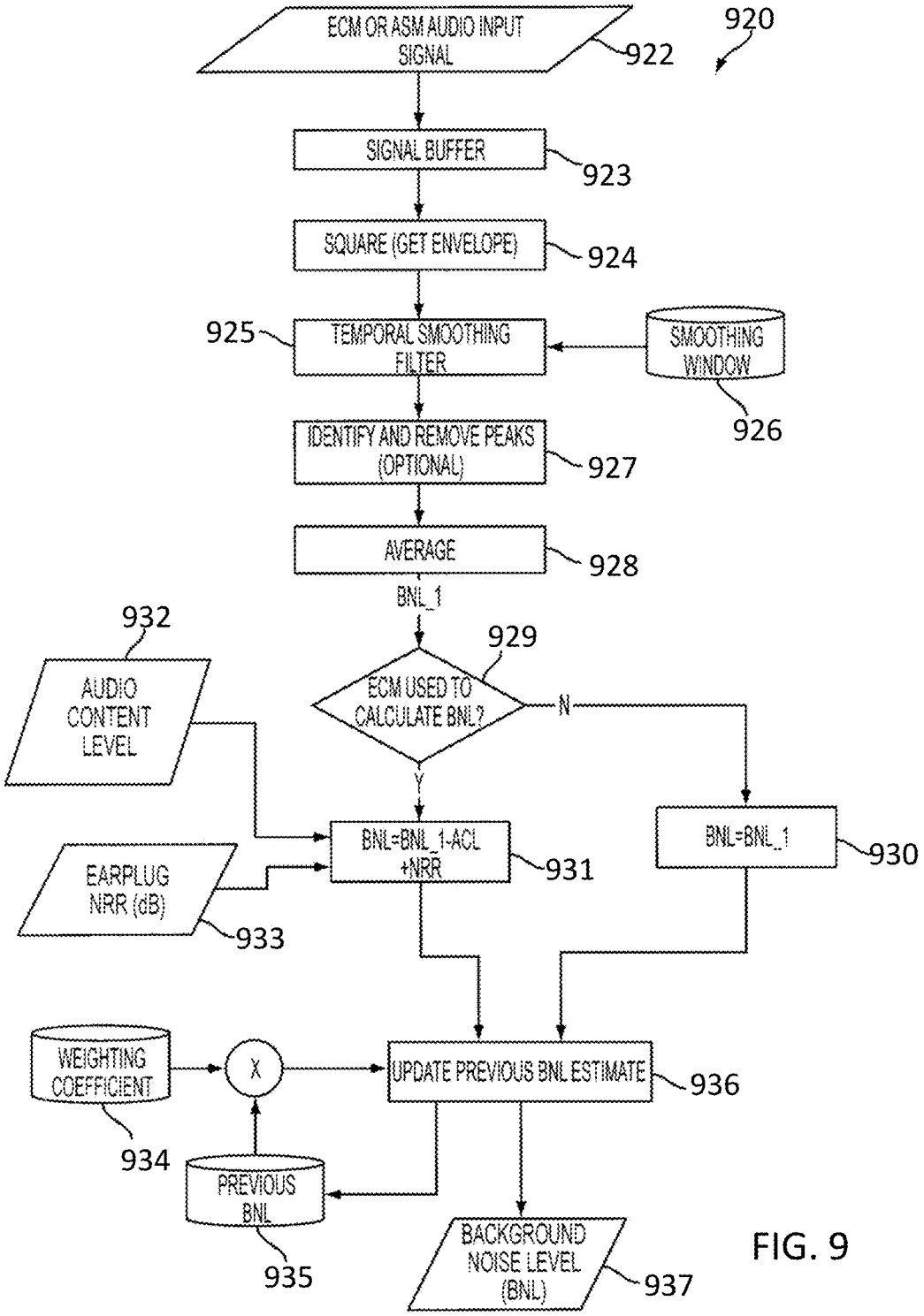

ECM OR ASM AUDIO INPUT SIGNAL
922
920
SIGNAL BUFFER
923
SQUARE (GET ENVELOPE)
924
925
TEMPORAL SMOOTHING FILTER
SMOOTHING WINDOW
926
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
927
AVERAGE
928
BNL_1
932
AUDIO CONTENT LEVEL
929
ECM USED TO CALCULATE BNL?
N
Y
BNL=BNL_1-ACL +NRR
931
BNL=BNL_1
930
EARPLUG NRR (dB)
933
WEIGHTING COEFFICIENT
X
UPDATE PREVIOUS BNL ESTIMATE
936
934
PREVIOUS BNL
935
BACKGROUND NOISE LEVEL (BNL)
937

FIG. 9

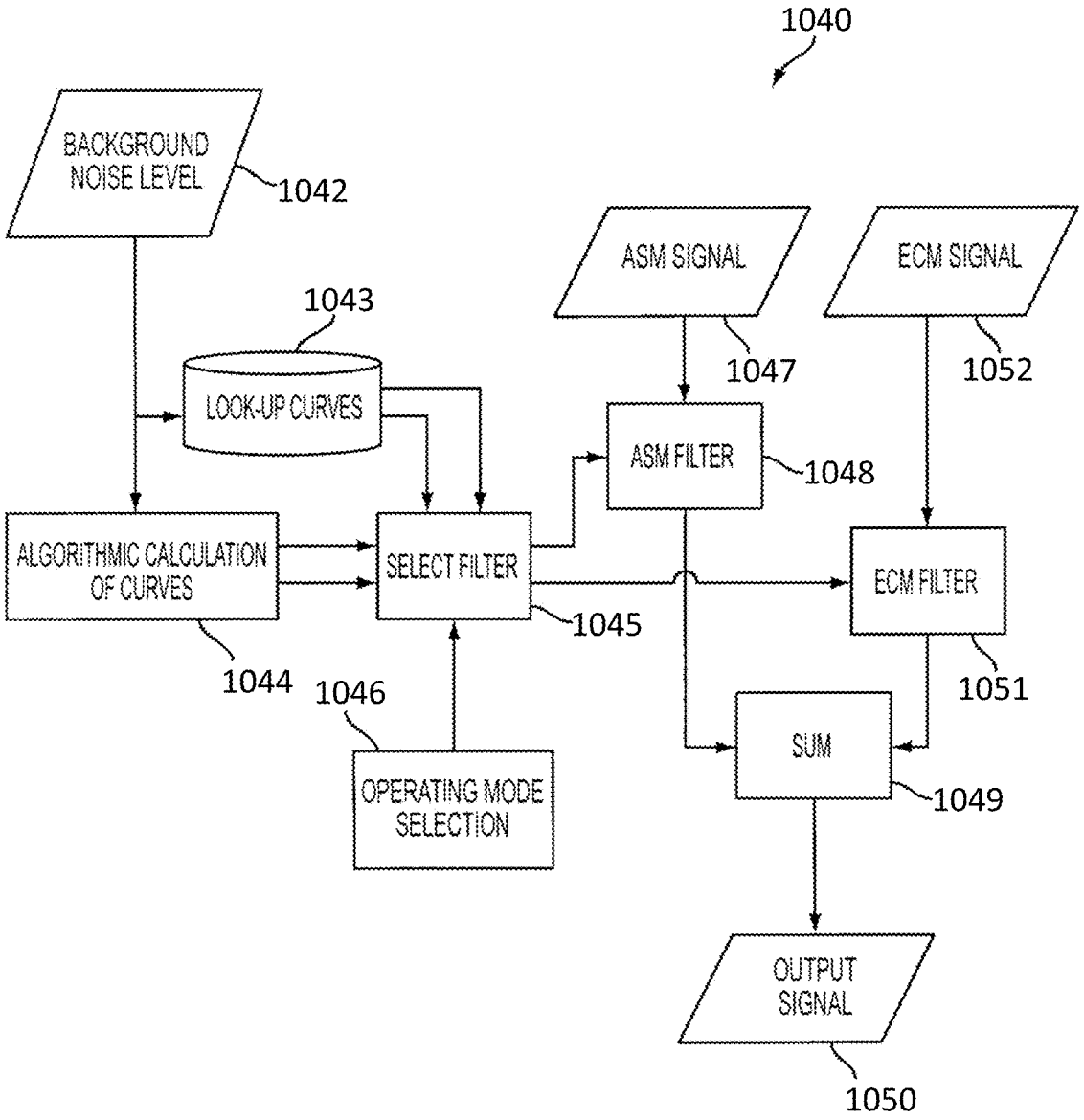

1040
BACKGROUND NOISE LEVEL
1042
1043
LOOK-UP CURVES
ASM SIGNAL
1047
ECM SIGNAL
1052
ASM FILTER
1048
ALGORITHMIC CALCULATION OF CURVES
SELECT FILTER
1045
ECM FILTER
1044
1046
1051
SUM
OPERATING MODE SELECTION
1049
OUTPUT SIGNAL
1050

FIG. 10

FIG. 11

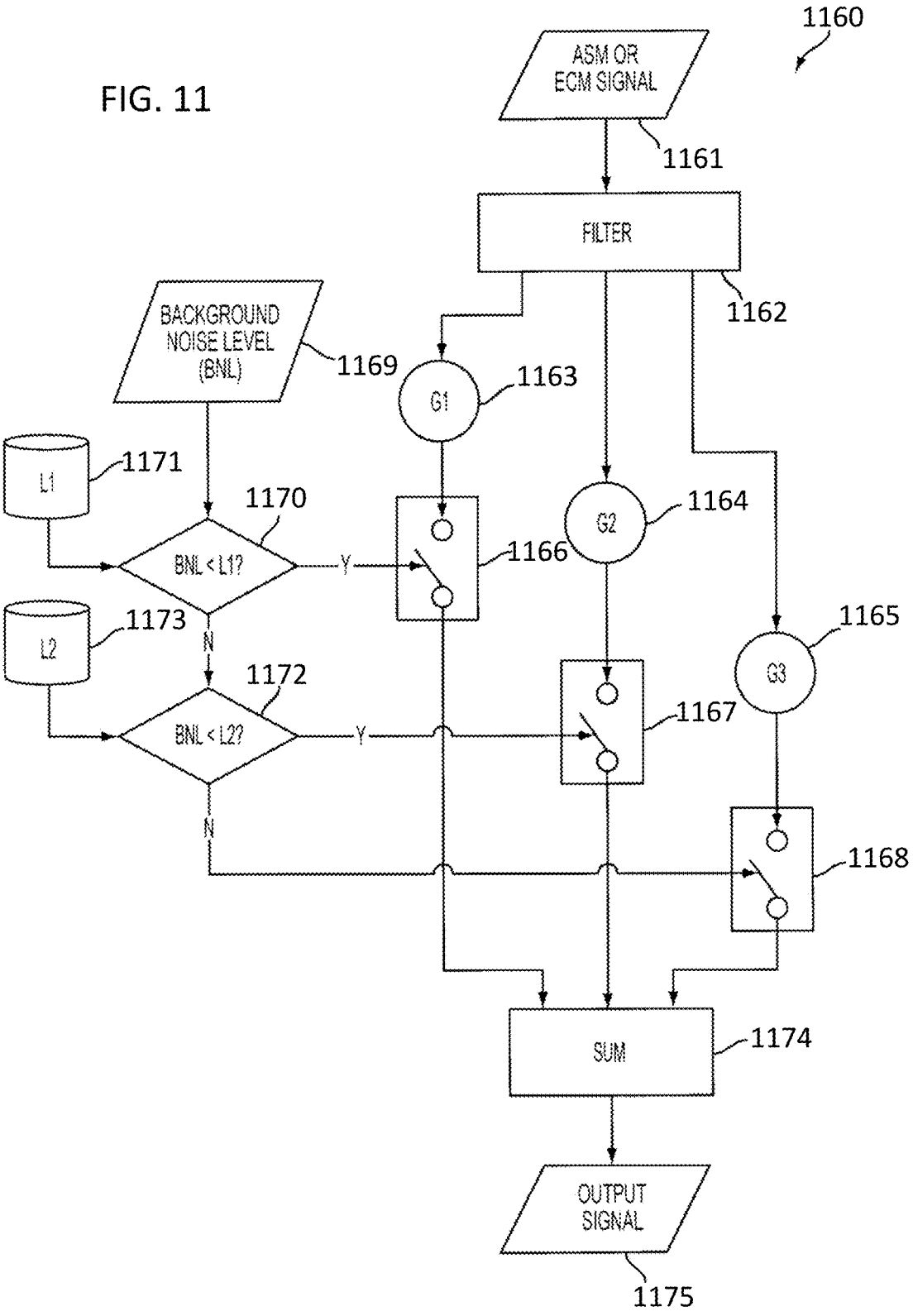
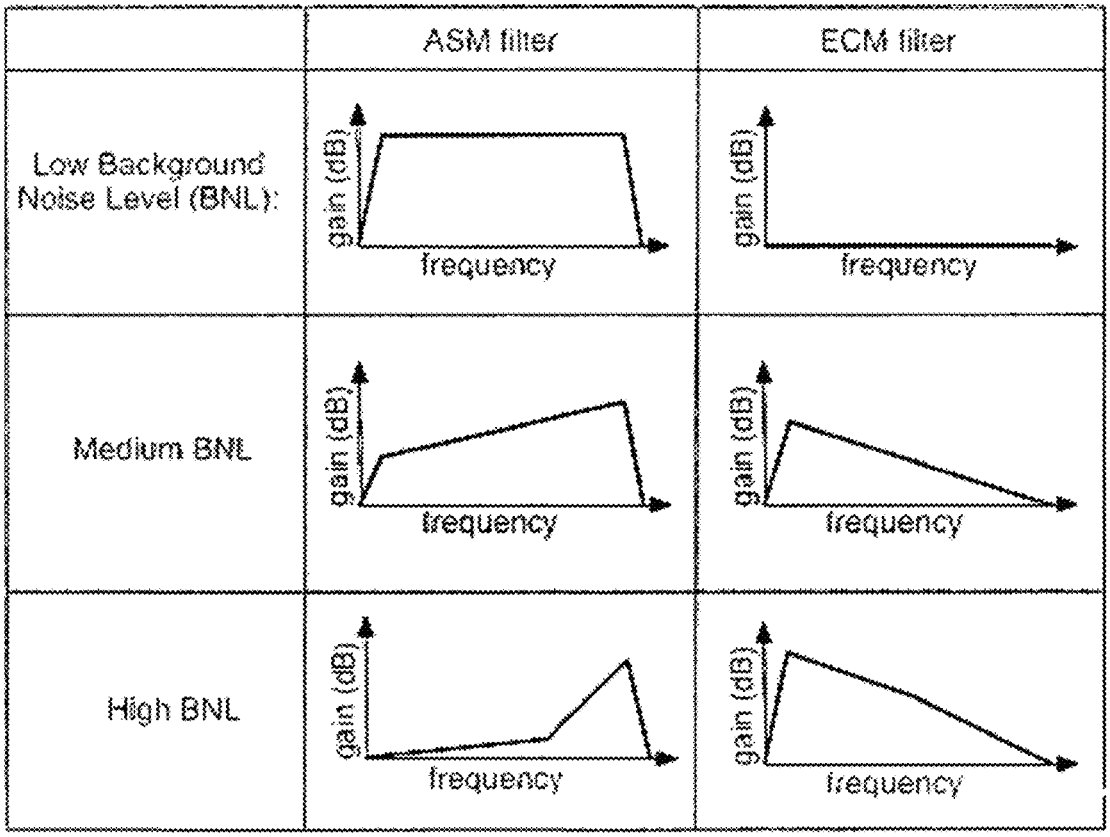

1160
ASM OR ECM SIGNAL
1161
FILTER
1162
BACKGROUND NOISE LEVEL (BNL)
1169
G1
1163
L1
1171
1170
BNL < L1?
Y
1166
G2
1164
L2
1173
N
1172
BNL < L2?
Y
1167
G3
1165
N
1168
SUM
1174
OUTPUT SIGNAL
1175

FIG. 12


ASM filter
ECM filter
Low Background Noise Level (BNL):
Medium BNL
High BNL
gain (dB)
frequency

METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

CROSS REFERENCE TO RELATED APPLICATIONS

This application is a Continuation in Part of U.S. patent application Ser. No. 16/247,186, filed 14 Jan. 2019, which is a Continuation of U.S. patent application Ser. No. 13/956,767, filed on 1 Aug. 2018, now U.S. Pat. No. 10,182,289, which is a Continuation of U.S. patent application Ser. No. 12/170,171, filed on 9 Jul. 2008, now U.S. Pat. No. 8,526,645, which is a Continuation in Part of application Ser. No. 12/115,349 filed on May 5, 2008, now U.S. Pat. No. 8,081,780 which claims the priority benefit of Provisional Application No. 60/916,271 filed on May 4, 2007, the entire disclosure of all of which are incorporated herein by reference.

FIELD OF THE INVENTION

The present invention pertains to sound reproduction, sound recording, audio communications and hearing protection using earphone devices designed to provide variable acoustical isolation from ambient sounds while being able to audition both environmental and desired audio stimuli. Particularly, the present invention describes a method and device for suppressing echo in an ear-canal when capturing a user's voice when using an ambient sound microphone and an ear canal microphone.

BACKGROUND OF THE INVENTION

People use headsets or earpieces primarily for voice communications and music listening enjoyment. A headset or earpiece generally includes a microphone and a speaker for allowing the user to speak and listen. An ambient sound microphone mounted on the earpiece can capture ambient sounds in the environment; sounds that can include the user's voice. An ear canal microphone mounted internally on the earpiece can capture voice resonant within the ear canal; sounds generated when the user is speaking.

An earpiece that provides sufficient occlusion can utilize both the ambient sound microphone and the ear canal microphone to enhance the user's voice. An ear canal receiver mounted internal to the ear canal can loopback sound captured at the ambient sound microphone or the ear canal microphone to allow the user to listen to captured sound. If the earpiece is however not properly sealed within the ear canal, the ambient sounds can leak through into the ear canal and create an echo feedback condition with the ear canal microphone and ear canal receiver. In such cases, the feedback loop can generate an annoying "howling" sound that degrades the quality of the voice communication and listening experience.

SUMMARY OF THE INVENTION

Embodiments in accordance with the present invention provide a method and device for background noise control, ambient sound mixing and other audio control methods associated with an earphone. Note that although this application is filed as a continuation in part of U.S. patent application Ser. No. 16/247,186, the subject matter material can be found in U.S. patent application Ser. No. 12/170,171, filed on 9 Jul. 2008, now U.S. Pat. No. 8,526,645, application Ser. No. 12/115,349 filed on May 5, 2008, now U.S. Pat. No. 8,081,780, and Application No. 60/916,271 filed on May 4, 2007, all of which were incorporated by reference in U.S. patent application Ser. No. 16/247,186 and are incorporated by reference in their entirety herein.

In a first embodiment, a method for in-ear canal echo suppression control can include the steps of capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal, measuring a background noise signal from the electronic ambient signal and the electronic internal signal, and capturing in the ear canal an internal sound from an Ear Canal Microphone (ECM) to produce an electronic internal signal. The electronic internal signal includes an echo of a spoken voice generated by a wearer of the earpiece. The echo in the electronic internal signal can be suppressed to produce a modified electronic internal signal containing primarily the spoken voice. A voice activity level can be generated for the spoken voice based on characteristics of the modified electronic internal signal and a level of the background noise signal. The electronic ambient signal and the electronic internal signal can then be mixed in a ratio dependent on the background noise signal to produce a mixed signal without echo that is delivered to the ear canal by way of the ECR.

An internal gain of the electronic internal signal can be increased as background noise levels increase, while an external gain of the electronic ambient signal can be decreased as the background noise levels increase. Similarly, the internal gain of the electronic internal signal can be increased as background noise levels decrease, while an external gain of the electronic ambient signal can be increased as the background noise levels decrease. The step of mixing can include filtering the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal. The characteristic can be a level of the background noise level, a spectral profile, or an envelope fluctuation.

At low background noise levels and low voice activity levels, the electronic ambient signal can be amplified relative to the electronic internal signal in producing the mixed signal. At medium background noise levels and voice activity levels, low frequencies in the electronic ambient signal and high frequencies in the electronic internal signal can be attenuated. At high background noise levels and high voice activity levels, the electronic internal signal can be amplified relative to the electronic ambient signal in producing the mixed signal.

The method can include adapting a first set of filter coefficients of a Least Mean Squares (LMS) filter to model an inner ear-canal microphone transfer function (ECTF). The voice activity level of the modified electronic internal signal can be monitored, and an adaptation of the first set of filter coefficients for the modified electronic internal signal can be frozen if the voice activity level is above a predetermined threshold. The voice activity level can be determined by an energy level characteristic and a frequency response characteristic. A second set of filter coefficients for a replica of the LMS filter can be generated during the freezing and substituted back for the first set of filter coefficients when the voice activity level is below another predetermined threshold. The modified electronic internal signal can be transmitted to another voice communication device and looped back to the ear canal.

In a second embodiment, a method for in-ear canal echo suppression control can include capturing an ambient sound

from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, delivering audio content to an ear canal by way of an Ear Canal Receiver (ECR) to produce an acoustic audio content, capturing in the ear canal by way of an Ear Canal Receiver (ECR) the acoustic audio content to produce an electronic internal signal, generating a voice activity level of a spoken voice in the presence of the acoustic audio content, suppressing an echo of the spoken voice in the electronic internal signal to produce a modified electronic internal signal, and controlling a mixing of the electronic ambient signal and the electronic internal signal based on the voice activity level. At least one voice operation of the earpiece can be controlled based on the voice activity level. The modified electronic internal signal can be transmitted to another voice communication device and looped back to the ear canal.

The method can include measuring a background noise signal from the electronic ambient signal and the electronic internal signal, and mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal that is delivered to the ear canal by way of the ECR. An acoustic attenuation level of the earpiece and an audio content level reproduced can be accounted for when adjusting the mixing based on a level of the audio content, the background noise level, and an acoustic attenuation level of the earpiece. The electronic ambient signal and the electronic internal signal can be filtered based on a characteristic of the background noise signal. The characteristic can be a level of the background noise level, a spectral profile, or an envelope fluctuation. The method can include applying a first gain (G1) to the electronic ambient signal, and applying a second gain (G2) to the electronic internal signal. The first gain and second gain can be a function of the background noise level and the voice activity level.

The method can include adapting a first set of filter coefficients of a Least Mean Squares (LMS) filter to model an inner ear-canal microphone transfer function (ECTF). The adaptation of the first set of filter coefficients can be frozen for the modified electronic internal signal if the voice activity level is above a predetermined threshold. A second set of filter coefficients for a replica of the LMS filter can be adapted during the freezing. The second set can be substituted back for the first set of filter coefficients when the voice activity level is below another predetermined threshold. The adaptation of the first set of filter coefficients can then be unfrozen.

In a third embodiment, an earpiece to provide in-ear canal echo suppression can include an Ambient Sound Microphone (ASM) configured to capture ambient sound and produce an electronic ambient signal, an Ear Canal Receiver (ECR) to deliver audio content to an ear canal to produce an acoustic audio content, an Ear Canal Microphone (ECM) configured to capture internal sound including spoken voice in an ear canal and produce an electronic internal signal, and a processor operatively coupled to the ASM, the ECM and the ECR. The audio content can be a phone call, a voice message, a music signal, or the spoken voice. The processor can be configured to suppress an echo of the spoken voice in the electronic internal signal to produce a modified electronic internal signal, generate a voice activity level for the spoken voice based on characteristics of the modified electronic internal signal and a level of the background noise signal, and mix the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal that is delivered to the ear canal by way of the ECR. The processor can play the mixed signal back to the ECR for loopback listening. A transceiver operatively coupled to the processor can transmit the mixed signal to a second communication device.

A Least Mean Squares (LMS) echo suppressor can model an inner ear-canal microphone transfer function (ECTF) between the ASM and the ECM. A voice activity detector operatively coupled to the echo suppressor can adapt a first set of filter coefficients of the echo suppressor to model an inner ear-canal microphone transfer function (ECTF), and freeze an adaptation of the first set of filter coefficients for the modified electronic internal signal if the voice activity level is above a predetermined threshold. The voice activity detector during the freezing can also adapt a second set of filter coefficients for the echo suppressor, and substitute the second set of filter coefficients for the first set of filter coefficients when the voice activity level is below another predetermined threshold. Upon completing the substitution, the processor can unfreeze the adaptation of the first set of filter coefficients

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 block diagram for an acoustic management module in accordance with an exemplary embodiment;

FIG. **4** is a schematic for the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal as a function of a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **5** is a more detailed schematic of the acoustic management module of FIG. **3** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **6** is a block diagram of a system for in-ear canal echo suppression in accordance with an exemplary embodiment;

FIG. **7** is a schematic of a control unit for controlling adaptation of a first set and second set of filter coefficients of an echo suppressor for in-ear canal echo suppression in accordance with an exemplary embodiment;

FIG. **8** is a block diagram of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

FIG. **9** is a block diagram of a method for calculating background noise levels in accordance with an exemplary embodiment;

FIG. **10** is a block diagram for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment;

FIG. **11** is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment; and

FIG. **12** is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL) 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.

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.

Various embodiments herein provide a method and device for automatically mixing audio signals produced by a pair of microphone signals that monitor a first ambient sound field and a second ear canal sound field, to create a third new mixed signal. An Ambient Sound Microphone (ASM) and an Ear Canal Microphone (ECM) can be housed in an earpiece that forms a seal in the ear of a user. The third mixed signal can be auditioned by the user with an Ear Canal Receiver (ECR) mounted in the earpiece, which creates a sound pressure in the occluded ear canal of the user. A voice activity detector can determine when the user is speaking and control an echo suppressor to suppress associated feedback in the ECR.

When the user engages in a voice communication, the echo suppressor can suppress feedback of the spoken voice from the ECR. The echo suppressor can contain two sets of filter coefficients; a first set that adapts when voice is not present and becomes fixed when voice is present, and a second set that adapts when the first set is fixed. The voice activity detector can discriminate between audible content, such as music, that the user is listening to, and spoken voice generated by the user when engaged in voice communication. The third mixed signal contains primarily the spoken voice captured at the ASM and ECM without echo, and can be transmitted to a remote voice communications system, such as a mobile phone, personal media player, recording device, walkie-talkie radio, etc. Before the ASM and ECM signals are mixed, they can be echo suppressed and subjected to different filters and at optional additional gains. This permits a single earpiece to provide full-duplex voice communication with proper or improper acoustic sealing.

The characteristic responses of the ASM and ECM filter can differ based on characteristics of the background noise and the voice activity level. In some exemplary embodiments, the filter response can depend on the measured Background Noise Level (BNL). A gain of a filtered ASM and a filtered ECM signal can also depend on the BNL. The (BNL) can be calculated using either or both the conditioned ASM and/or ECM signal(s). The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and can be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level (i.e. the high and low frequencies are attenuated before the level of the microphone signals are calculated).

At least one exemplary embodiment of the invention is directed to an earpiece for voice operated control. Reference is made to FIG. **1** in which an earpiece device, generally indicated as earpiece **100**, is constructed and operates 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 **131**. 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 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 frequency response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum resulting from the sound field at the entrance to the ear canal **131**. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

Located adjacent to the ECR **125**, is the ECM **123**, which is acoustically coupled to the (closed or partially 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 the earpiece **100**. In one arrangement, the ASM **111** can be housed in the assembly **113** to monitor 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**.

The earpiece **100** can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece **100** in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

The earpiece **100** can measure ambient sounds in the environment 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.

The earpiece **100** can generate an Ear Canal Transfer Function (ECTF) to model the ear canal **131** using ECR **125** and ECM **123**, as well as an Outer Ear Canal Transfer function (OETF) using ASM **111**. For instance, the ECR **125** can deliver an impulse within the ear canal and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal. The earpiece **100** can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece **100** to safely administer and monitor sound exposure to the ear.

Referring to FIG. **2**, a block diagram **200** of the earpiece **100** in accordance with an exemplary embodiment is shown. As illustrated, the earpiece **100** can include the processor **121** 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 **121** 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 processor **121** can also include a clock to record a time stamp.

As illustrated, the earpiece **100** can include an acoustic management module **201** to mix sounds captured at the ASM **111** and ECM **123** to produce a mixed sound. The processor **121** can then provide the mixed signal to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor or communication device. The acoustic management module **201** can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the functionality of the acoustic management module **201** can be provided by way of software, such as program code, assembly language, or machine language.

The memory **208** can also store program instructions for execution on the processor **121** as well as captured audio processing data and filter coefficient data. The memory **208** can be off-chip and external to the processor **121** and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor **121**. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor **121** to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

The earpiece **100** can include an audio interface **212** operatively coupled to the processor **121** and acoustic management module **201** to receive audio content, for example from a media player, cell phone, or any other communication device, and deliver the audio content to the processor **121**. The processor **121** responsive to detecting spoken voice from the acoustic management module **201** can adjust the audio content delivered to the ear canal. For instance, the processor **121** (or acoustic management module **201**) can lower a volume of the audio content responsive to detecting a spoken voice. The processor **121** by way of the ECM **123** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the acoustic management module **201**.

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 location receiver **232** can utilize common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece **100**.

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 **121** 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 block diagram of the acoustic management module **201** in accordance with an exemplary embodiment. Briefly, the Acoustic management module **201** facilitates monitoring, recording and transmission of user-generated voice (speech) to a voice communication system. User-generated sound is detected with the ASM **111** that monitors a sound field near the entrance to a user's ear, and with the ECM **123** that monitors a sound field in the user's occluded ear canal. A new mixed signal **323** is created by filtering and mixing the ASM and ECM microphone signals. The filtering and mixing process is automatically controlled depending on the background noise level of the ambient sound field to enhance intelligibility of the new mixed signal **323**. For instance, when the background noise level is high, the acoustic management module **201** automatically increases the level of the ECM **123** signal relative to the level of the ASM **111** to create the new signal mixed **323**. When the background noise level is low, the acoustic management module **201** automatically decreases the level of the ECM **123** signal relative to the level of the ASM **111** to create the new signal mixed **323**

As illustrated, the ASM **111** is configured to capture ambient sound and produce an electronic ambient signal **426**, the ECR **125** is configured to pass, process, or play acoustic audio content **402** (e.g., audio content **321**, mixed signal **323**) to the ear canal, and the ECM **123** is configured to capture internal sound in the ear canal and produce an electronic internal signal **410**. The acoustic management module **201** is configured to measure a background noise signal from the electronic ambient signal **426** or the electronic internal signal **410**, and mix the electronic ambient signal **426** with the electronic internal signal **410** in a ratio dependent on the background noise signal to produce the mixed signal **323**. The acoustic management module **201**

filters the electronic ambient signal **426** and the electronic internal **410** signal based on a characteristic of the background noise signal using filter coefficients stored in memory or filter coefficients generated algorithmically.

In practice, the acoustic management module **201** mixes sounds captured at the ASM **111** and the ECM **123** to produce the mixed signal **323** based on characteristics of the background noise in the environment and a voice activity level. The characteristics can be a background noise level, a spectral profile, or an envelope fluctuation. The acoustic management module **201** manages echo feedback conditions affecting the voice activity level when the ASM **111**, the ECM **123**, and the ECR **125** are used together in a single earpiece for full-duplex communication, when the user is speaking to generate spoken voice (captured by the ASM **111** and ECM **123**) and simultaneously listening to audio content (delivered by ECR **125**).

In noisy ambient environments, the voice captured at the ASM **111** includes the background noise from the environment, whereas, the internal voice created in the ear canal **131** captured by the ECM **123** has less noise artifacts, since the noise is blocked due to the occlusion of the earpiece **100** in the ear. It should be noted that the background noise can enter the ear canal if the earpiece **100** is not completely sealed. In this case, when speaking, the user's voice can leak through and cause an echo feedback condition that the acoustic management module **201** mitigates.

FIG. **4** is a schematic of the acoustic management module **201** illustrating a mixing of the electronic ambient signal **426** with the electronic internal signal **410** as a function of a background noise level (BNL) and a voice activity level (VAL) in accordance with an exemplary embodiment. As illustrated, the acoustic management module **201** includes an Automatic Gain Control (AGC) **302** to measure background noise characteristics. The acoustic management module **201** also includes a Voice Activity Detector (VAD) **306**. The VAD **306** can analyze either or both the electronic ambient signal **426** and the electronic internal signal **410** to estimate the VAL. As an example, the VAL can be a numeric range such as 0 to 10 indicating a degree of voicing. For instance, a voiced signal can be predominately periodic due to the periodic vibrations of the vocal cords. A highly voiced signal (e.g., vowel) can be associated with a high level, and a non-voiced signal (e.g., fricative, plosive, consonant) can be associated with a lower level.

The acoustic management module **201** includes a first gain (G1) **304** applied to the AGC processed electronic ambient signal **426**. A second gain (G2) **308** is applied to the VAD processed electronic internal signal **410**. The acoustic management module **201** applies the first gain (G1) **304** and the second gain (G2) **308** as a function of the background noise level and the voice activity level to produce the mixed signal **323**, where

$$G1=f(\mathrm{BNL})+f(\mathrm{VAL}) \text{ and } G2=f(\mathrm{BNL})+f(\mathrm{VAL})$$

As illustrated, the mixed signal **323** is the sum **310** of the G1 scaled electronic ambient signal and the G2 scaled electronic internal signal. The mixed signal **323** can then be transmitted to a second communication device (e.g. second cell phone, voice recorder, etc.) to receive the enhanced voice signal. The acoustic management module **201** can also play the mixed signal **323** back to the ECR for loopback listening. The loopback allows the user to hear himself or herself when speaking, as though the earpiece **100** and associated occlusion effect were absent. The loopback can also be mixed with the audio content **321** based on the background noise level, the VAL, and audio content level. The acoustic management module **201** can also account for an acoustic attenuation level of the earpiece, and account for the audio content level reproduced by the ECR when measuring background noise characteristics. Echo conditions created as a result of the loopback can be mitigated to ensure that the voice activity level is accurate.

FIG. **5** is a more detailed schematic of the acoustic management module **201** illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment. In particular, the gain blocks for G1 and G2 of FIG. **4** are a function of the BNL and the VAL and are shown in greater detail. As illustrated, the AGC produces a BNL that can be used to set a first gain **322** for the processed electronic ambient signal **311** and a second gain **324** for the processed electronic internal signal **312**. For instance, when the BNL is low (<70 dBA), gain **322** is set higher relative to gain **324** so as to amplify the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. When the BNL is high (>85 dBA), gain **322** is set lower relative to gain **324** so as to attenuate the electronic ambient signal **311** in greater proportion than the electronic internal signal **312**. The mixing can be performed in accordance with the relation:

$$\text{Mixed signal}=(1-\beta)*\text{electronic ambient signal}+(\beta)*\text{electronic internal signal}$$

where $(1-\beta)$ is an external gain, (β) is an internal gain, and the mixing is performed with $0<\beta<1$.

As illustrated, the VAD produces a VAL that can be used to set a third gain **326** for the processed electronic ambient signal **311** and a fourth gain **328** for the processed electronic internal signal **312**. For instance, when the VAL is low (e.g., 0-3), gain **326** and gain **328** are set low so as to attenuate the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is not detected. When the VAL is high (e.g., 7-10), gain **326** and gain **328** are set high so as to amplify the electronic ambient signal **311** and the electronic internal signal **312** when spoken voice is detected.

The gain scaled processed electronic ambient signal **311** and the gain scaled processed electronic internal signal **312** are then summed at adder **320** to produce the mixed signal **323**. The mixed signal **323**, as indicated previously, can be transmitted to another communication device, or as loopback to allow the user to hear his or her self.

FIG. **6** is an exemplary schematic of an operational unit **600** of the acoustic management module for in-ear canal echo suppression in accordance with an embodiment. The operational unit **600** may contain more or less than the number of components shown in the schematic. The operational unit **600** can include an echo suppressor **610** and a voice decision logic **620**.

The echo suppressor **610** can be a Least Mean Squares (LMS) or Normalized Least Mean Squares (NLMS) adaptive filter that models an ear canal transfer function (ECTF) between the ECR **125** and the ECM **123**. The echo suppressor **610** generates the modified electronic signal, e(n), which is provided as an input to the voice decision logic **620**; e(n) is also termed the error signal e(n) of the echo suppressor **610**. Briefly, the error signal e(n) **412** is used to update the filter H(w) to model the ECTF of the echo path. The error signal e(n) **412** closely approximates the user's spoken voice signal u(n) **607** when the echo suppressor **610** accurately models the ECTF.

In the configuration shown the echo suppressor **610** minimizes the error between the filtered signal, γ(n), and the electronic internal signal, z(n), in an effort to obtain a

transfer function H' which is a best approximation to the H(w) (i.e., ECTF). H(w) represents the transfer function of the ear canal and models the echo response. (z(n)=u(n)+y(n)+v(n), where u(n) is the spoken voice **607**, y(n) is the echo **609**, and v(n) is background noise (if present, for instance due to improper sealing).)

During operation, the echo suppressor **610** monitors the mixed signal **323** delivered to the ECR **125** and produces an echo estimate $\tilde{y}(n)$ of an echo y(n) **609** based on the captured electronic internal signal **410** and the mixed signal **323**. The echo suppressor **610**, upon learning the ECTF by an adaptive process, can then suppress the echo y(n) **609** of the acoustic audio content **603** (e.g., output mixed signal **323**) in the electronic internal signal z(n) **410**. It subtracts the echo estimate $\tilde{y}(n)$ from the electronic internal signal **410** to produce the modified electronic internal signal e(n) **412**.

The voice decision logic **620** analyzes the modified electronic signal **412** e(n) and the electronic ambient signal **426** to produce a voice activity level **622**, α. The voice activity level α identifies a probability that the user is speaking, for example, when the user is using the earpiece for two way voice communication. The voice activity level **622** can also indicate a degree of voicing (e.g., periodicity, amplitude), When the user is speaking, voice is captured externally (such as from acoustic ambient signal **424**) by the ASM **111** in the ambient environment and also by the ECM **123** in the ear canal. The voice decision logic provides the voice activity level α to the acoustic management module **201** as an input parameter for mixing the ASM **111** and ECM **123** signals. Briefly referring back to FIG. **4**, the acoustic management module **201** performs the mixing as a function of the voice activity level α and the background noise level (see G=f(BNL)+f(VAL)).

For instance, at low background noise levels and low voice activity levels, the acoustic management module **201** amplifies the electronic ambient signal **426** from the ASM **111** relative to the electronic internal signal **410** from the ECM **123** in producing the mixed signal **323**. At medium background noise levels and medium voice activity levels, the acoustic management module **201** attenuates low frequencies in the electronic ambient signal **426** and attenuates high frequencies in the electronic internal signal **410**. At high background noise levels and high voice activity levels, the acoustic management module **201** amplifies the electronic internal signal **410** from the ECM **123** relative to the electronic ambient signal **426** from the ASM **111** in producing the mixed signal. The acoustic management module **201** can additionally apply frequency specific filters based on the characteristics of the background noise.

FIG. **7** is a schematic of a control unit **700** for controlling adaptation of a first set (**736**) and a second set (**738**) of filter coefficients of the echo suppressor **610** for in-ear canal echo suppression in accordance with an exemplary embodiment. Briefly, the control unit **700** illustrates a freezing (fixing) of weights upon detection of spoken voice. The echo suppressor resumes weight adaptation when e(n) is low, and freezes weights when e(n) is high signifying a presence of spoken voice.

When the user is not speaking, the ECR **125** can pass through ambient sound captured at the ASM **111**, thereby allowing the user to hear environmental ambient sounds. As previously discussed, the echo suppressor **610** models an ECTF and suppresses an echo of the mixed signal **323** that is looped back to the ECR **125** by way of the ASM **111** (see dotted line Loop Back path). When the user is not speaking, the echo suppressor continually adapts to model the ECTF. When the ECTF is properly modeled, the echo suppressor **610** produces a modified internal electronic signal e(n) that is low in amplitude level (i.e., low in error). The echo suppressor adapts the weights to keep the error signal low. When the user speaks, the echo suppressor however initially produces a high-level e(n) (e.g., the error signal increases). This happens since the speaker's voice is uncorrelated with the audio signal played out the ECR **125**, which disrupts the echo suppressor's ECTF modeling ability.

The control unit **700** upon detecting a rise in e(n), freezes the weights of the echo suppressor **610** to produce a fixed filter H'(w) fixed **738**. Upon detecting the rise in e(n) the control unit adjusts the gain **734** for the ASM signal and the gain **732** for the mixed signal **323** that is looped back to the ECR **125**. The mixed signal **323** fed back to the ECR **125** permits the user to hear themselves speak. Although the weights are frozen when the user is speaking, a second filter H'(w) **736** continually adapts the weights for generating a second e(n) that is used to determine a presence of spoken voice. That is, the control unit **700** monitors the second error signal e(n) produced by the second filter **736** for monitoring a presence of the spoken voice.

The first error signal e(n) (in a parallel path) generated by the first filter **738** is used as the mixed signal **323**. The first error signal contains primarily the spoken voice since the ECTF model has been fixed due to the weights. That is, the second (adaptive) filter is used to monitor a presence of spoken voice, and the first (fixed) filter is used to generate the mixed signal **323**.

Upon detecting a fall of e(n), the control unit restores the gains **734** and **732** and unfreezes the weights of the echo suppressor, and the first filter H'(w) returns to being an adaptive filter. The second filter H'(w) **736** remains on stand-by until spoken voice is detected, and at which point, the first filter H'(w) **738** goes fixed, and the second filter H'(w) **736** begins adaptation for producing the e(n) signal that is monitored for voice activity. Notably, the control unit **700** monitors e(n) from the first filter **738** or the second filter **736** for changes in amplitude to determine when spoken voice is detected based on the state of voice activity.

FIG. **8** is a block diagram **800** of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment.

As illustrated the mixing circuitry **816** (shown in center) receives an estimate of the background noise level **812** for mixing either or both the right earpiece ASM signal **802** and the left earpiece ASM signal **804** with the left earpiece ECM signal **806**. (The right earpiece ECM signal can be used similarly.) An operating mode selection system **814** selects a switching **808** (e.g., 2-in, 1-out) between the left earpiece ASM signal **804** and the right earpiece ASM signal **802**. As indicated earlier, the ASM signals and ECM signals can be first amplified with a gain system and then filtered with a filter system (the filtering may be accomplished using either analog or digital electronics or both). The audio input signals **802**, **804**, and **806** are therefore taken after this gain and filtering process, if any gain and filtering are used.

The Acoustic Echo Cancellation (AEC) system **810** can be activated with the operating mode selection system **814** when the mixed signal audio output **828** is reproduced with the ECR **125** in the same ear as the ECM **123** signal used to create the mixed signal audio output **828**. The acoustic echo cancellation platform **810** can also suppress an echo of a spoken voice generated by the wearer of the earpiece **100**. This ensures against acoustic feedback ("howlback").

The Voice Activated System (VOX) **818** in conjunction with a de-bouncing circuit **822** activates the electronic switch **826** to control the mixed signal output **828** from the mixing circuitry **816**; the mixed signal is a combination of the left ASM signal **804** or right ASM signal **802**, with the left ECM **806** signal. Though not shown, the same arrangement applies for the other earphone device for the right ear, if present. Note that earphones can be used in both ears simultaneously. In a contra-lateral operating mode, as selected by operating mode selection system **814**, the ASM and ECM signal are taken from opposite earphone devices, and the mix of these signals is reproduced with the ECR in the earphone that is contra-lateral to the ECM signal, and the same as the ASM signal.

For instance, in the contra-lateral operating mode, the ASM signal from the Right earphone device is mixed with the ECM signal from the left earphone device, and the audio signal corresponding to a mix of these two signals is reproduced with the Ear Canal Receiver (ECR) in the Right earphone device. The mixed signal audio output **828** therefore can contain a mix of the ASM and ECM signals when the user's voice is detected by the VOX. This mixed signal audio output can be used in loopback as a user Self-Monitor System to allow the user to hear their own voice as reproduced with the ECR **125**, or it may be transmitted to another voice system, such as a mobile phone, walkie-talkie radio etc. The VOX system **818** that activates the switch **826** may be one a number of VOX embodiments.

In a particular operating mode, specified by unit **814**, the conditioned ASM signal is mixed with the conditioned ECM signal with a ratio dependent on the BNL using audio signal mixing circuitry and the method described in either FIG. **10** or FIG. **11**. As the BNL increases, then the ASM signal is mixed with the ECM signal with a decreasing level. When the BNL is above a particular value, then a minimal level of the ASM signal is mixed with the ECM signal. When the VOX switch **618** is active, the mixed ASM and ECM signals are then sent to mixed signal output **828**. The switch de-bouncing circuit **826** ensures against the VOX **818** rapidly closing on and off (sometimes called chatter). This can be achieved with a timing circuit using digital or analog electronics. For instance, with a digital system, once the VOX has been activated, a time starts to ensure that the switch **826** is not closed again within a given time period, e.g. 100 ms. The delay unit **824** can improve the sound quality of the mixed signal audio output **828** by compensating for any latency in voice detection by the VOX system **818**. In some exemplary embodiments, the switch debouncing circuit **822** can be dependent by the BNL. For instance, when the BNL is high (e.g. above 85 dBA), the de-bouncing circuit can close the switch **826** sooner after the VOX output **818** determines that no user speech (e.g. spoken voice) is present.

FIG. **9** is a block diagram of a method **920** for calculating background noise levels in accordance with an exemplary embodiment. Briefly, the background noise levels can be calculated according to different contexts, for instance, if the user is talking while audio content is playing, if the user is talking while audio content is not playing, if the user is not talking but audio content is playing, and if the user is not talking and no audio content is playing. For instance, the system takes as its inputs either the ECM and/or ASM signal, depending on the particular system configuration. If the ECM signal is used, then the measured BNL accounts for an acoustic attenuation of the earpiece and a level of reproduced audio content.

As illustrated, modules **922-928** provide exemplary steps for calculating a base reference background noise level. The ECM or ASM audio input signal **922** can be buffered **923** in real-time to estimate signal parameters. An envelope detector **924** can estimate a temporal envelope of the ASM or ECM signal. A smoothing filter **925** can minimize abruptions in the temporal envelope. (A smoothing window **926** can be stored in memory). An optional peak detector **927** can remove outlier peaks to further smooth the envelope. An averaging system **928** can then estimate the average background noise level (BNL_1) from the smoothed envelope.

If at step **929**, it is determined that the signal from the ECM was used to calculate the BNL_1, an audio content level **932** (ACL) and noise reduction rating **933** (NRR) can be subtracted from the BNL_1 estimate to produce the updated BNL **931**. This is done to account for the audio content level reproduced by the ECR **125** that delivers acoustic audio content to the earpiece **100**, and to account for an acoustic attenuation level (i.e. Noise Reduction Rating **933**) of the earpiece. For example, if the user is listening to music, the acoustic management module **201** takes into account the audio content level delivered to the user when measuring the BNL. If the ECM is not used to calculate the BNL at step **929**, the previous real-time frame estimate of the BNL **930** is used.

At step **936**, the acoustic management module **201** updates the BNL based on the current measured BNL and previous BNL measurements **935**. For instance, the updated BNL **937** can be a weighted estimate 934 of previous BNL estimates according to BNL=2*previous BNL+(1−w)*current BNL, where 0<W<1. The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and may be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level.

FIG. **10** is a block diagram **1040** for mixing an external microphone signal with an internal microphone signal based on a background noise level to produce a mixed output signal in accordance with an exemplary embodiment. The block diagram can be implemented by the acoustic management module **201** or the processor **121**. In particular, FIG. **10** primarily illustrates the selection of microphone filters based on the background noise level. The microphone filters are used to condition the external and internal microphone signals before mixing.

As shown, the filter selection module **1045** can select one or more filters to apply to the microphone signals before mixing. For instance, the filter selection module **1045** can apply an ASM filter **1048** to the ASM signal **1047** and an ECM filter **1051** to the ECM signal **1052** based on the background noise level **1042**. The ASM and ECM filters can be retrieved from memory based on the characteristics of the background noise. An operating mode **1046** can determine whether the ASM and ECM filters are look-up curves **1043** from memory or filters whose coefficients are determined in real-time based on the background noise levels.

Prior to mixing with summing unit **1049** to produce output signal **1050**, the ASM signal **1047** is filtered with ASM filter **1048**, and the ECM signal **1052** is filtered with ECM filter **1051**. The filtering can be accomplished by a time-domain transversal filter (FIR-type filter), an IIR-type filter, or with frequency-domain multiplication. The filter can be adaptive (i.e. time variant), and the filter coefficients can be updated on a frame-by-frame basis depending on the BNL. The filter coefficients for a particular BNL can be loaded from computer memory using pre-defined filter curves **1043**, or can be calculated using a predefined algorithm **1044**, or using a combination of both (e.g. using an

interpolation algorithm to create a filter curve for both the ASM filter **1048** and ECM filter **1051** from predefined filters).

FIG. **11** is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment.

In particular, FIG. **11** shows a method **1160** for the filtering of the ECM and ASM signals using analog electronic circuitry prior to mixing. The analog circuit can process both the ECM and ASM signals in parallel; that is, the analog components apply to both the ECM and ASM signals. In one exemplary embodiment, the input audio signal **1161** (e.g., ECM signal, ASM signal) is first filtered with a fixed filter **1162**. The filter response of the fixed filter **1162** approximates a low-pass shelf filter when the input signal **1161** is an ECM signal, and approximates a high-pass filter when the input signal **1161** is an ASM signal. In an alternate exemplary embodiment, the filter **1162** is a unity-pass filter (i.e. no spectral attenuation) and the gain units G1, G2 etc instead represent different analog filters. As illustrated, the gains are fixed, though they may be adapted in other embodiments. Depending on the BNL **1169**, the filtered signal is then subjected to one of three gains; G1 **1163**, G2 **1164**, or G3 **1165**. (The analog circuit can include more or less than the number of gains shown.)

For low BNLs (e.g. when BNL<L **1170**, where L1 is a predetermined level threshold **1171**), a G1 is determined for both the ECM signal and the ASM signal. The gain G1 for the ECM signal is approximately zero; i.e. no ECM signal would be present in the output signal **1175**. For the ASM input signal, G1 would be approximately unity for low BNL.

For medium BNLs (e.g. when BNL<L2 **1172**, where L2 is a predetermined level threshold **1173**), a G2 is determined for both the ECM signal and the ASM signal. The gain G2 for the ECM signal and the ASM signal is approximately the same. In another embodiment, the gain G2 can be frequency dependent so as to emphasize low frequency content in the ECM and emphasize high frequency content in the ASM signal in the mix. For high BNL; G3 **1165** is high for the ECM signal, and low for the ASM signal. The switches **1166**, **1167**, and **1168** ensure that only one gain channel is applied to the ECM signal and ASM signal. The gain scaled ASM signal and ECM signal are then summed at junction **1174** to produce the mixed output signal **1175**.

Examples of filter response curves for three different BNL are shown in FIG. **12**, which is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL).

The basic trend for the ASM and ECM filter response at different BNLs is that at low BNLs (e.g. <60 dBA), the ASM signal is primarily used for voice communication. At medium BNL; ASM and ECM are mixed in a ratio depending on the BNL, though the ASM filter can attenuate low frequencies of the ASM signal, and attenuate high frequencies of the ECM signal. At high BNL (e.g. >85 dB), the ASM filter attenuates most al the low frequencies of the ASM signal, and the ECM filter attenuates most all the high frequencies of the ECM signal. In another embodiment of the Acoustic Management System, the ASM and ECM filters may be adjusted by the spectral profile of the background noise measurement. For instance, if there is a large Low Frequency noise in the ambient sound field of the user, then the ASM filter can reduce the low-frequencies of the ASM signal accordingly, and boost the low-frequencies of the ECM signal using the ECM filter.

Where applicable, the present embodiments of the invention can be realized in hardware, software or a combination of hardware and software. Any kind of computer system or other apparatus adapted for carrying out the methods described herein are suitable. A typical combination of hardware and software can be a mobile communications device with a computer program that, when being loaded and executed, can control the mobile communications device such that it carries out the methods described herein. Portions of the present method and system may also be embedded in a computer program product, which comprises all the features enabling the implementation of the methods described herein and which when loaded in a computer system, is able to carry out these methods.

While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

We claim:

1. A device, comprising:

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

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

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

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

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

a memory that stores instructions; and

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

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

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

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

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

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

sending a modified mixed signal to the speaker.

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

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

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

sending a modified ambient signal to the speaker.

4. The device according to claim 3, where the modified ambient signal is the ambient signal.

5. The device according to claim 3, wherein the operations further comprise:

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

6. The device according to claim 3, wherein the operations further comprise:

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

7. The device according to claim 1, wherein the modified mixed signal is the mixed signal.

8. The device according to claim 1, wherein the operations further comprise:

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

9. The device according to claim 8, wherein the operations further comprise:

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

10. The device according to claim 9, wherein the operations further comprise:

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

11. The device according to claim 9, wherein the operations further comprise:

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

* * * * *

Exhibit G

US011665493B2

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

(10) **Patent No.: US 11,665,493 B2**
(45) **Date of Patent: May 30, 2023**

(54) **ACOUSTIC SEALING ANALYSIS SYSTEM**

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H04R 1/10 (2006.01)
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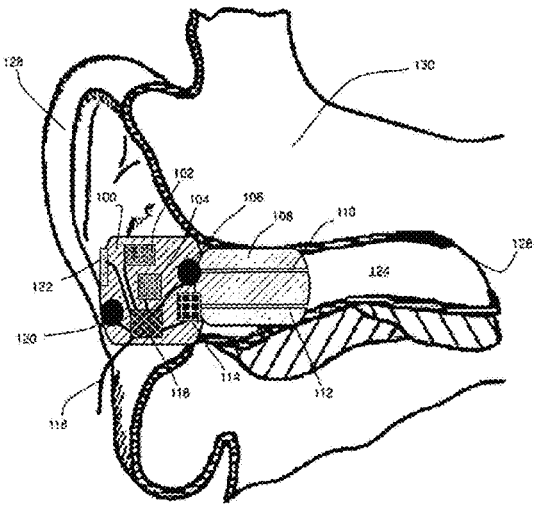
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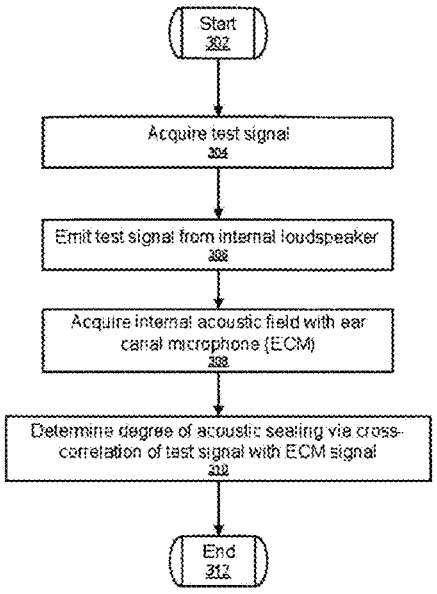
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(57) **ABSTRACT**

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

20 Claims, 15 Drawing Sheets





Related U.S. Application Data

continuation of application No. 16/414,136, filed on May 16, 2019, now Pat. No. 10,701,499, which is a continuation of application No. 15/700,511, filed on Sep. 11, 2017, now Pat. No. 10,299,053, which is a continuation of application No. 14/827,332, filed on Aug. 17, 2015, now Pat. No. 9,781,530, which is a continuation of application No. 14/054,015, filed on Oct. 15, 2013, now Pat. No. 9,113,267, which is a division of application No. 12/555,864, filed on Sep. 9, 2009, now Pat. No. 8,600,067.

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See application file for complete search history.

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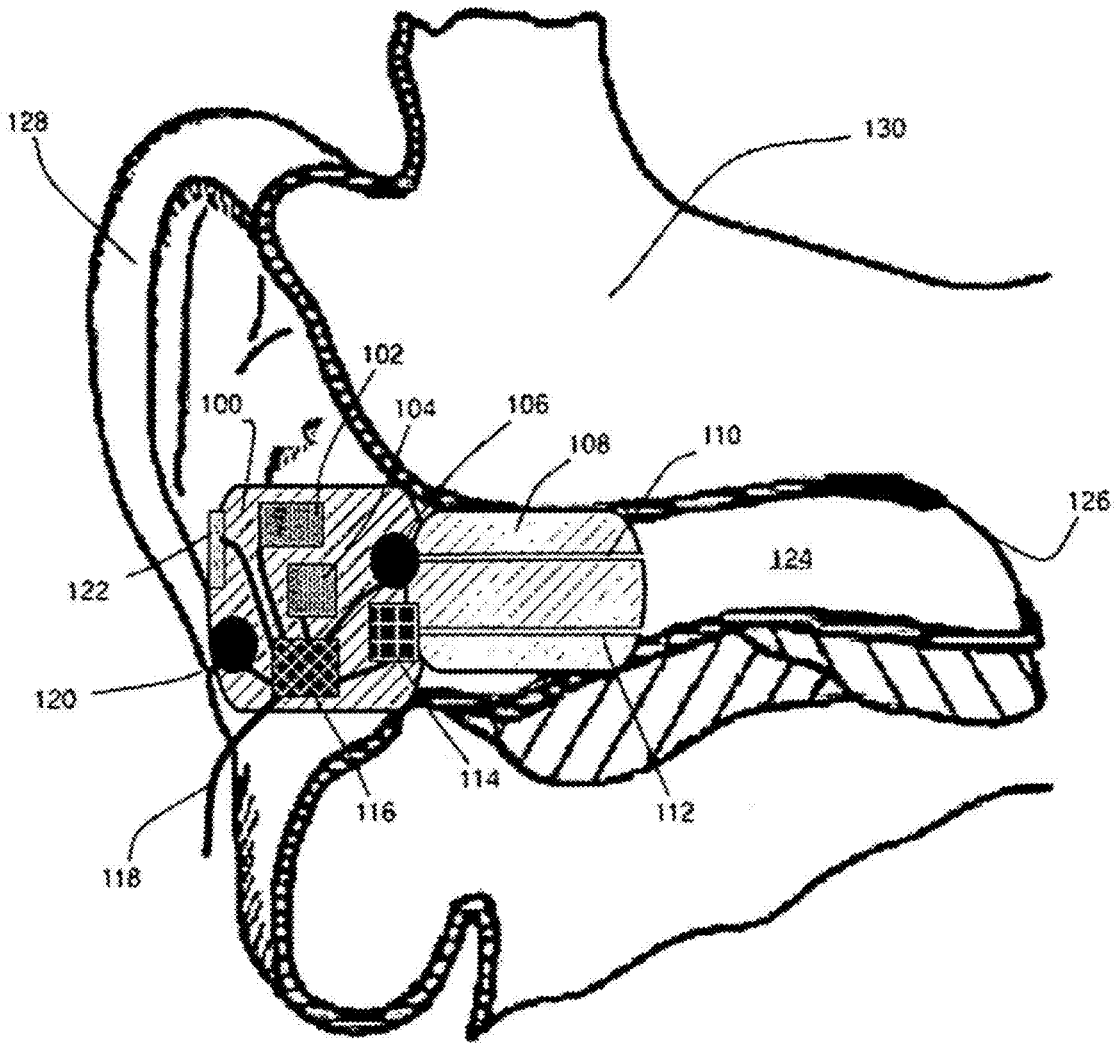
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Samsung Electronics Co., Ltd., and Samsung Electronics, America, Inc., v. *Staton Techiya, LLC*, IPR2022-00410, Feb. 18, 2022.
U.S. Appl. No. 90/015,146, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., Request for Ex Parte Reexamination of U.S. Pat. No. 10,979,836.

* cited by examiner

FIG 1



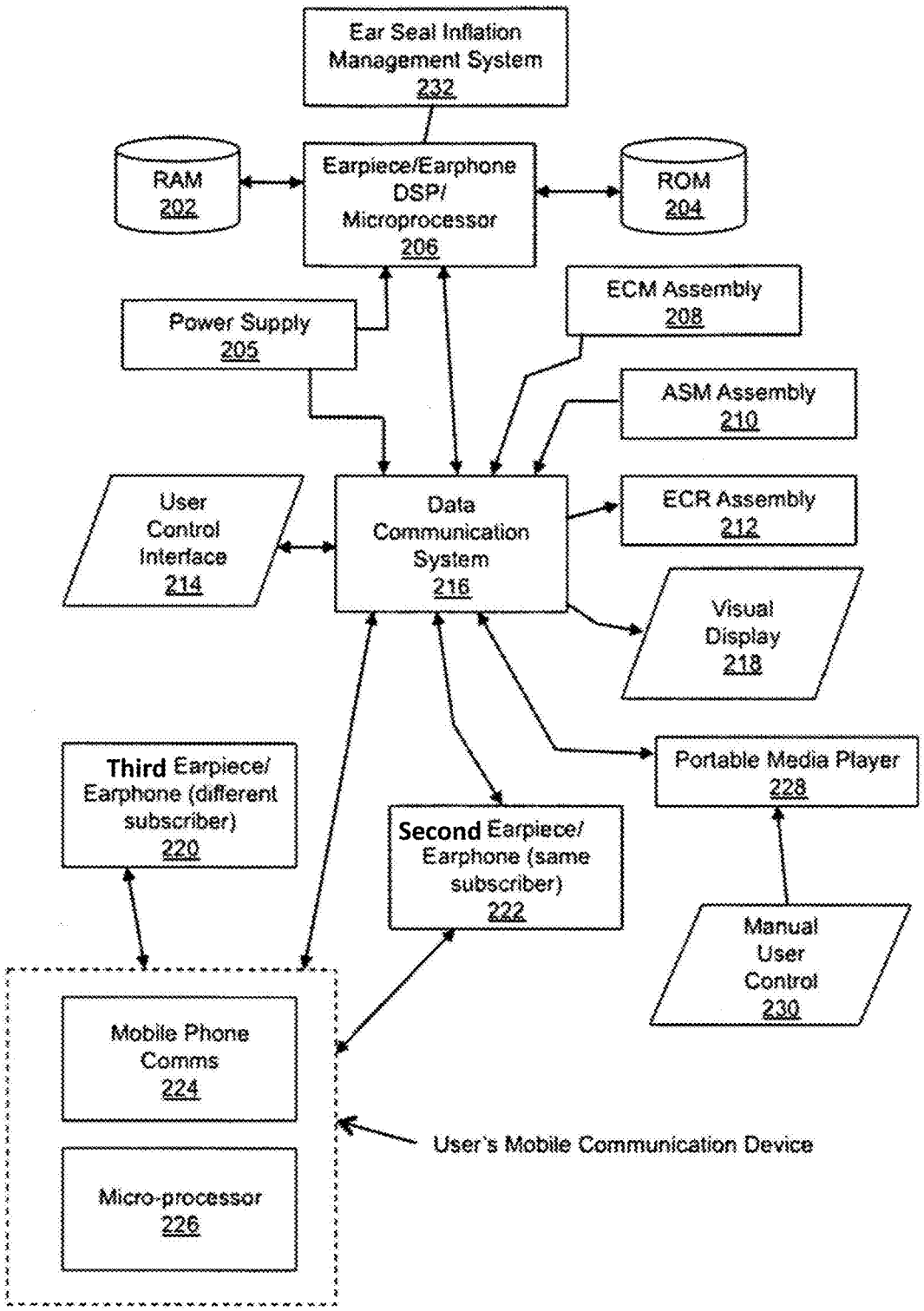

Ear Seal Inflation Management System 232
RAM 202
Earpiece/Earphone DSP/ Microprocessor 206
ROM 204
Power Supply 205
ECM Assembly 208
ASM Assembly 210
User Control Interface 214
Data Communication System 216
ECR Assembly 212
Visual Display 218
Third Earpiece/ Earphone (different subscriber) 220
Second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

FIG. 2

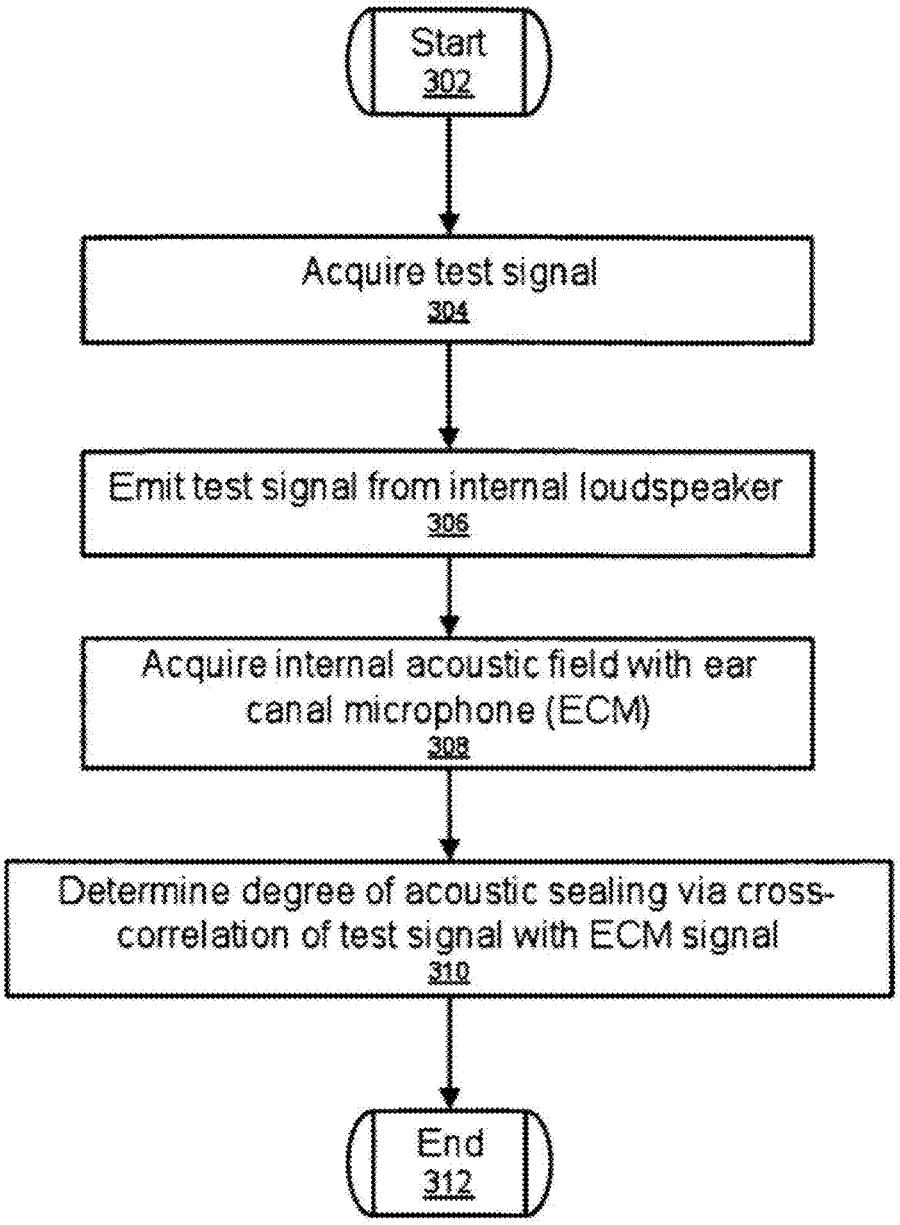

Start
302
Acquire test signal
304
Emit test signal from internal loudspeaker
306
Acquire internal acoustic field with ear canal microphone (ECM)
308
Determine degree of acoustic sealing via cross-correlation of test signal with ECM signal
310
End
312

FIG. 3

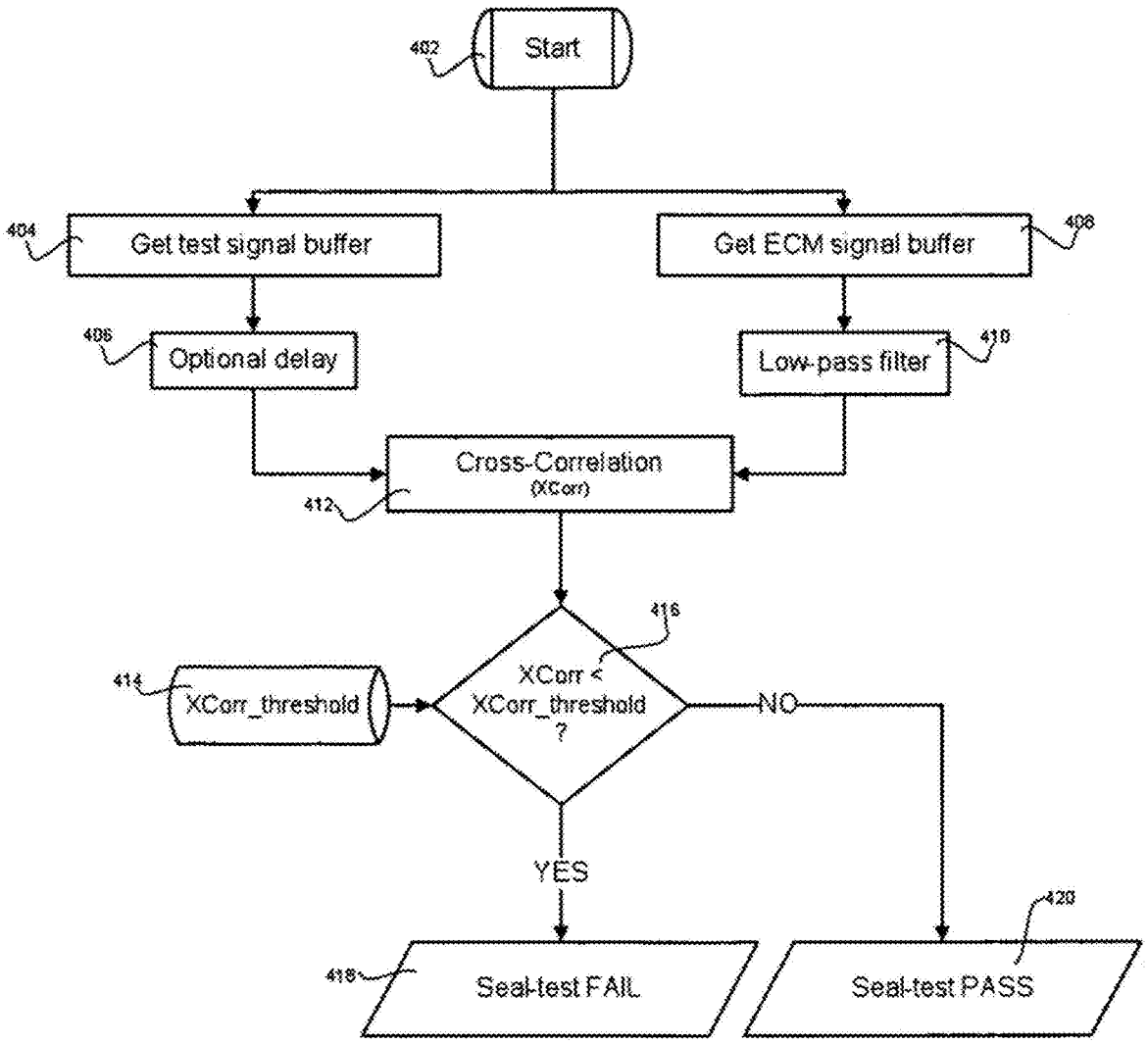

402
Start
404
Get test signal buffer
406
Optional delay
408
Get ECM signal buffer
410
Low-pass filter
412
Cross-Correlation
(XCorr)
414
XCorr_threshold
416
XCorr < XCorr_threshold ?
NO
YES
418
Seal-test FAIL
420
Seal-test PASS

FIG. 4

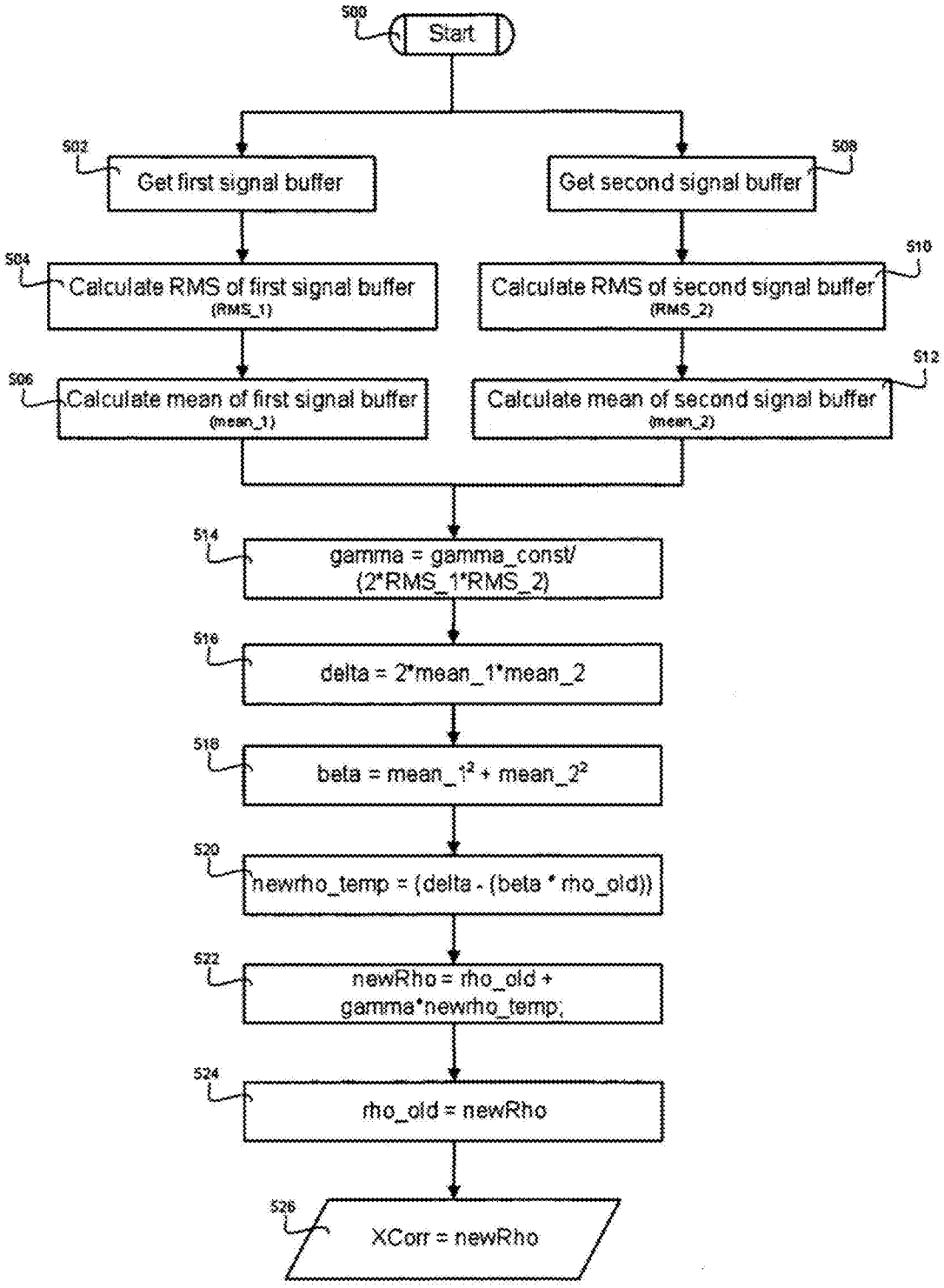

S00
Start
S02
Get first signal buffer
S08
Get second signal buffer
S04
Calculate RMS of first signal buffer
(RMS_1)
S10
Calculate RMS of second signal buffer
(RMS_2)
S06
Calculate mean of first signal buffer
(mean_1)
S12
Calculate mean of second signal buffer
(mean_2)
S14
gamma = gamma_const/
(2*RMS_1*RMS_2)
S16
delta = 2*mean_1*mean_2
S18
beta = mean_1² + mean_2²
S20
newrho_temp = (delta - (beta * rho_old))
S22
newRho = rho_old +
gamma*newrho_temp;
S24
rho_old = newRho
S26
XCorr = newRho

FIG.5

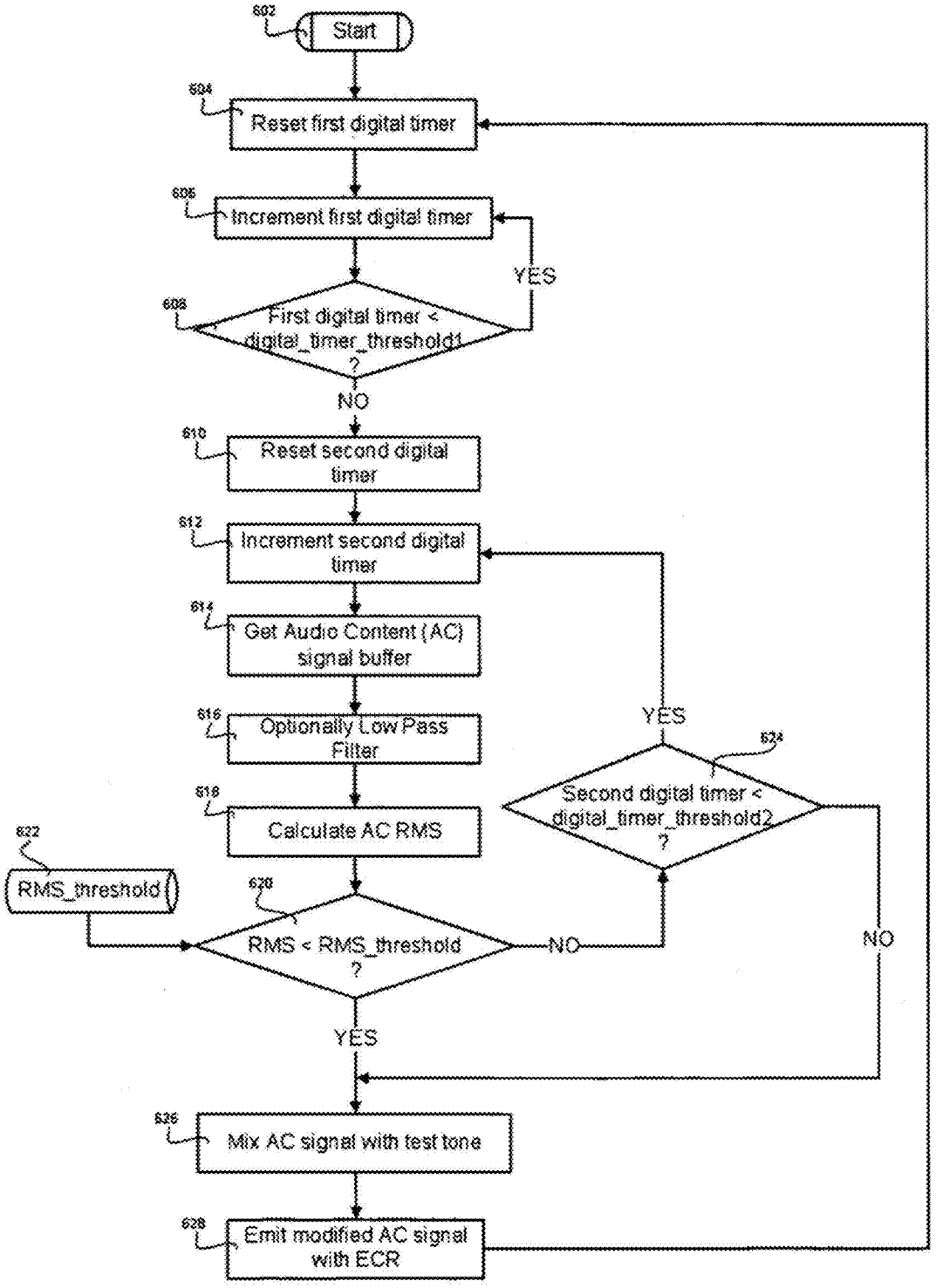

602
Start
604
Reset first digital timer
606
Increment first digital timer
YES
608
First digital timer < digital_timer_threshold1 ?
NO
610
Reset second digital timer
612
Increment second digital timer
614
Get Audio Content (AC) signal buffer
616
Optionally Low Pass Filter
618
Calculate AC RMS
622
RMS_threshold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

FIG. 6

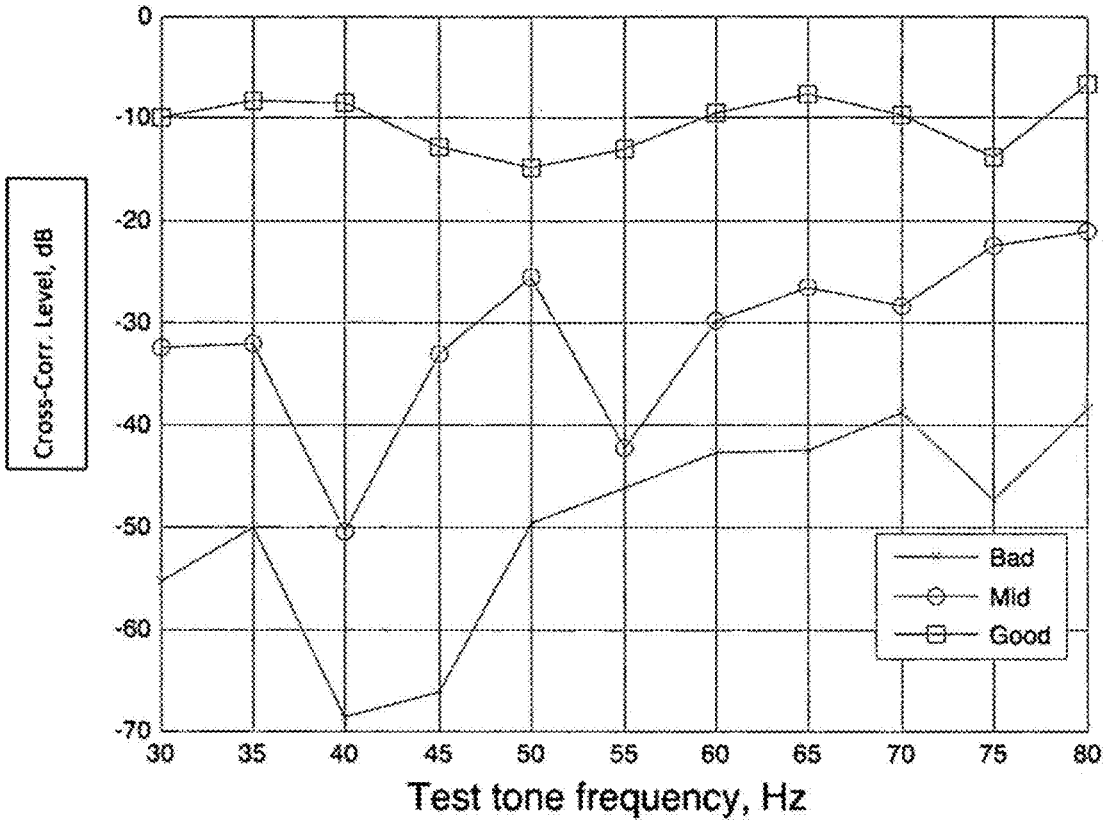

Cross-Corr. Level, dB
0
-10
-20
-30
-40
-50
-60
-70
30
35
40
45
50
55
60
65
70
75
80
Test tone frequency, Hz
Bad
Mid
Good

FIG. 7

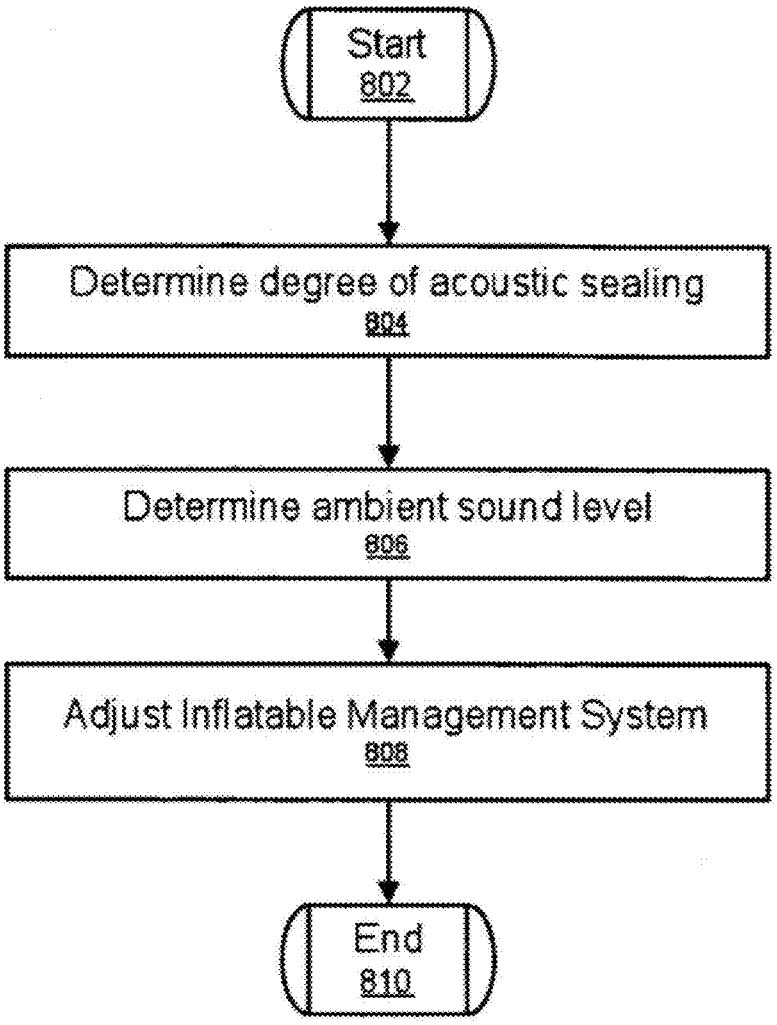

Start
802
Determine degree of acoustic sealing
804
Determine ambient sound level
806
Adjust Inflatable Management System
808
End
810

FIG. 8

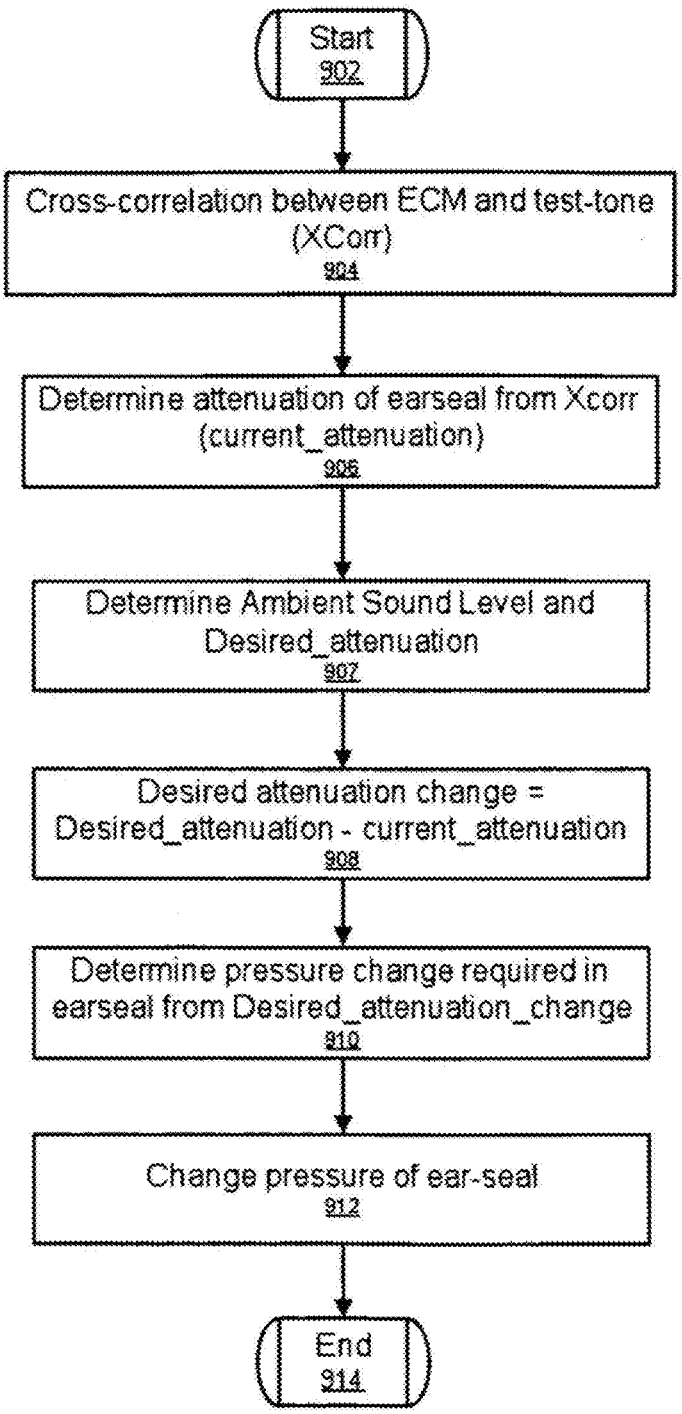

Start
902
Cross-correlation between ECM and test-tone (XCorr)
904
Determine attenuation of earseal from Xcorr (current_attenuation)
906
Determine Ambient Sound Level and Desired_attenuation
907
Desired attenuation change = Desired_attenuation - current_attenuation
908
Determine pressure change required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

FIG. 9

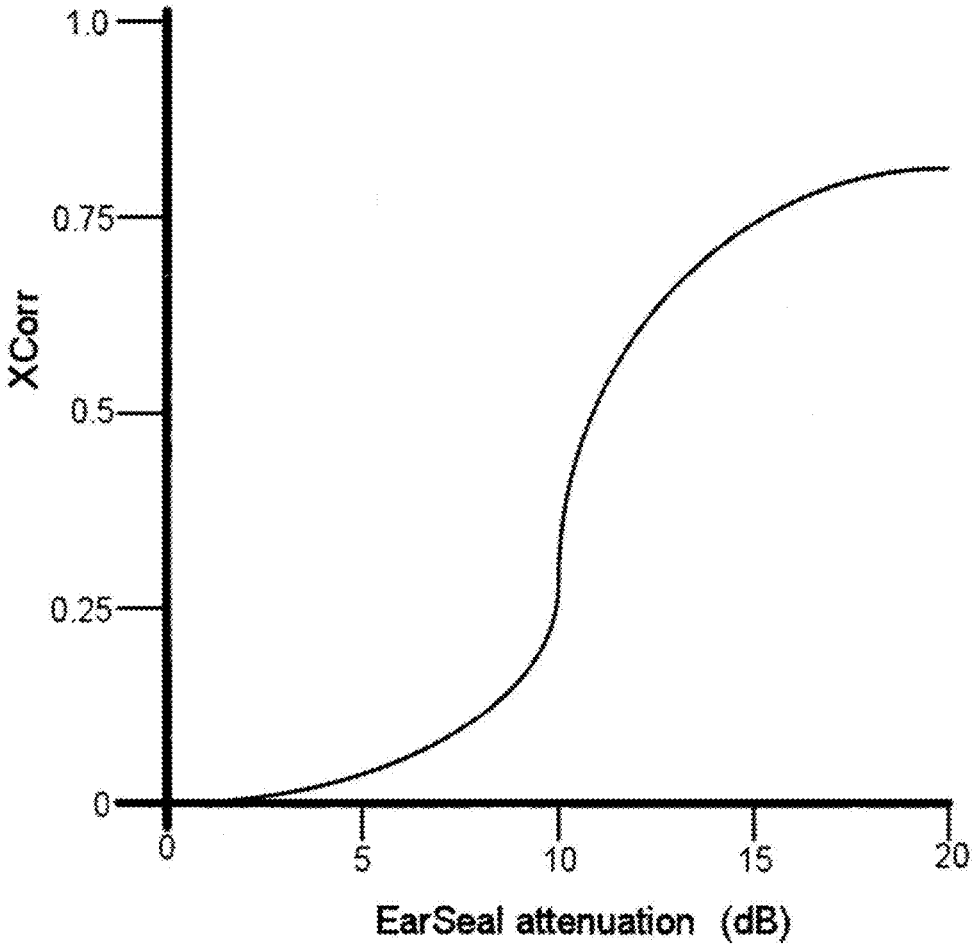


FIG. 10

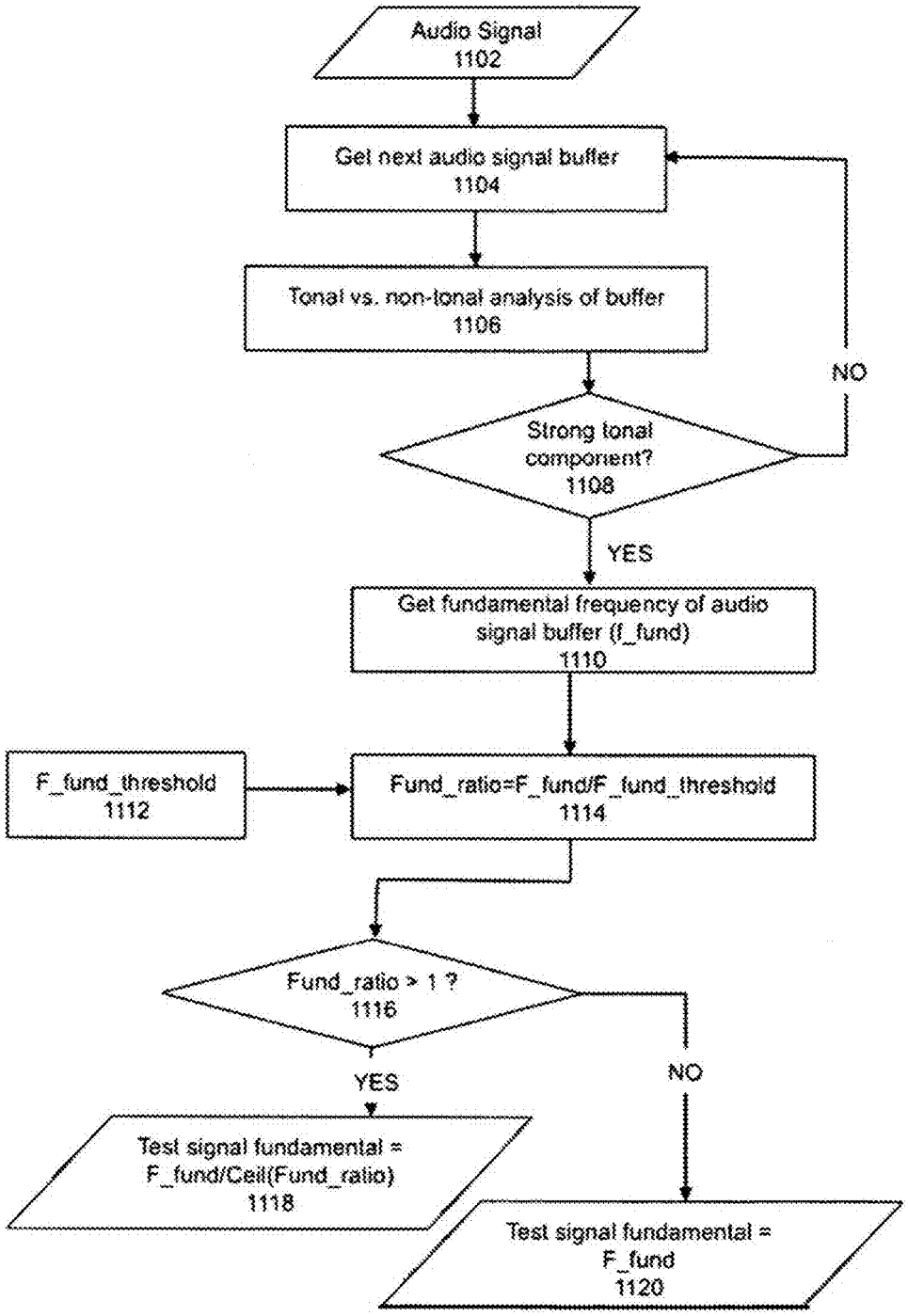

Audio Signal
1102
Get next audio signal buffer
1104
Tonal vs. non-tonal analysis of buffer
1106
Strong tonal component?
1108
NO
YES
Get fundamental frequency of audio signal buffer (f_fund)
1110
F_fund_threshold
1112
Fund_ratio=F_fund/F_fund_threshold
1114
Fund_ratio > 1 ?
1116
YES
NO
Test signal fundamental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

FIG. 11

Fig. 12

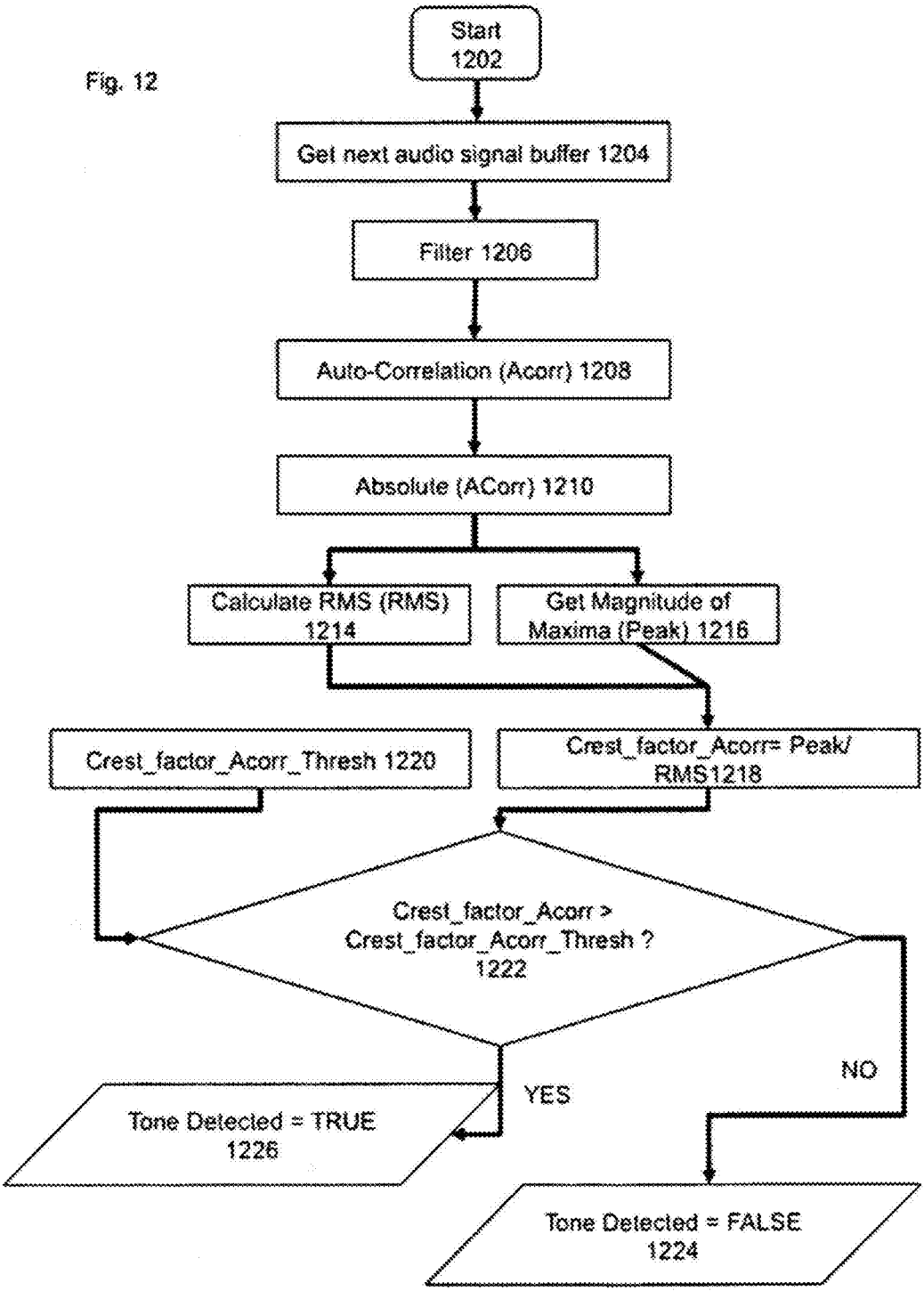

Start
1202
Get next audio signal buffer 1204
Filter 1206
Auto-Correlation (Acorr) 1208
Absolute (ACorr) 1210
Calculate RMS (RMS)
1214
Get Magnitude of
Maxima (Peak) 1216
Crest_factor_Acorr_Thresh 1220
Crest_factor_Acorr= Peak/
RMS1218
Crest_factor_Acorr >
Crest_factor_Acorr_Thresh ?
1222
YES
NO
Tone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

Fig. 13

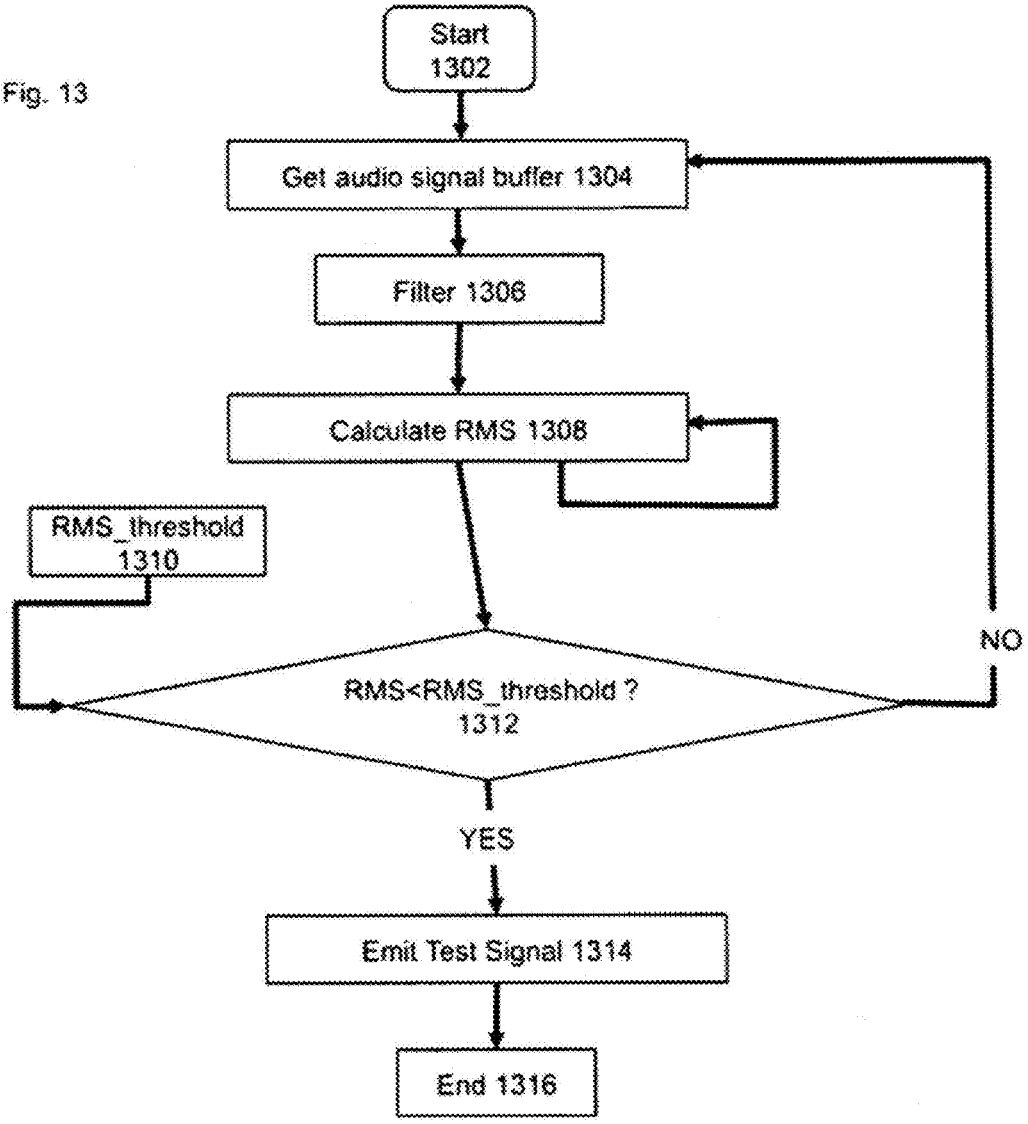

Start
1302
Get audio signal buffer 1304
Filter 1306
Calculate RMS 1308
RMS_threshold
1310
RMS<RMS_threshold ?
1312
NO
YES
Emit Test Signal 1314
End 1316

Fig. 14

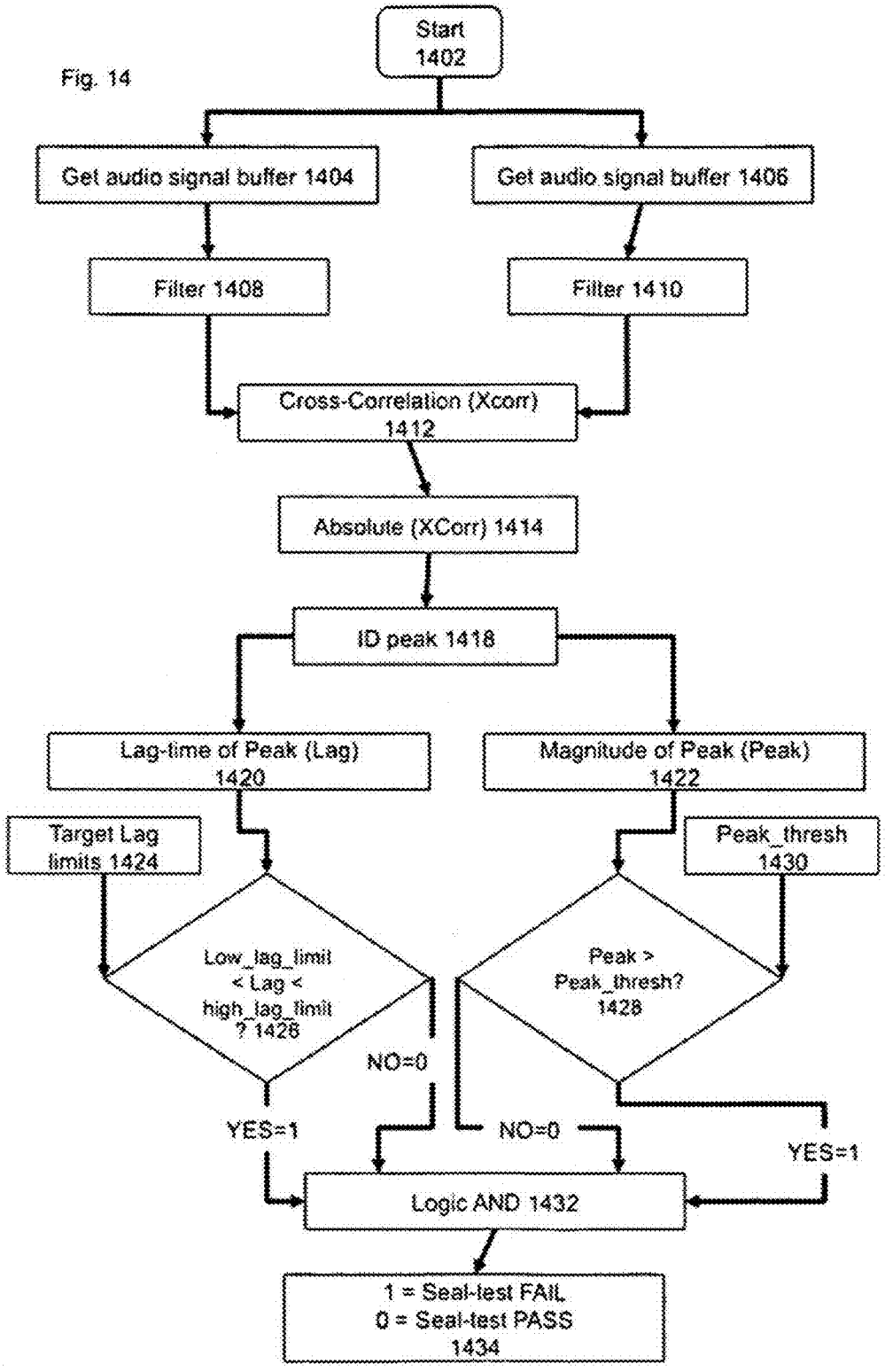

Start
1402
Get audio signal buffer 1404
Get audio signal buffer 1406
Filter 1408
Filter 1410
Cross-Correlation (Xcorr)
1412
Absolute (XCorr) 1414
ID peak 1418
Lag-time of Peak (Lag)
1420
Magnitude of Peak (Peak)
1422
Target Lag
limits 1424
Peak_thresh
1430
Low_lag_limit
< Lag <
high_lag_limit
? 1428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

FIG. 15A

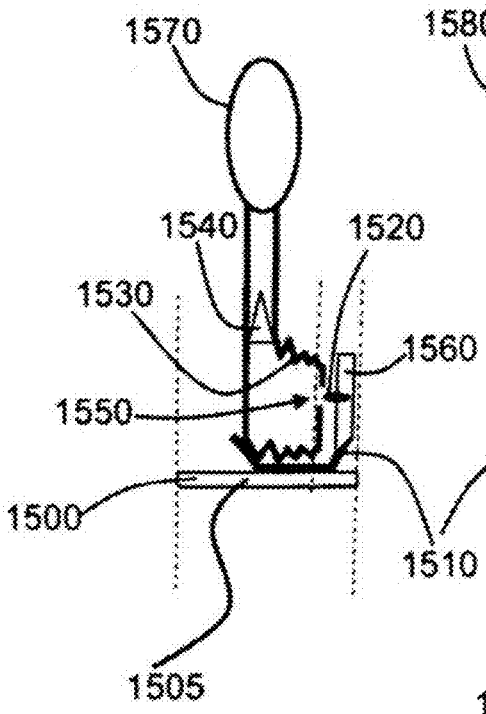


FIG. 15B

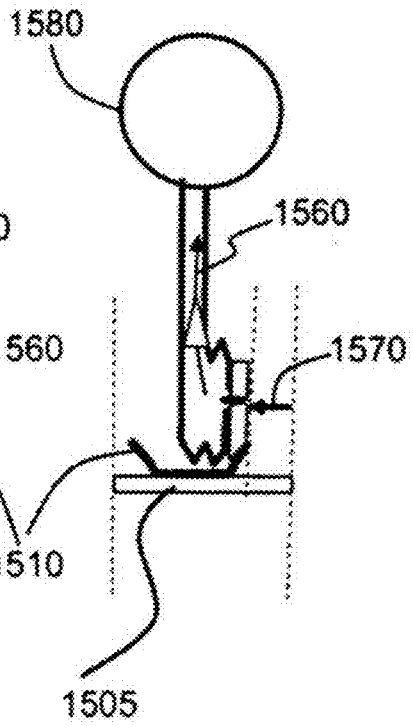
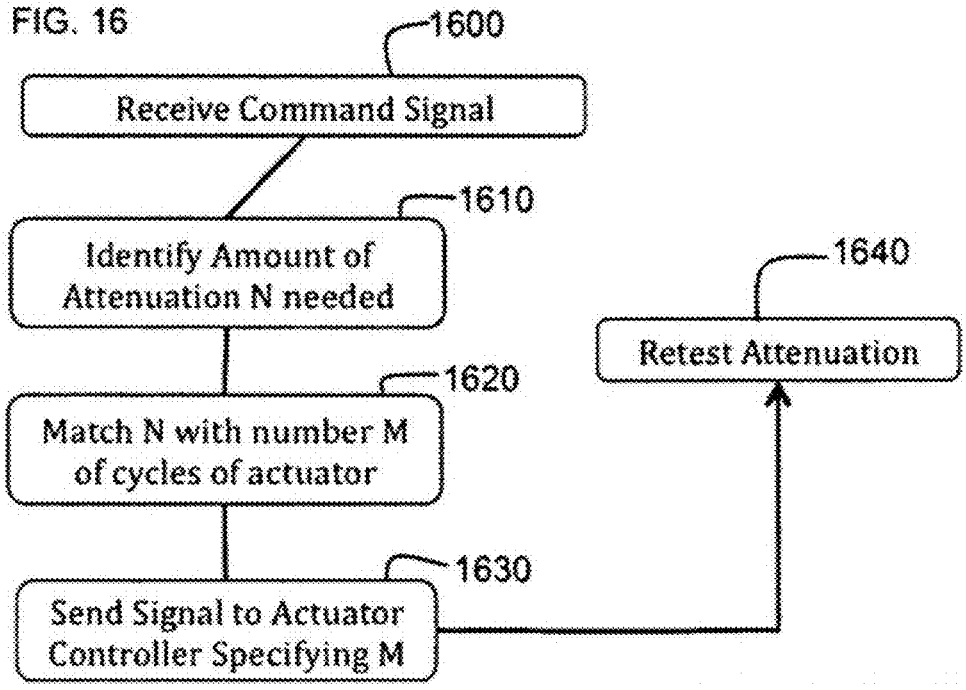


FIG. 16



ACOUSTIC SEALING ANALYSIS SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

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

FIELD OF THE INVENTION

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

BACKGROUND OF THE INVENTION

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

Although audio processing technologies can adequately suppress noise, the earpiece is generally sound agnostic and cannot differentiate sounds. Thus, one method to prevent ambient sound from entering the ear is to seal or provide an acoustic barrier at the opening of the ear canal. Sealing minimizes ambient sound leakage into the ear canal, and under the correct conditions can provide a level of noise suppression under high background noise conditions. Certain types of acoustic software (e.g., communication in a noisy environment via an ear canal microphone) may require some minimum noise isolation from the ambient sound to provide adequate performance to the user. Additionally, user conditions may change substantially during the operation of the earpiece, and in some circumstances, the earpiece may become misaligned or may be fit incorrectly such that it is not sealed correctly. A method of seal detection is needed to optimize performance.

SUMMARY OF THE INVENTION

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

In one embodiment, the device can include a sealing section forming an acoustic barrier between a first volume and a second volume. An ear canal receiver (ECR) can be configured to generate an acoustic signal in the first volume. An Ear Canal Microphone (ECM) in the first volume can be configured to measure the acoustic signal in the first volume.

The first acoustic signal emitted by the ECR can be cross-correlated with the first acoustic signal detected with the ECM to determine if the sealing section is sealed properly.

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

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

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

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

BRIEF DESCRIPTION OF THE DRAWINGS

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

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

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

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

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

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

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

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

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

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

FIG. **10** illustrates a sample relationship between EarSeal attenuation and XCorr in accordance with at least one exemplary embodiment;

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

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

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

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

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

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

DETAILED DESCRIPTION

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

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

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

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

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

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

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

Sealing unit **108** is an acoustic barrier having a first side corresponding to ear canal **124** and a second side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit **108** includes an ear canal microphone tube **110** and an ear canal receiver tube **112**. Sealing unit **108** creates a closed cavity of approximately 5 cc or less between the first side of sealing unit **108** and the tympanic membrane **126** in ear canal **124**. In at least one exemplary embodiment the sealing facilitates using the ECR (speaker) **114** to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum **126** resulting from the sound field at the entrance to the ear canal **124**. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

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

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

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

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

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

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

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

In general, data communication system **216** is a communication pathway to components of the earpiece **201** and components external to the earpiece **201**. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system **216** is configured to communicate with ECM **208**, ASM **210**, visual display **218**, and user control interface **214** of the earpiece **201**. As shown, user control interface **214** can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system **216** is capable of communication to devices exterior to the earpiece **201** such as the user's mobile phone **234**, a second earpiece **222**, and a portable media player **228**. Portable media player **228** can be controlled by a manual user control **230**.

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

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

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

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

detecting an event for transmitting the acute sound to the ear canal of the user. The processor **206** by way of the ECM **208** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system **216**.

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

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

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

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

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

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

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

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

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

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

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

As shown, electronic housing unit **100** of the earpiece is adjacent to the second side of sealing unit **108**. Electronic housing unit **100** is exposed to the ambient environment and for purposes of acoustic sealing analysis is considered the second side of sealing unit **108**. Electronic housing unit **100** includes Ambient Sound Microphone **120** (ASM **120**), which is configured to measure sounds in the ambient

environment. Thus, ASM **120** receives and measures an ambient signal corresponding to a signal incident on the second side of sealing unit **108**. ASM **120** provides the measured ambient signal to processor **116**.

Ideally, sealing unit **108** is an acoustic barrier preventing the test signal or very little of the test signal from getting past sealing unit **108** and into the ambient environment. Conversely, sealing unit **108** if improperly sealed will pass some of the test signal. Processor **116** compares the test signal to the signal provided by ECM **106** corresponding to the acoustic signal in ear canal **124**. In particular, processor **116** undertakes the cross-correlation between emitted test signal and the ECM signal.

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

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

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

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

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

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

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

-continued

Where

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

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

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

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

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

$$\partial_k = 2x_k y_k$$

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

Where

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

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

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

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

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

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

1. A first signal buffer **502** is accumulated. This signal buffer corresponds to the emitted test signal (i.e. the sine wave).
2. The RMS level of the first buffer is calculated **504** (x_{RMS})
3. The mean level of the first buffer is calculated **506**.
4. A second signal buffer **508** is accumulated. This signal buffer corresponds to the filtered ECM signal.
5. The RMS level of the second buffer is calculated **510** (y_{RMS}).
6. The mean level of the second buffer is calculated **512**.
7. In step **514**, γ (gamma) is approximated as:

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

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

8. In step **516**, ∂_k (delta) calculated as twice the product of the first signal buffer mean and the second signal buffer mean.
9. In step **518**, beta is calculated as the sum of the square of the mean value of the first buffer with the sum of the square of the mean value of the second buffer.

10. In step **520**, the new temporary estimate of the correlation newRho_temp is calculated as: newRho_temp=(delta-beta*rho_old))

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

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

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

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

A test sequence is initiated in a step **602**. The previous seal test event resets the first digital timer in a step **604**. A time delay is generated by the loop comprising steps **606** and **608**. The first digital timer is time incremented in the step **606**. After each added time increment, the first digital timer is compared against a digital_timer_threshold1. The first digital timer is time incremented (after the time has advanced another increment) after the comparison in the step **608** if the first digital timer is less than the digital_timer_threshold1.

A second digital timer is reset in a step **610** when the first digital timer is greater than the digital_timer_threshold1. The second digital timer is time incremented in a step **612**. Audio content (AC) from a signal buffer is retrieved in a step **614**. The audio content can be filtered through a low pass filter in an optional step **616**. The RMS of the audio content is calculated in a step **618**. The calculated RMS of the audio content is compared against an RMS_threshold **622** in a step **620**. The second digital timer is compared against a digital timer_threshold2 in a step **624** if the RMS of the audio content is greater than the RMS_threshold. The second digital timer is time incremented (after the time has advanced another increment) when the second digital timer is less than digital_timer_threshold2 in the step of **624**.

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

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

In at least one exemplary embodiment, the test signal for testing a seal of a sealing section is less than 200 hertz. The frequency of the emitted test signal is chosen to satisfy the requirements of being able to reveal small degradations in ear seal quality. It is also beneficial if the selected test signal frequency can be acoustically masked by reproduced audio to minimize detection of the test by an earpiece user. Both of these criteria are met using a test signal frequency below 200 Hz. The sensitivity is highest from the measured data at frequencies below 50 Hz. Conversely, as the test signal frequency increases the cross-correlation difference between a "good" and "bad" acoustic seal decreases. For example, with a 40 Hz test tone, the cross-correlation for a "good" ear seal is −8 dB, and for a bad ear seal it is −68 dB (i.e. a 60 dB difference). At a test signal frequency of 80 Hz, the cross-correlation for a "good" ear seal is −8 dB and for a bad ear seal it is −38 dB (i.e. a 30 dB difference). Thus, above 200 Hz the cross-correlation difference between a "good" and "bad" acoustic seal is further reduced thereby reducing the sensitivity of the test.

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

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

Referring to FIG. **9**, a more detailed flowchart to adjust the degree of acoustic sealing of an Inflation Management

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

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

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

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

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

A step **1106** determines if the buffer of audio content **1102** comprises a strong tonal signal component. Mixing the test signal having a similar fundamental frequency as audio content **1102** will mask the test signal when played to the user. Thus, the test signal is musically in harmony with the reproduced music and results in very little perceptual degradation in sound quality.

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

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

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

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

Referring to FIG. **12**, a flowchart of an exemplary embodiment to determine tonal presence in audio content is shown. In particular, the exemplary embodiment relates to step **1106** of FIG. **11** that analyzes audio content stored in a buffer. The method begins at step **1202**. A step **1204** gets the audio content stored in an audio signal buffer hereinafter called the audio signal. A filter step **1206** filters the audio signal to a frequency range of interest that relates to a sealing test frequency. For example, a band pass filter in the range of 20 Hz to 500 Hz could be used to filter the audio signal where the test signal is in the lower audio frequency range. An auto-correlation step **1208** analyzes the audio signal

where a strong tonal signal component is represented by peaks in the analysis results. A step **1210** generates Absolute (Acorr) which is a number representing the absolute magnitude of the peaks from the analysis. For example, Absolute (Acorr) can be the square of the results from the auto-correlation.

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

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

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

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

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

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

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

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

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

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

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

Magnitude of Peak is greater than the Peak_threshold indicating that the acoustic barrier is sealed improperly. All other conditions indicate a PASS output **1434** and the acoustic barrier is sealed correctly.

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

The music or speech is buffered in memory **104** or memory in processor **116**. Processor **116** analyzes the audio content in the buffer to identify a strong tonal content. A test signal can be created once audio content with strong tonal content is found. The test signal will have at least one fundamental pitch corresponding to the strong tonal content and optionally further harmonics. Processor **116** also analyzes the measured signals from ECM **106** and ASM **120** to determine when to emit the test signal. Processor **116** monitors and compares the sound level in the ambient environment and ear canal **124**. Processor **116** will provide the generated test signal to ECR **114** during an optimum time for test accuracy such as when the ambient sound level is low, the ear canal sound level is low, or both. Also, processor **116** will not output the test signal if there is audio content similar to the test signal in the ear canal or ambient environment.

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

In an exemplary embodiment, a cross-correlation is used to measure the similarity between the signals in ear canal **124** and the ambient environment. Processor **116** performs the cross-correlation calculations using the measured acoustic signals from ECM **106** and ASM **120**. In particular, the cross-correlation is used to identify and compare the acoustic test signal present in the two volumes separated by the acoustic barrier. A cross-correlation between ASM and ECM signals is defined according to the following equation (5):

$$X\ \mathrm{Corr}(n,l)=\Sigma_{n=0}^{N}\mathrm{ASM}(n)\mathrm{ECM}(n-l), \tag{5}$$

Where: 1=0, 1, 2, . . . N

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

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

$$\gamma_{xy}^2=\frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \tag{6}$$

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

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

FIGS. **15**A and **15**B illustrate a method of varying the seal of an inflation system in accordance with at least one exemplary embodiment. In the non-limiting example, when a seal is essentially detected as being low, for example the calculated sound isolation of the system is 3 dB or less, a signal is sent to a seal varying device (e.g., **1500**) to vary the seal, in this case increase (i.e. increase the sound isolation of the system) the seal value. The signal can be instructions to send a current over a period of time to an actuator, which can decrease the overall volume of the system (hence increasing the pressure and effectively the sealing). For example a slider actuator such as the P-653 PILine® can be used. Which has the dimensions of 15 mm by 11 mm by 8 mm, which includes an attached electronics control board, and has a mass of 1 gram. General operation uses 5V and about 100 mamps with a typical speed of 50 to 90 mm/sec. Note that the max force of about 0.15 N but such systems can be tailored to enable pumping beyond atmospheric pressure (e.g., can increase the max force). The non limiting example illustrated in FIGS. **15**A and **15**B shows a slider actuator **1500**, with a moving slide **1510**, having attached a pumping arm **1560**, with a hole **1550** covering bump **1520**. Upon receipt of a signal **1505** the actuator can move **1570** such that the bump **1520** covers the hole **1550** in a bellows **1530** pump. The actuation compresses the miniature bellows **1530** pushing **1590** gas through the one-way valve **1540**, upon the back stroke the bump **1520** uncovers the hole **1550** and air rushes back into the bellows **1530** for the next pump. For example if the stroke length is 2 mm and the pump arm **1560** contact area is about 9 mm 2 then each stroke moves 18 mm 3 of volume. If an inflation system **1570** needs to be inflated more (e.g., more gas to increase sealing) then each stroke

can provide an additional volume of gas of 18 mm 3 into the system increasing the inflation system **1580**. If the inflation system is initially empty (e.g., needs 1000 mm 3 of gas volume to inflate) then about 56 strokes would be needed for inflation, which is about 110 mm one direction stroke length or about 2 seconds at P-653 PILine speeds. The number of oscillations and stroke length can be determined according to the signal **1505** sent, which can be specifically tailored depending upon the electronics controlling the actuators. Note that PI-653 is an example only. Other actuation systems can be used and controlled by signal **1505**.

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

While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. For example, if words such as "orthogonal", "perpendicular" are used the intended meaning is "substantially orthogonal" and "substantially perpendicular" respectively. Additionally although specific numbers may be quoted in the claims, it is intended that a number close to the one stated is also within the intended scope, i.e. any stated number (e.g., 90 degrees) should be interpreted to be "about" the value of the stated number (e.g., about 90 degrees).

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

We claim:

1. An audio device configured to test a seal of an earphone comprising:

a memory that stores instructions; and

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

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

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

receiving a first microphone signal from the first microphone;

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

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

generating a message to convey the seal quality; and

sending the message to the user.

2. The audio device according to claim **1**, wherein the message is an acoustic message.

3. The audio device according to claim **1**, wherein the message is a visual message sent to a display of the audio device.

4. The audio device according to claim **1**, wherein the speaker plays the test signal from the first side.

5. The audio device according to claim **4**, wherein the first microphone measures sound from the first side.

6. The audio device according to claim **4**, wherein the first microphone measures sound from the second side.

7. The audio device according to claim **1**, wherein the first signal is the test signal and the second signal is the first microphone signal.

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

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

10. The audio device of claim **1**, wherein the first signal is a modified test signal and the second signal is the first microphone signal.

11. The audio device according to claim **1**, wherein the first signal is a modified test signal and the second signal is a modified first microphone signal.

12. The audio device according to claim **11**, wherein the first microphone signal is generated by applying a low pass filter to the first microphone signal, wherein the upper cutoff frequency is less than or equal to 200 Hz.

13. The audio device according to claim **1**, wherein the earphone includes a second microphone.

14. The audio device according to claim **13**, wherein the operations further comprise:

receiving a second microphone signal from the second microphone.

15. The audio device according to claim **14**, wherein the first signal is the first microphone signal and the second signal is the second microphone signal.

16. The audio device of claim **14**, wherein the first signal is a modified first microphone signal and the second signal is the second microphone signal.

17. The audio device of claim **14**, wherein the first signal is a modified first microphone signal and the second signal is a modified second microphone signal.

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

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

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

* * * * *

Exhibit H

US011610587B2

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

(10) **Patent No.: US 11,610,587 B2**
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(54) **PERSONALIZED SOUND MANAGEMENT AND METHOD**

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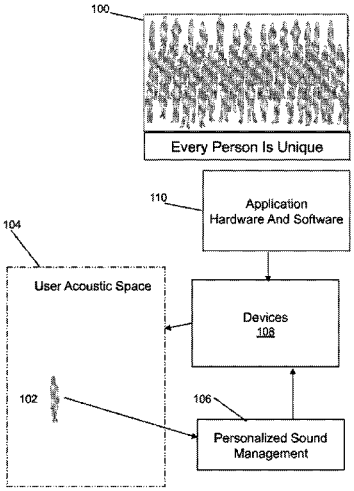
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(57) **ABSTRACT**

A personalized sound management system for an acoustic space includes at least one transducer, a data communication system, one or more processors operatively coupled to the data communication system and the at least one transducer, and a medium coupled to the one or more processors. The processors access a database of sonic signatures and display a plurality of personalized sound management applications that perform at least one or more tasks among identifying a sonic signature, calculating a sound pressure level, storing metadata related to a sonic signature, monitoring sound pressure level dosage levels, switching to an ear canal microphone in a noisy environment, recording a user's voice, storing the user's voice in a memory of an earpiece device, or storing the user's voice in a memory of a server system, or converting received text received in texts or emails to voice using text to speech conversion. Other embodiments are disclosed.

12 Claims, 15 Drawing Sheets



Related U.S. Application Data

continuation of application No. 16/671,689, filed on Nov. 1, 2019, now Pat. No. 10,997,978, which is a continuation of application No. 14/846,994, filed on Sep. 7, 2015, now Pat. No. 10,529,325, which is a continuation of application No. 12/560,097, filed on Sep. 15, 2009, now Pat. No. 9,129,291.

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See application file for complete search history.

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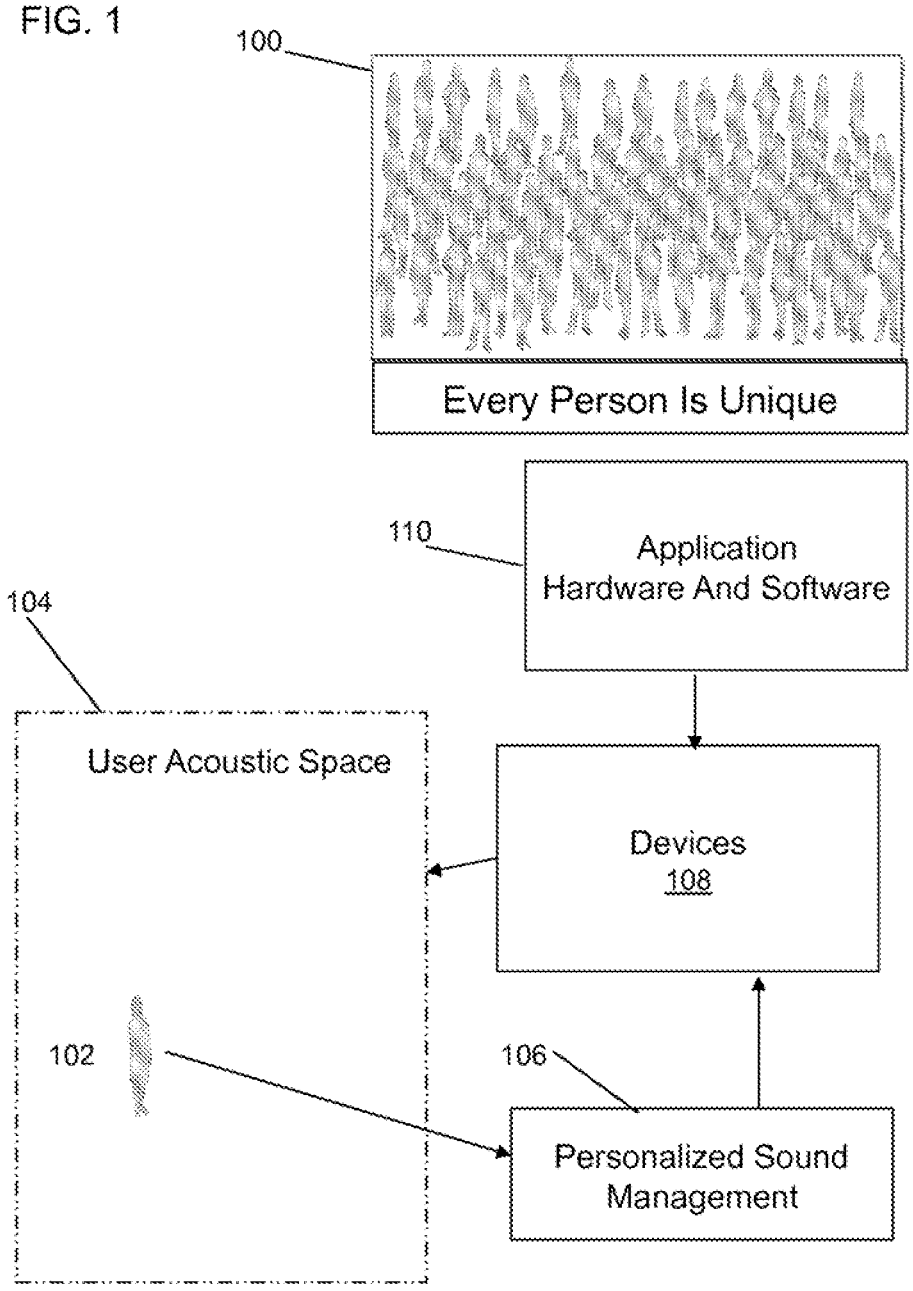
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FIG. 1


100
Every Person Is Unique
110
Application
Hardware And Software
104
User Acoustic Space
Devices
108
102
106
Personalized Sound
Management

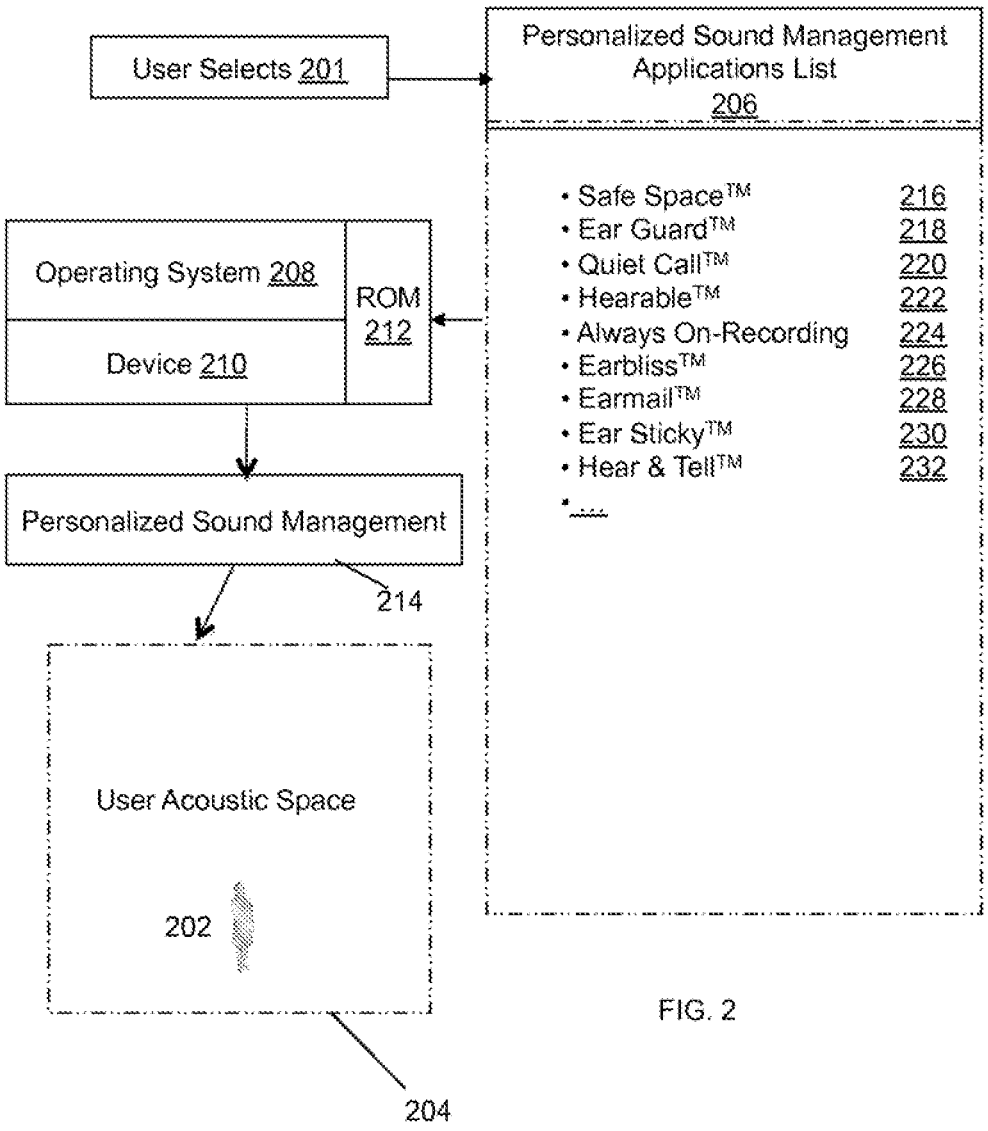

User Selects 201
Personalized Sound Management Applications List 206
• Safe Space™ 216
• Ear Guard™ 218
• Quiet Call™ 220
• Hearable™ 222
• Always On-Recording 224
• Earbliss™ 226
• Earmail™ 228
• Ear Sticky™ 230
• Hear & Tell™ 232
•....
Operating System 208
Device 210
ROM 212
Personalized Sound Management
214
User Acoustic Space
202
204

FIG. 2

FIG. 3

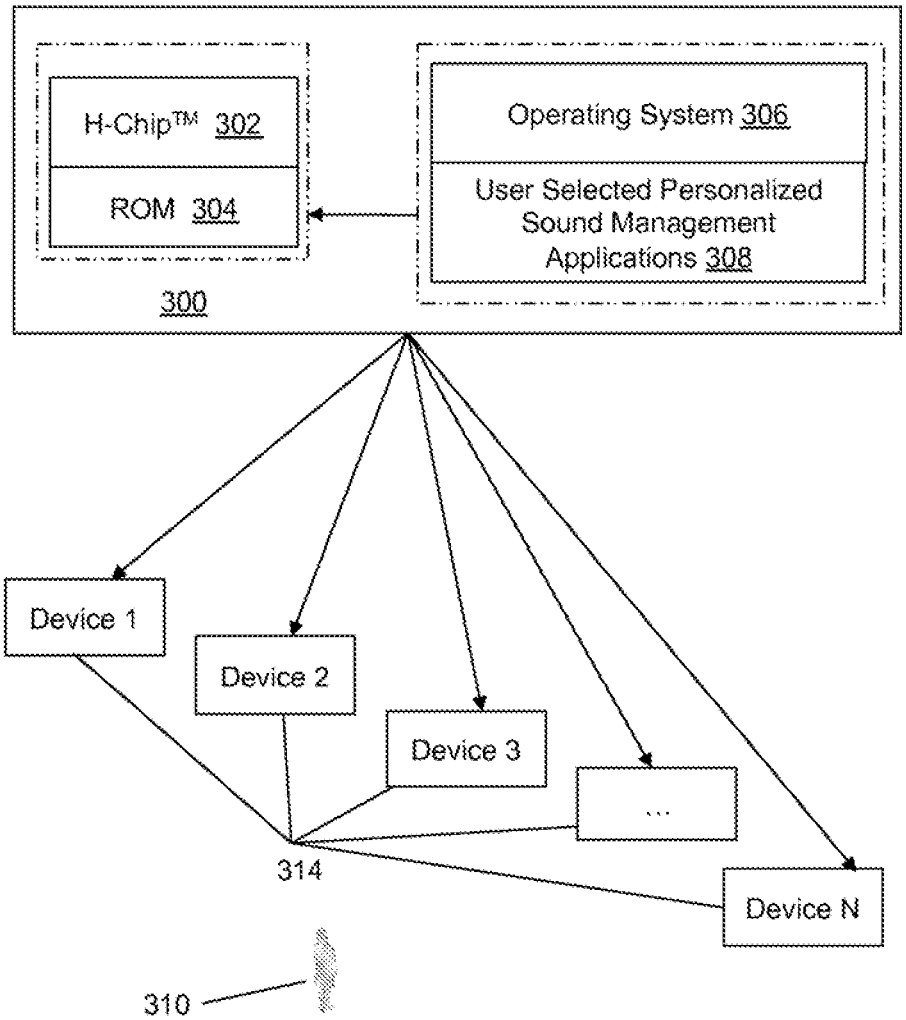

H-Chip™ 302
ROM 304
Operating System 306
User Selected Personalized Sound Management Applications 308
300
Device 1
Device 2
Device 3
...
Device N
314
310

FIG. 4

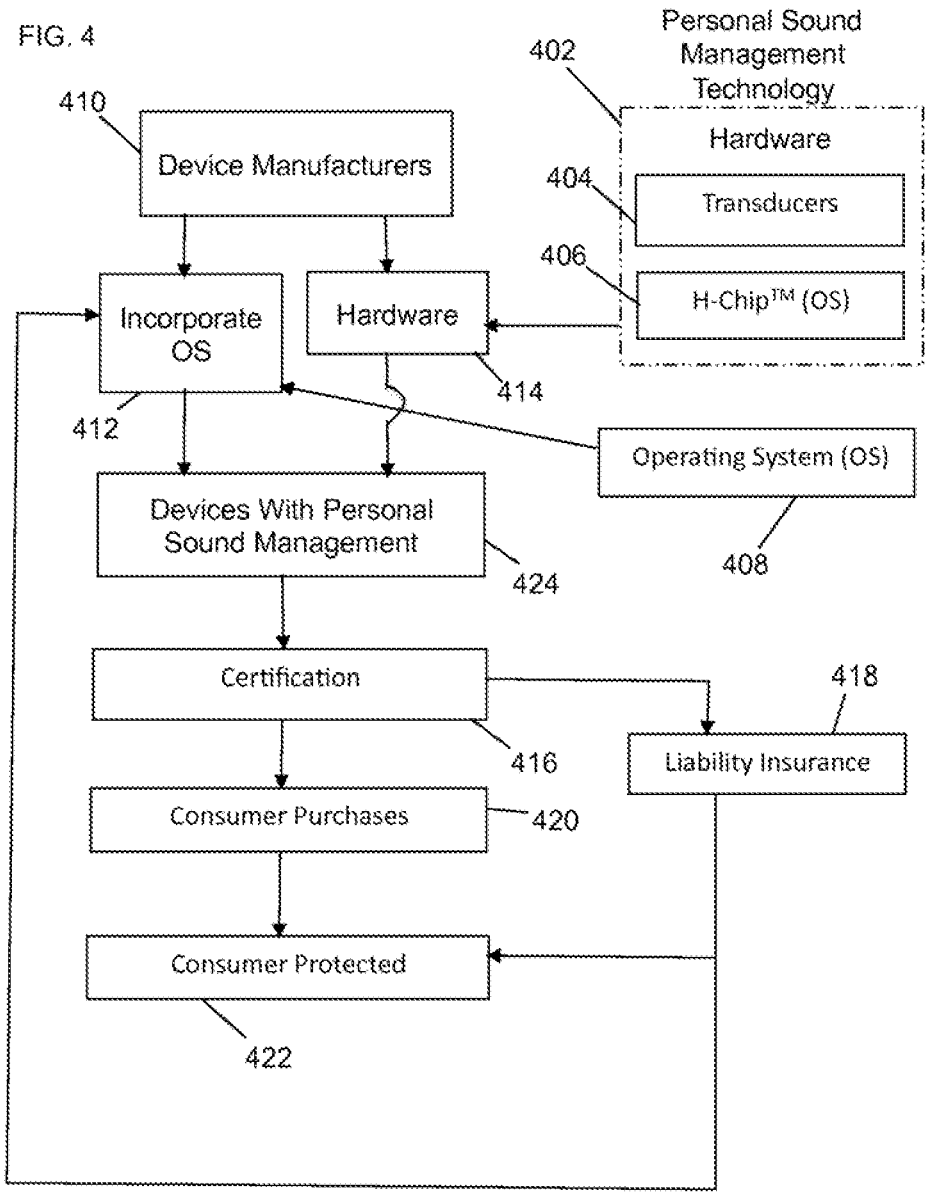
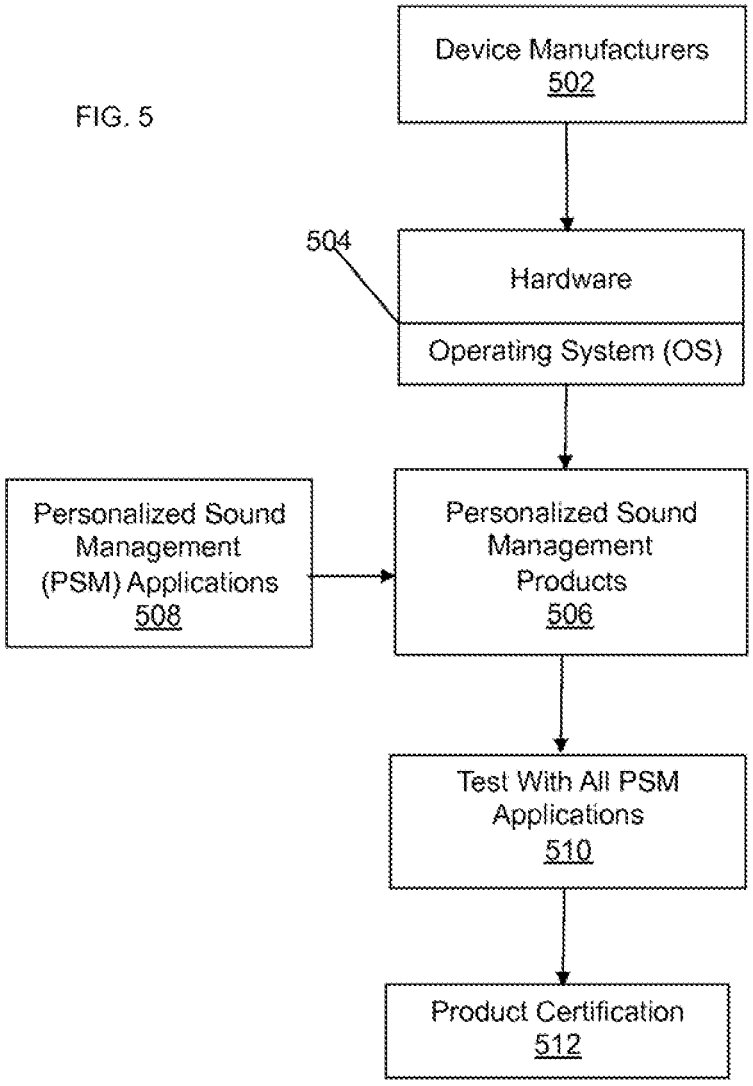

Personal Sound Management Technology
402
Hardware
404
Transducers
406
H-Chip™ (OS)
410
Device Manufacturers
Incorporate OS
Hardware
412
414
Operating System (OS)
408
Devices With Personal Sound Management
424
Certification
416
418
Liability Insurance
Consumer Purchases
420
Consumer Protected
422

FIG. 5


Device Manufacturers
502
504
Hardware
Operating System (OS)
Personalized Sound Management (PSM) Applications
508
Personalized Sound Management Products
506
Test With All PSM Applications
510
Product Certification
512

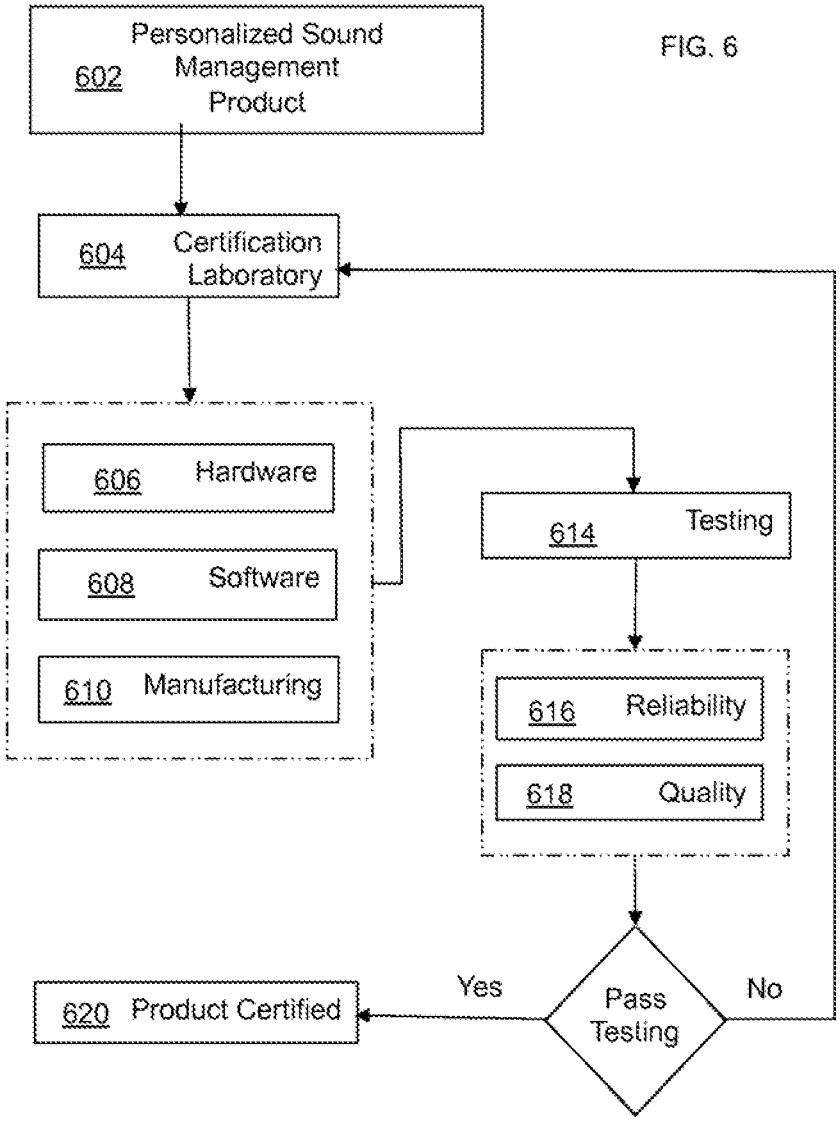
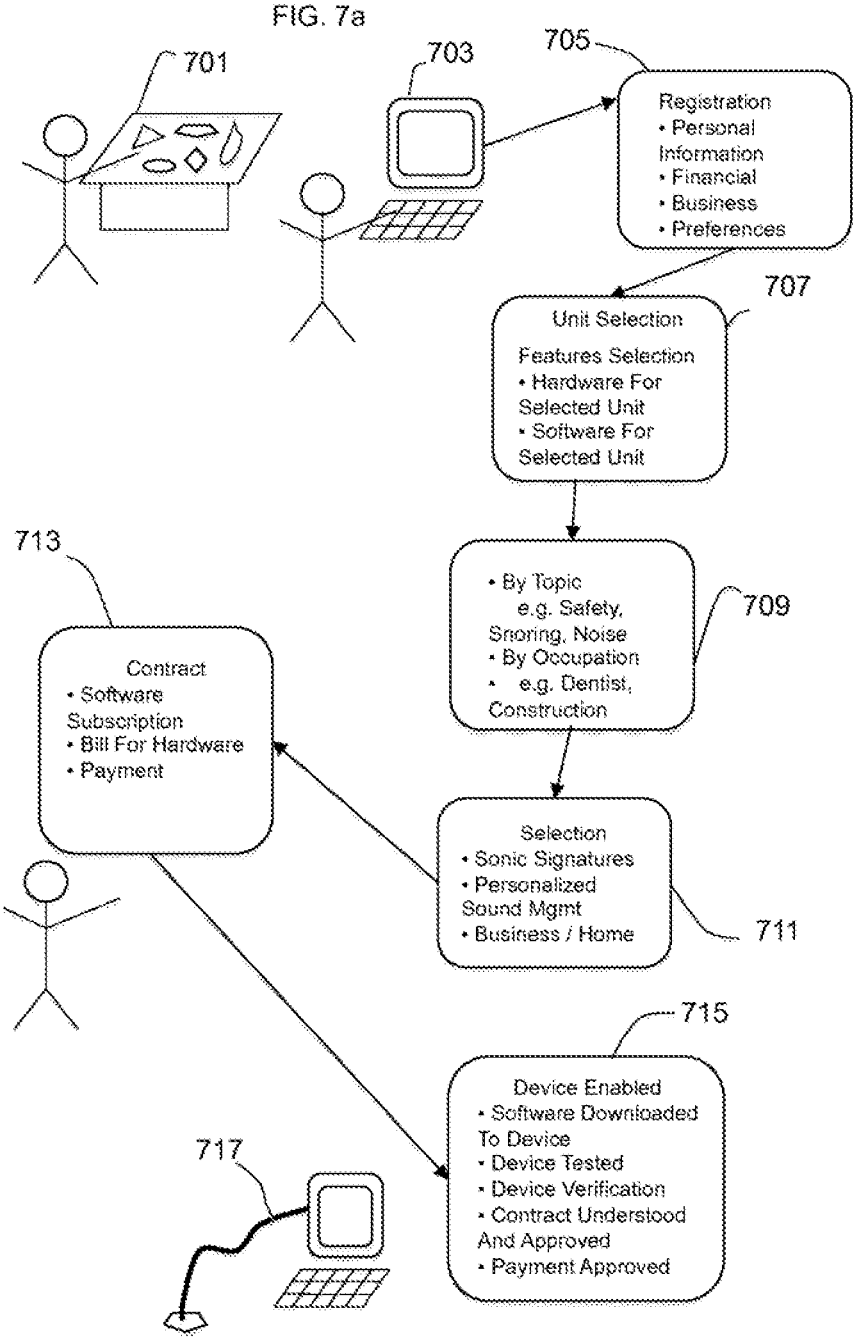

Personalized Sound Management Product
602
604 Certification Laboratory
606 Hardware
608 Software
610 Manufacturing
614 Testing
616 Reliability
618 Quality
Pass Testing
Yes
No
620 Product Certified

FIG. 6

FIG. 7a


701
703
705
Registration
• Personal Information
• Financial
• Business
• Preferences
707
Unit Selection
Features Selection
• Hardware For Selected Unit
• Software For Selected Unit
713
• By Topic
e.g. Safety, Snoring, Noise
• By Occupation
• e.g. Dentist, Construction
709
Contract
• Software Subscription
• Bill For Hardware
• Payment
Selection
• Sonic Signatures
• Personalized Sound Mgmt
• Business / Home
711
715
Device Enabled
• Software Downloaded To Device
• Device Tested
• Device Verification
• Contract Understood And Approved
• Payment Approved
717

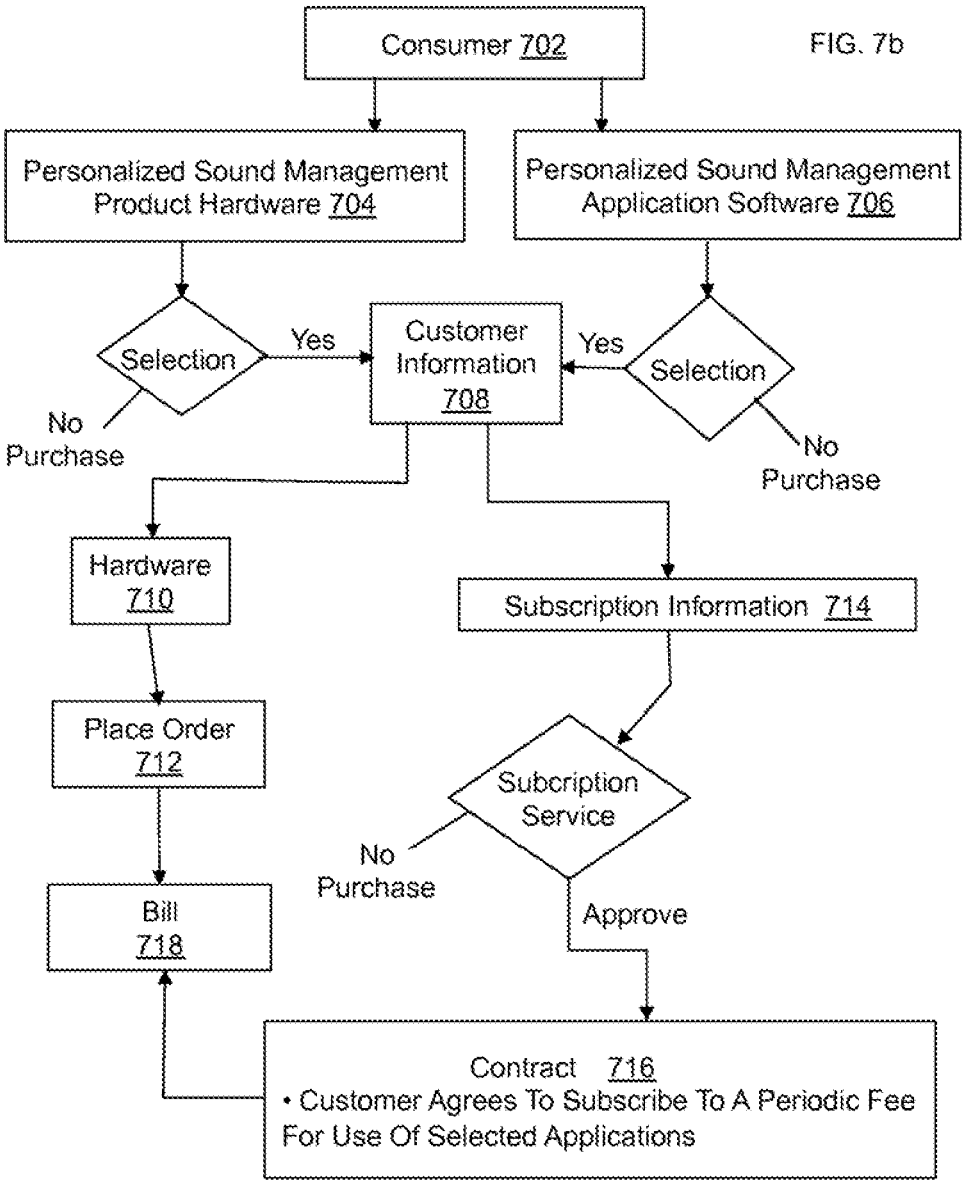

Consumer 702
Personalized Sound Management Product Hardware 704
Personalized Sound Management Application Software 706
Selection
Yes
Customer Information 708
Yes
Selection
No Purchase
No Purchase
Hardware 710
Subscription Information 714
Place Order 712
Subcription Service
No Purchase
Approve
Bill 718
Contract 716
• Customer Agrees To Subscribe To A Periodic Fee For Use Of Selected Applications

FIG. 7b

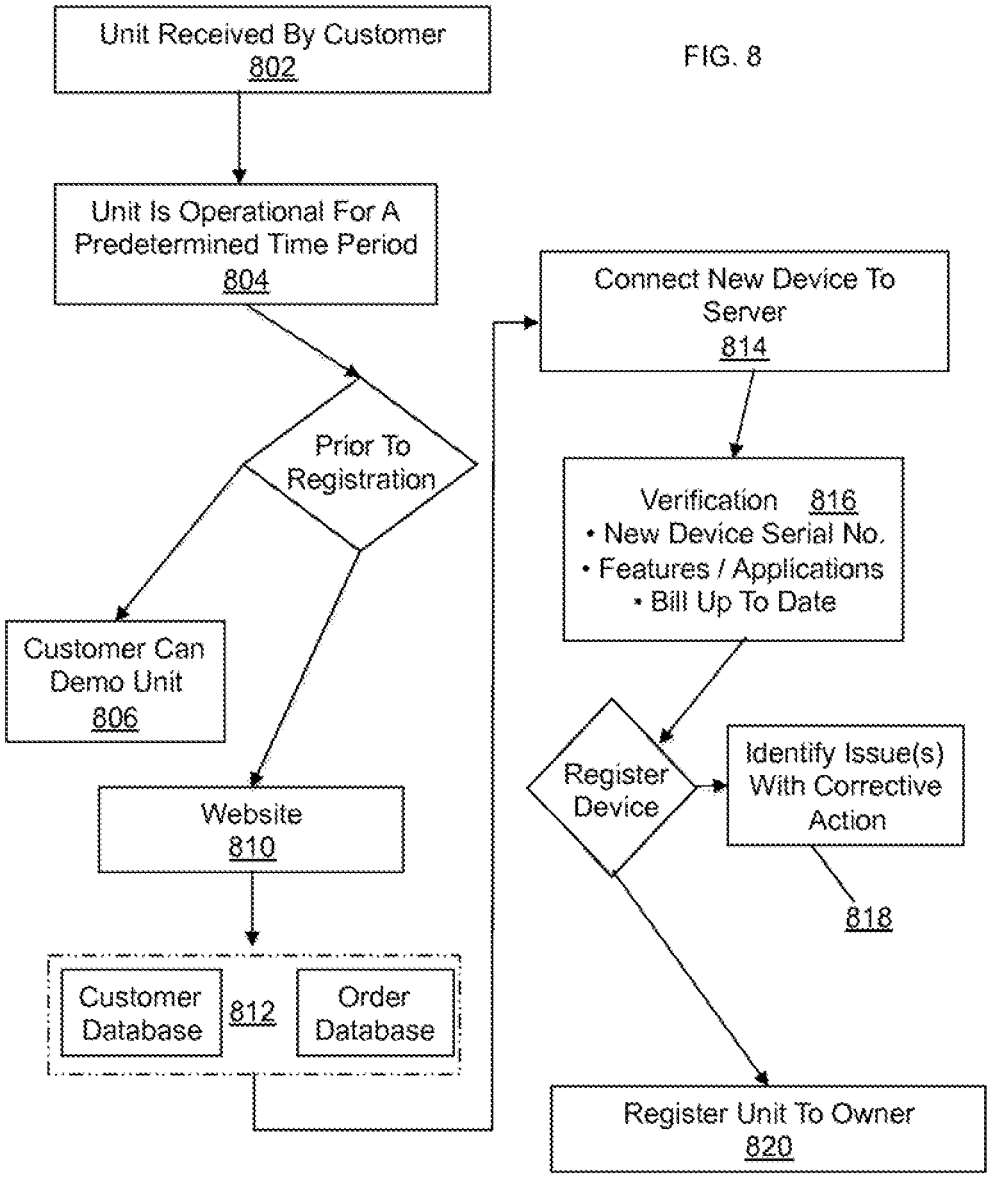
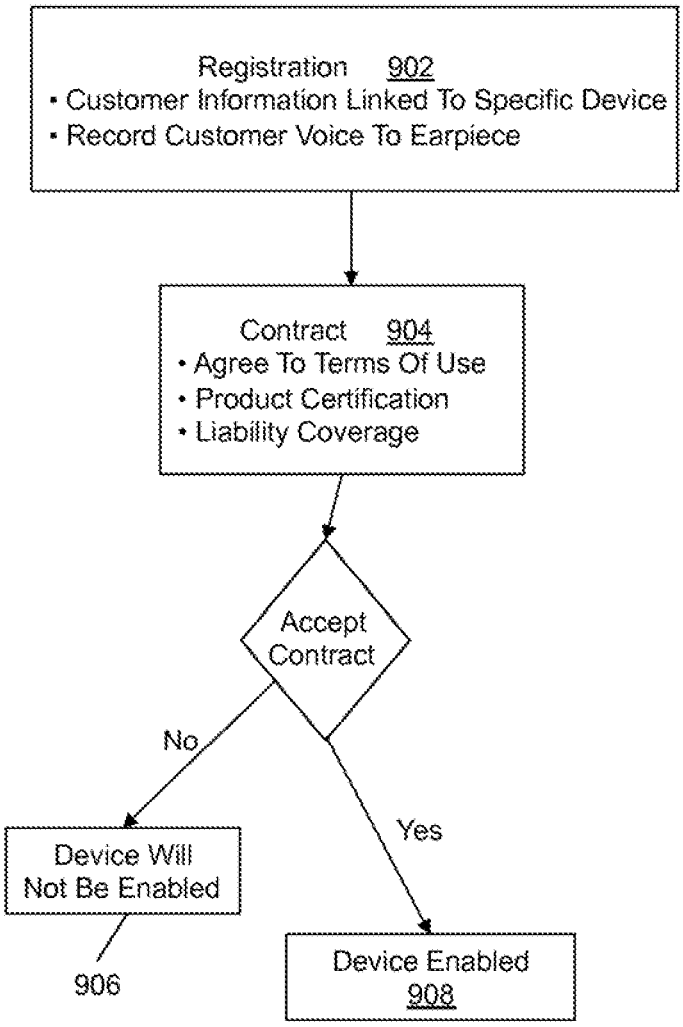

Unit Received By Customer
802
Unit Is Operational For A Predetermined Time Period
804
Prior To Registration
Customer Can Demo Unit
806
Website
810
Customer Database
812
Order Database
Connect New Device To Server
814
Verification 816
• New Device Serial No.
• Features / Applications
• Bill Up To Date
Register Device
Identify Issue(s) With Corrective Action
818
Register Unit To Owner
820

FIG. 8

FIG. 9


Registration 902
• Customer Information Linked To Specific Device
• Record Customer Voice To Earpiece
Contract 904
• Agree To Terms Of Use
• Product Certification
• Liability Coverage
Accept Contract
No
Yes
Device Will Not Be Enabled
906
Device Enabled 908

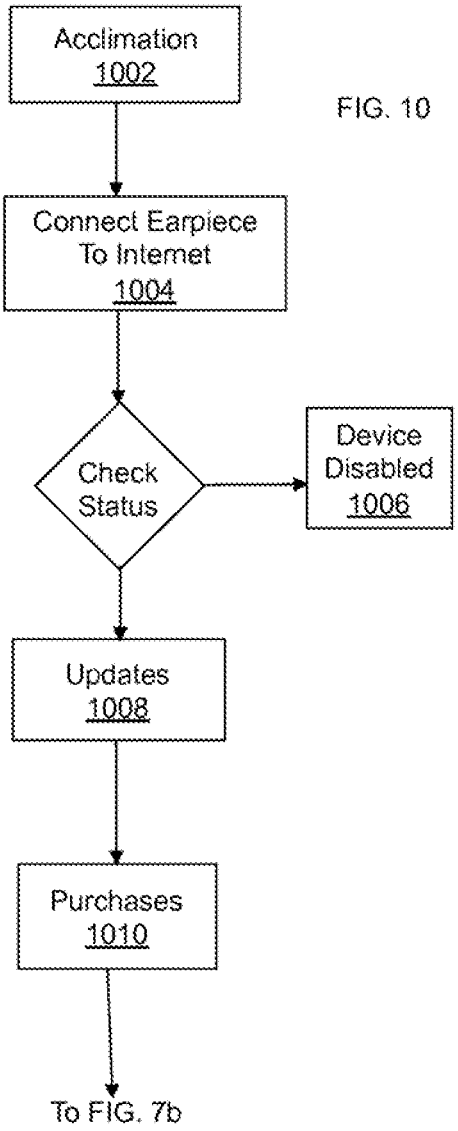

Acclimation
1002
Connect Earpiece
To Internet
1004
Check
Status
Device
Disabled
1006
Updates
1008
Purchases
1010
To FIG. 7b

FIG. 10

FIG. 11

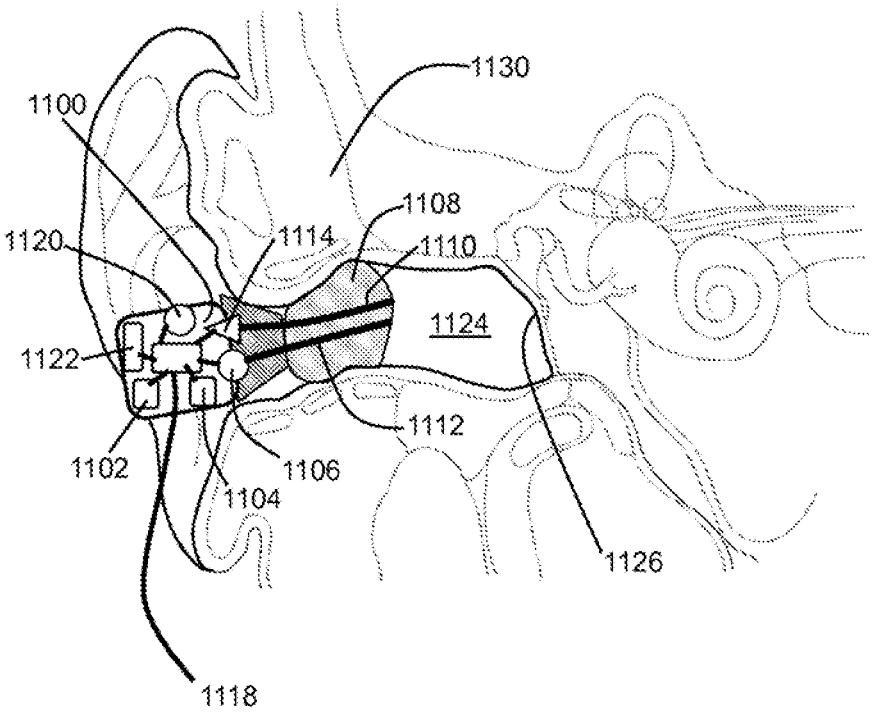

1130
1100
1108
1120
1114
1110
1124
1122
1112
1102
1106
1104
1126
1118

FIG. 12

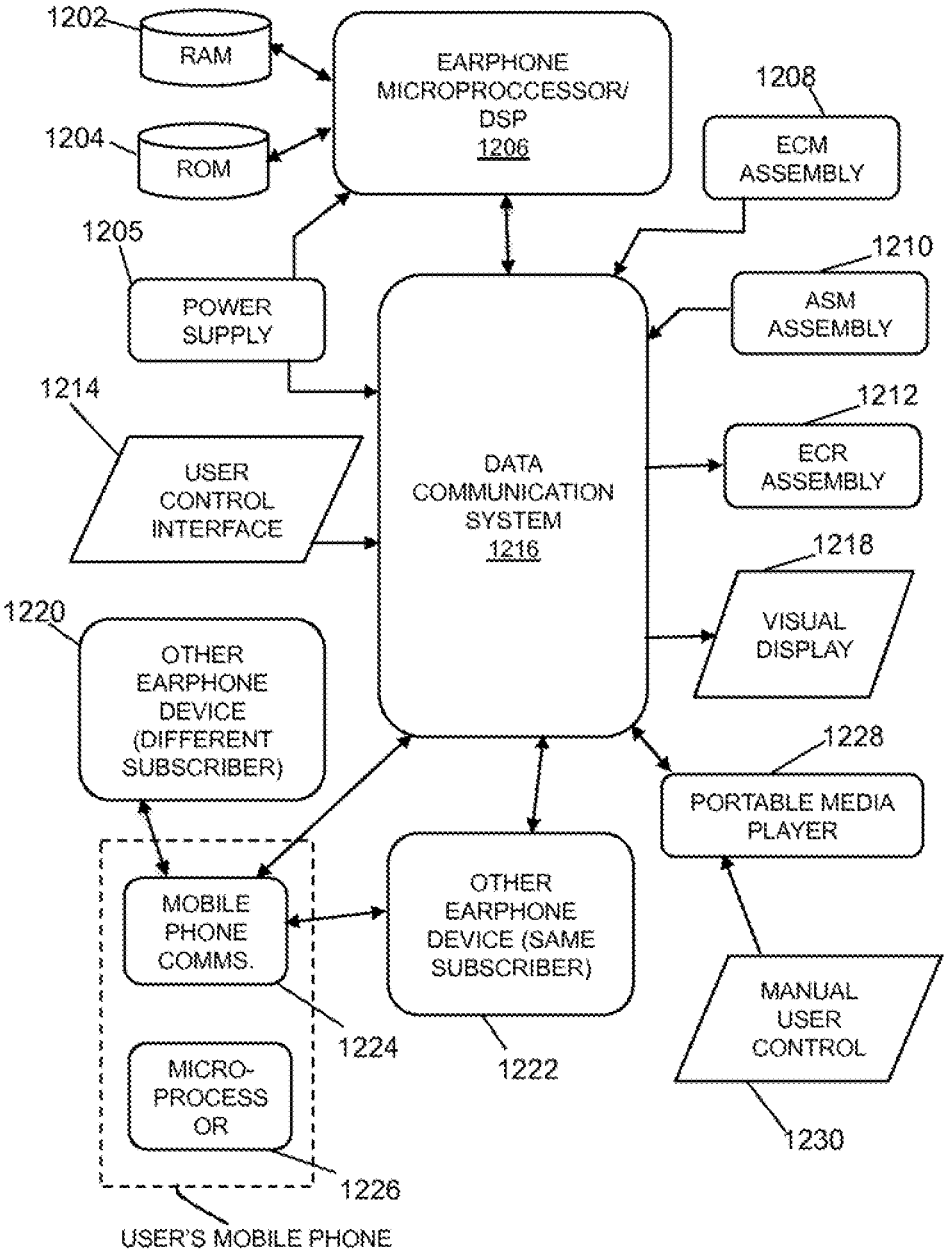

1202
RAM
1204
ROM
EARPHONE MICROPROCCESSOR/ DSP 1206
1208
ECM ASSEMBLY
1205
POWER SUPPLY
1210
ASM ASSEMBLY
1214
USER CONTROL INTERFACE
DATA COMMUNICATION SYSTEM 1216
1212
ECR ASSEMBLY
1218
VISUAL DISPLAY
1220
OTHER EARPHONE DEVICE (DIFFERENT SUBSCRIBER)
1228
PORTABLE MEDIA PLAYER
MOBILE PHONE COMMS.
OTHER EARPHONE DEVICE (SAME SUBSCRIBER)
MANUAL USER CONTROL
1224
1222
MICRO-PROCESS OR
1230
1226
USER'S MOBILE PHONE

FIG. 13

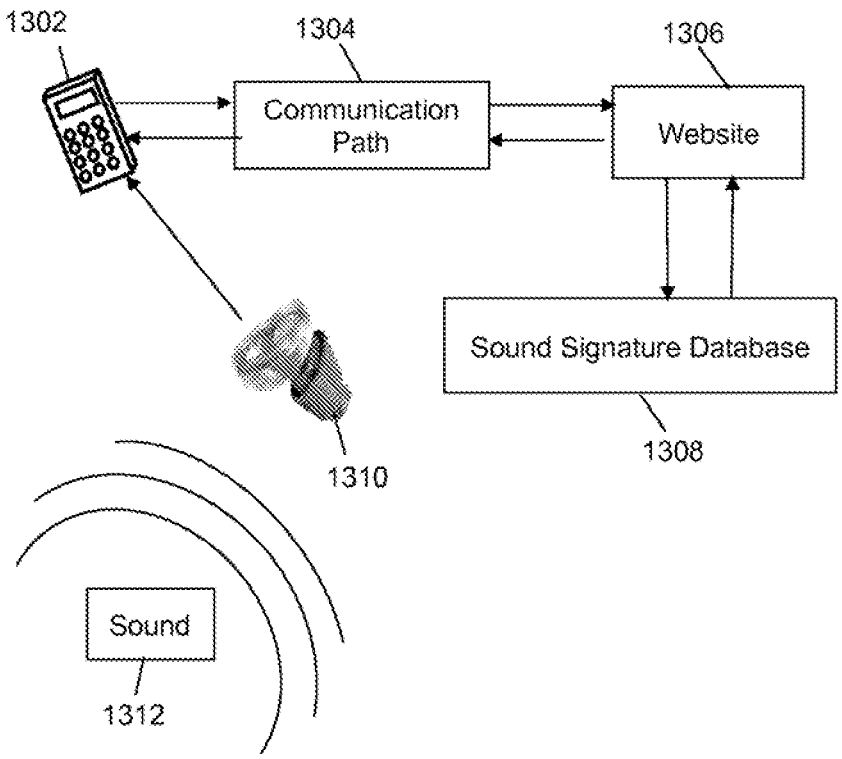
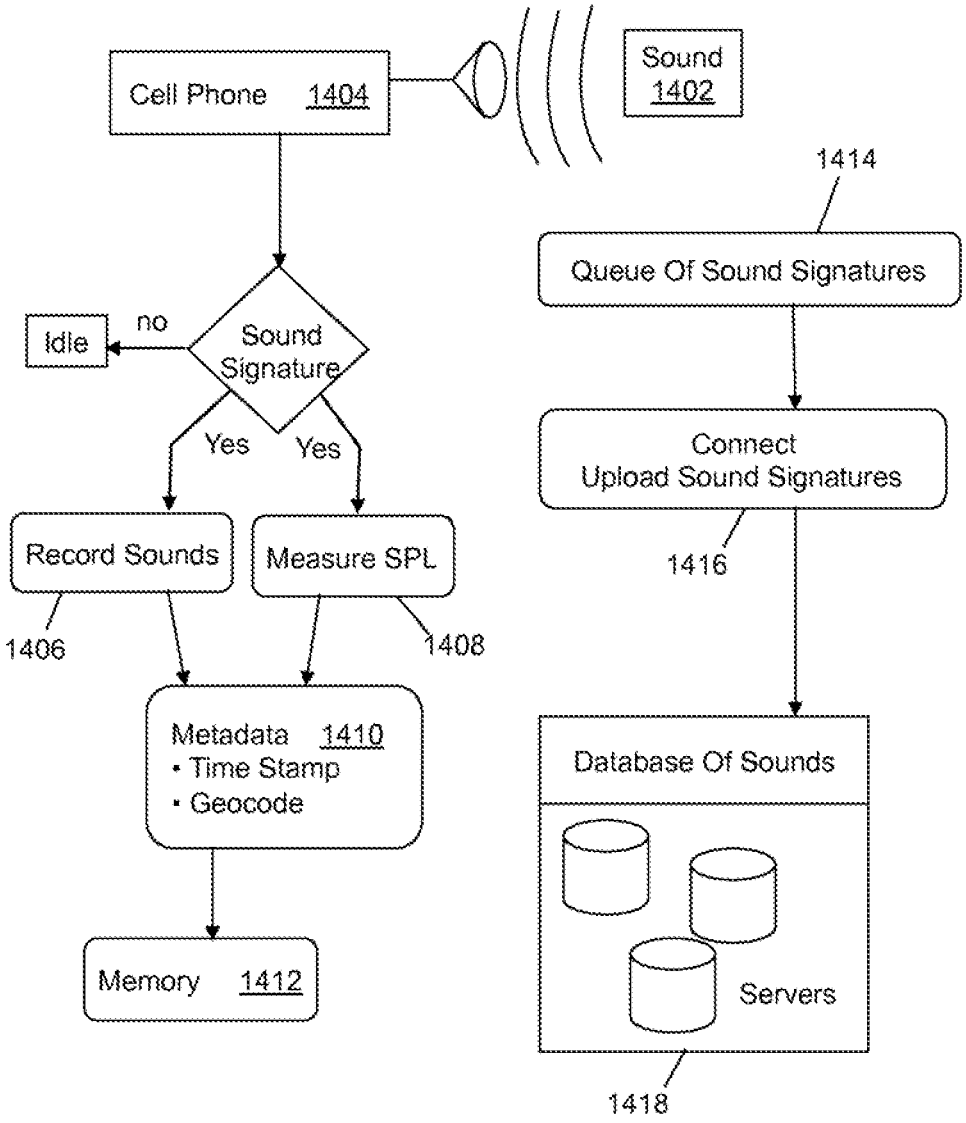

1302
1304
1306
Communication Path
Website
Sound Signature Database
1308
1310
Sound
1312

FIG. 14


Cell Phone 1404
Sound 1402
1414
Queue Of Sound Signatures
Idle
no
Sound Signature
Yes
Yes
Connect Upload Sound Signatures
Record Sounds
Measure SPL
1416
1406
1408
Metadata 1410
• Time Stamp
• Geocode
Database Of Sounds
Servers
Memory 1412
1418

PERSONALIZED SOUND MANAGEMENT AND METHOD

CROSS-REFERENCE TO RELATED APPLICATIONS

This application is a continuation of and claims priority benefit to U.S. application Ser. No. 17/203,731, filed 16 Mar. 2021, which is a continuation of U.S. application Ser. No. 16/671,689, filed 1 Nov. 2019, which is a continuation of U.S. application Ser. No. 14/846,994, filed 7 Sep. 2015. U.S. application Ser. No. 14/846,994 is a continuation of U.S. application Ser. No. 12/560,097 filed on 15 Sep. 2009 and claims the benefit of U.S. provisional patent application No. 61/098,914 filed 22 Sep. 2008. The disclosure of the aforementioned applications are all incorporated herein by reference in their entireties.

FIELD OF THE INVENTION

The invention relates in general to methods of managing sound, and particularly though not exclusively, is related to personalized sound management.

BACKGROUND OF THE INVENTION

The world of two hundred years ago is substantially different than the present day earth. Similarly, the acoustic environment that surrounds us is also changing. For example, the sounds of a large city have changed as the mode of transportation transitioned from walking and horse and buggy, to cars, subways, and airplanes.

In general, humans are continuously inundated by a diversity of sounds. Many of the sounds are not critical to our lives but our brain processes these sounds and tries to distinguish between them. Background sound levels can also make it difficult to hear sounds that are important. Too much acoustic information can cause an auditory overload that can impact both the health and safety of an individual.

SUMMARY

The invention relates in general to methods and system for implementing a suite of personalized sound applications for modifying a user's acoustic environment and more particularly, though not exclusively, to facilitating the adoption of the technology, ensuring the technology functions properly, protecting both the manufacturers and consumer, and providing user selection and control over the management of sound.

At least one exemplary embodiment is directed to a method of personalized sound management comprising the steps of: selecting at least one of a plurality of personalized sound management applications through a client system where the user selects the at least one of the plurality of personalized sound management applications from a website; accepting a subscription contract for using the at least one of the personalized sound management applications; and loading the selected at least one of the plurality of applications from a server system to a device where the device has at least one microphone, at least one speaker, and a processor configured to identify sonic signatures where each sonic signature is identified using a Gaussian mixture model.

At least one exemplary embodiment is directed to a method of implementing personalized sound management comprising the steps of: recording sound with a microphone of a communication device; analyzing the sound for acoustic information relevant for personalized sound management applications; storing a sonic signature in a memory of the communication device; calculating a sound pressure level of the sonic signature; and attaching and storing metadata related to the sonic signature and sound pressure level including a time stamp and geocode.

BRIEF DESCRIPTION OF THE DRAWINGS

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

FIG. **1** illustrates a block diagram of the interaction of personalized sound management in accordance with at least one exemplary embodiment;

FIG. **2** illustrates a block diagram of a partial list of applications for personalized sound management in accordance with at least one exemplary embodiment;

FIG. **3** illustrates a block diagram of the interaction of a personalized management enabled device with other devices in accordance with at least one exemplary embodiment;

FIG. **4** illustrates a flow chart of a process of providing personalized sound management in accordance with at least one exemplary embodiment;

FIG. **5** illustrates a flow chart of an application testing in accordance with at least one exemplary embodiment;

FIG. **6** illustrates a flow chart of the testing of a personal sound management device in accordance with at least one exemplary embodiment;

FIGS. **7***a* and **7***b* are diagrams illustrating a consumer purchase process in accordance with at least one exemplary embodiment;

FIG. **8** illustrates a flow chart of registering a new device that includes personalized sound management applications in accordance with at least one exemplary embodiment;

FIG. **9** illustrates a flow chart of enabling the new device in accordance with at least one exemplary embodiment;

FIG. **10** illustrates a flow chart of updating a unit or device in accordance with at least one exemplary embodiment;

FIG. **11** illustrates a diagram of a device for implementing personalized sound management in accordance with at least one exemplary embodiment;

FIG. **12** illustrates a block diagram of a device for implementing personalized sound management in accordance with at least one exemplary embodiment;

FIG. **13** illustrates a diagram of a communication device or earpiece configured to provide sonic signatures to a sonic signature database in accordance with at least one exemplary embodiment; and

FIG. **14** illustrates a block diagram of a cell phone capturing a sonic signature and providing the sonic signature to a database of sounds in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS OF THE PRESENT INVENTION

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

Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo

experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

Additionally, the sizes of structures used in exemplary embodiments are not limited by any discussion herein (e.g., the sizes of structures can be macro (centimeter, meter, millimeter, micro (micro meter), nanometer size and smaller).

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

In all of the examples illustrated and discussed herein, any specific values, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

FIG. **1** is a diagram of personal sound management in accordance with at least one exemplary embodiment. In our global community **100**, each person is unique. The differences between individuals can be genetic, cultural, environmental, personal, or physical to name just a few. The combination of these traits is what makes us unique. One of the main senses a human relies on is our auditory sense. Hearing impacts every aspect of our life from communication to safety. How each individual perceives and uses sound is also unique.

From an acoustic perspective, sound pressure levels have been rising steadily. The diversity of sounds is also increasing. The human brain continuously processes the acoustic information provided by the ears. Both the sound pressure levels and the sound diversity puts increasing strain on a person to determine what they need to hear versus what they are hearing. Ultimately, this "acoustic overload" can manifest itself in physical ailments and health risks such as stress, sleeplessness, and depression.

A person **102** has an acoustic space **104** from which a majority of sounds they receive emanate. Some of the sounds are useful to person **102** and some sounds may have no use. Acoustic space **104** can be large or small. Acoustic space **104** will change with time and location. For example, acoustic space **104** can be a room, a stadium, a forest, an automobile, a plane, or the ear canal of person **102**. Personalized sound management (PSM™) **106** is the ability to modify what is received from acoustic space **104** thereby tailoring or personalizing the received acoustic information to meet the needs of person **102**.

Devices **108** are a source of acoustic information within acoustic space **104**. In general, device(s) **108** have a speaker, a microphone, or one or more of both. In general, application hardware and software **110** is incorporated in, although can be external also, device(s) **108** to allow personalization of acoustic space **104**. Person **102** selects the applications and controls device(s) **108** to modify acoustic space **104** to meet their personal needs and wants. The benefit and utility of personalizing and managing sound received by person **102** will be disclosed in more detail.

FIG. **2** is a diagram illustrating a partial list of applications for personalized sound management in accordance with at least one exemplary embodiment. A user **202** has an acoustic space **204** that includes both real sounds and potential sounds. In many cases, acoustic space **204** can be inundated with a variety of sounds not under control of user **202**. This lack of control over acoustic information and sound within acoustic space **204** can make for a reduction in quality of life, loss of efficiency, or more seriously affecting user's **202** health and safety.

In at least one exemplary embodiment, a user **202** selects **201** at least one personalized sound management application from a personalized sound management applications list **206**. Although only a partial list, applications such as Safe Space™ **216**, EarGuard™ **218**, Quiet Call™ **220**, Ear Sticky™ **230**, Hearable™ **222**, Always On-Recording™ **224**, Earbliss™ **226**, Hear & Tell™ **232**, and Ear Mai™ **228** will be used to illustrate how acoustic space **204** is personalized and managed for the benefit of user **202**. A more detailed explanation of personalized sound management applications **206** will be described.

The selected applications of user **202** are loaded into ROM **212**. An operating system **208** is configured operably with device **210** and ROM **212**. Operating system **208**, in conjunction with personalized sound management applications **206**, provides an interface to the user for personalized sound management **214** to meet user needs to manage user acoustic space **202** and for controlling device **210**.

ROM **212** can be read only memory such that the selected applications from personalized sound management list **206** cannot be tampered with or rewritten. However in at least one exemplary embodiment ROM **212** can also be memory that can be read and written to so that a user can change settings. Alternately, ROM **212** can be other types of memory such as fe-ram, phase change memory, magneto-resistive memory, hard drive, sram, dram, eprom, eeprom, and other non-volatile or read only memories where the selected applications are secure from any type of downloading, tampering, or modification through physical or software protection. Monitoring of ROM **212** and the selected applications can also be added where notification is sent or device **210** is disabled when an improper action is detected. ROM **212** ensures that an application will operate as disclosed for user **202**.

Components of device **210** can be built in a single unit or operatively coupled units. For example, multiple devices of **210** can be wired, optically connected, wirelessly connected or a combination thereof. Operating system **208** can be run remotely to operate device **210** or reside within device **210**. Similarly, ROM **212** can be located in device **210** or remote to device **210**. In at least one exemplary embodiment, operating system **208** resides in ROM **212**. Device **210** typically has at least one microphone or at least one speaker, or both. The at least one microphone provides acoustic information for use in conjunction with personalized sound management applications. The at least one speaker provides acoustic information to user **202** and acoustic space **204**. Device **210** has a microprocessor (not shown) for running applications **206**. In at least one exemplary embodiment the microprocessor is dedicated to running personalized sound management applications **206**.

In general, personalized sound management applications are customizable point solutions that allow user **202** to handle a variety of tasks associated with managing acoustic space **204**. The selection of personalized sound management applications will depend on the person, the device, and the acoustic space being managed. Non-limiting examples of devices that can use personalized sound management applications with the appropriate hardware and operating system, are earpieces, media devices, and vehicles.

Described hereinbelow are brief overviews of the user selected personalized sound management applications:

Safe Space™ **216** is an "intelligent hearing" application that detects and outputs a response based on the identification of a recognized sound. The concept of recognizing sounds is described as Sonic Signature Detection. Safe

Space™ **216** also includes a user defined hierarchy and can provide responses based on the hierarchy.

A brief example is provided of how Safe Space™ **216** manages the acoustic space corresponding to the interior of an automobile. As is well known, automobiles are designed to have a quiet interior space. Sound insulation is deliberately placed around the car interior to attenuate sounds coming from the car exterior. Furthermore, automobile drivers often listen to high volume music while driving. This combination makes it difficult to hear sounds such as emergency vehicles. People often get seriously injured because they hear the emergency vehicle when it is in close proximity to the vehicle and panic causing an accident. All descriptions below describing personalized sound management applications assume that the device using the application has incorporated within it the hardware and software required to run the application and perform the application function.

In this example, Safe Space™ **216** detects emergency vehicle horns when they are a substantial distance away. In at least one exemplary embodiment, a microphone is exterior to the car and can pick up sounds in the ambient or car exterior. Sonic signatures related to emergency vehicles such as fire trucks, ambulances, and police cars are stored in the system. Safe Space™ **216** analyses sounds from the microphone. A response is provided if one of the stored sonic signatures is detected. In at least one exemplary embodiment, upon detecting a fire truck siren, Safe Space™ **216** can initiate playing the identified signal through the car stereo system for the car driver to hear and respond to. In at least one exemplary embodiment, Safe Space™ **216** can calculate the direction, distance, and street (through GPS) of the approaching emergency vehicle. The information is then provided visually or vocally to the driver. For example, the car stereo automatically turns down the music and states through the speaker system that an ambulance is coming east bound on 3.sup.rd Street or an ambulance is approaching from the right.

EarGuard™ **218** is a personalized sound management application that improves listening quality and safety. Hearing disorders are increasing at a very high rate. Many people live in an environment such as a noisy urban environment or an industrial manufacturing area where sound pressure levels are consistently high. Moreover, people themselves are subjecting themselves to loud and sustained sounds. Examples of extreme or sustained sound exposure are portable media players using earbuds, sport shooting, and rock concerts. The trend is a growing population of people that have or will have hearing problems.

The ear is a very sensitive instrument having a large dynamic range that allows us to hear sounds ranging from a whisper to a shout. Subjecting the ear to man-made cacophony is now known to metabolically exhaust the highly specialized sensory cells in the inner ear, causing them to die and be replaced by scar tissue in the hearing organ. Ear Guard™ **218** is an application that monitors the sound exposure of the user and protects the ear from damage.

Briefly, Earguard™ **218** improves listening quality and safety by employing a personal hearing zone that measures sound levels within the ear and in one exemplary embodiment an intelligent level control to protect against over amplification. Earguard™ **218** includes a sound pressure level (SPL) dose management system. The SPL dose management system takes into account both transient and long term sound. In at least one exemplary embodiment, when sound at the user's eardrum is above a certain threshold known to begin the process of metabolically exhausting the ear, the metric known as SPL Dose increases. The SPL dose decreases when sound is below that threshold according to a recovery function that mirrors the recovery of the ear from excessive sound exposure. In at least one exemplary embodiment, Earguard™ **218** will indicate to the user damage to the ear is likely if the sound exposure is continued. In this scenario, the user has the control to take the appropriate action to protect his/her ears.

In at least one exemplary embodiment, an intelligent level adjustment system is a personalized sound management application that automatically estimates the preferred listening level of an audio content signal (e.g. speech or music audio from a device) depending on an analysis of the level, other acoustic features of the ambient environment, and an analysis of the audio content signal. Thus, a human-machine relationship is nurtured with this bi-directional control flow from human to the intelligent level adjustment system and from intelligent level adjustment system to the human user.

A substantial benefit of an application such as Earguard™ **218** is the protection of the ear from damage. Furthermore, this safeguard if propagated will prevent hearing loss thereby reducing the financial burden on the economy in years to come for the individual, business, and government for hearing loss remedies. Thus, using Earguard™ **218** could not only stem the growth in hearing related problems but greatly reduce it over the years to come.

Quiet Call™ **220** is an application that allows user **202** having heard a voice message to respond to the remote caller through a non-verbal means. An example of a non-verbal response is a key-pad entered text message. The entered text message is converted to a speech audio message and is sent to the remote caller. The caller then receives the speech audio message.

An example of the utility of Quiet Call™ **220** is illustrated when user **202** is in an important business meeting but is required to provide input to another remote meeting. User **202** receives the voice message sent from someone in the remote meeting. User **202** responds by entering the response through his/her phone keypad; the text is converted to voice and sent to the person at the remote meeting who sent the original message. User **202** does not have to interrupt the meeting to listen to the voice message nor to reply. Thus, the meeting attended by user **202** can move forward with little or no loss of momentum that would occur if the meeting was disrupted by the call.

Hearable™ **222** is a speech enhancement application that improves voice communication. For example, Hearable™ **222** can be used with an earpiece device having at least two microphones, an ambient sound microphone for receiving sounds in the ambient environment and an ear canal microphone for receiving sounds in the ear canal. A common use for the earpiece would be with a communication device such as a cell phone or other phone system. The earpiece using the Hearable™ **222** application would normally use the ambient sound microphone for receiving and transmitting the user's spoken voice. In this mode, the user's voice will be natural sounding and easily recognizable on the receiving end.

In a noisy environment it can be difficult to carry on a telephone conversation. The ambient sound microphone will pick up the voice and the noise in the ambient environment. The earpiece will switch to the ear canal microphone when the ambient sound microphone detects a high background noise level. The user's voice is readily picked up by the ear canal microphone but the noise in the ambient environment is substantially reduced. Switching to the ear canal microphone allows the receiving party to clearly hear the user's

voice. The problem with using the ear canal microphone is that user's voice received in the ear canal sounds different because of frequency roll off in the upper spectrum of the voice range. Although the user can be heard clearly, the user's voice may not sound right to the receiving end.

Hearable™ **222** is a personalized sound management application that improves the sound quality of the user's voice. Hearable™ **222** uses a combination of the sound received by the ambient sound microphone and the ear canal microphone to create a more natural sounding voice. The combination of the two signals is a function of the background noise level. Explaining further the signal from the ambient sound microphone is used less as the background noise level increases. Hearable™ **222** allows a user to have a conversation in a noisy environment while providing a high quality voice signal that is intelligible and recognizable at the receiving end.

Always On-Recording™ **224** is a personalized sound application that acts as its name implies. A device using the Always On-Recording™ **224** application is recording the most current audio information for user recall. The recording is stored in a buffer that will allow the user to immediately access the audio information. The buffer will have a finite amount of storage. The recording time will be a function of the buffer or memory available for the Always On-Recording™ **224** application.

Always On-Recording™ **224** provides utility for short durations of recording. For example, in an application where a user is receiving information such as a phone number or driving directions. The user knows that the device employing Always On-Recording™ **224** has the stored acoustic information and can immediately listen-again to the buffer contents thereby repeating the phone number or driving directions. Similarly, if the user was discussing a contract term and wanted to know exactly what the person said, the user could immediately re-listen to make sure that what they thought they heard is exactly what they heard.

Earbliss™ **226** is a personalized sound management application to provide acoustic isolation from someone who snores while allowing other sounds to be heard. A large percentage of the population suffers though sleepless nights because of the sounds generated by people with sleep apnea and more generally due to loud snoring. Moreover, sleep deprivation can have serious consequences related to health and disposition.

Earbliss™ **226** is a sleep zone technology that utilizes sonic signature detection to insulate sleep partners against intrusion from snoring while still providing awareness of priority sounds. In this example, a sonic signature is acoustic information related to the sounds of a person snoring. In at least one exemplary embodiment, Earbliss™ **226** is an application that is used in conjunction earpieces.

Sealing the ear canal with ear plugs will attenuate the snoring but also blocks out all sounds. Hearing is one of the most vital senses that we have. Under normal conditions we cannot turn off our hearing which allows us to wake in a critical situation. Ear plugs will block out sounds of consequence. For example, the user may not hear their baby crying or a child leaving their room.

Earbliss™ **226** enables a user to attenuate snoring while hearing other sounds around them. The earpieces fit in the ear canal of the user to seal or partially seal the ear canal. The earpieces have an ambient sound microphone for receiving acoustic information from the ambient environment and an ear canal receiver for providing sound to the user's ear canal. As mentioned previously, the earpieces will have a sonic signature related to snorer's snoring stored in memory in the device.

The ambient sound microphone picks up all sounds in the ambient environment including the snorer's snoring. The earpiece processes all the acoustic information coming from the ambient sound microphone and looks for signals similar to stored sonic signatures. Pattern recognition approaches are applied based on the known sonic signatures to detect the snoring sounds from their corresponding sonic signatures. More specifically, sonic signatures can then be compared to learned models to identify a corresponding snoring sound. Once identified the snoring sound is suppressed and not output to the ear canal by the ear canal receiver. Thus, the wearer of the earpieces does not hear the snoring.

Conversely, sounds in the ambient environment that are not recognized through the processing of acoustic information by the earpieces can be passed transparently to the ear canal receiver for reproduction within the ear canal. In this mode, the sound produced in the ear canal sufficiently matches the ambient sound outside the ear canal, thereby providing a "transparency" effect other than the suppressed sonic signature (snoring). The earpieces can also enhance sound. For example, the earpieces having sonic signatures related to a fire truck siren or a baby crying can detect either signal and then amplify the signal (fire truck siren, baby crying) so as to make the wearer of the earpieces aware of the signal's detection. Thus, Earbliss™ **226** modifies the acoustic environment of the user to eliminate what does not need to be heard while allowing the user to be aware of other sounds in a normal context.

Ear Mail™ **228** is a personalized sound management application for converting text to voice. In particular, Ear Mail™ **228** provides great utility to users of email and text messaging although not limited to these examples. Email and text messaging is becoming a very popular form of communication among a large portion of the population. It is not always convenient or in some cases not prudent to review written text depending on the situation.

Ear Mail™ **228** converts the text of a message or email to a speech audio message using a text to speech algorithm. The converted speech audio message is replayed through a loudspeaker coupled to a device using the Ear Mail™ **228** application. For example, Ear Mail™ **228** used in a smart phone coupled to the stereo system of a car through a blue tooth connection could playback the text or emails through the car speakers. The user could hear their messages while driving safely down the road.

Ear Sticky™ **230** is a personalized sound management application for recording information that can be saved and used for future use. In at least one exemplary embodiment, Ear Sticky™ **230** is a mobile communication application that can be used in a device such as a cell phone or an earpiece that is operably coupled to other devices. Ear Sticky™ **230** can record communication through a device or sounds in the ambient environment. Sounds in the ambient environment are recorded by an ambient sound microphone.

In a first example of an Ear Sticky™ **230** application, a conversation between a husband and wife occurs and a list of items to pick up at several stores is disclosed. The user of Ear Sticky™ **230** does not need to write down or remember this list. The conversation is being recorded and stored in a buffer. The user activates Ear Sticky™ **230** to store the recorded conversation to be reviewed at a later time. Thus, the user of Ear Sticky™ **230** could recall and listen to the list on the way home to ensure that the right items are picked up at the appropriate store.

In a second example of an Ear Sticky™ **230** application, a device records with an ambient sound microphone. For example, the user of Ear Sticky™ **230** comes up with a great concept to solve a problem he or she has been working on for some time. The user can enable Ear Sticky™ **230** and uses the ambient sound microphone to record his or her voice to convey the idea, concept, or thoughts and store it for review at a later time. In general, Ear Sticky™ **230** provides utility and convenience in storing and recalling sounds in one's acoustic space.

Hear & Tell™ **232** is a personalized sound management application for recording a sound, training a Gaussian mixture model to learn features of the sound and then storing the Gaussian mixture model in memory of the device. A user of Hear & Tell™ **232** can record the sound or provide a sound, called herein as a sonic signature. The device is operably coupled to at least one microphone to compare sounds received from the microphone against the stored sonic signature. The device can perform an operation that modifies the user's acoustic space **204** once a sound is identified as being similar to a sonic signature. Examples of several operations the device can perform are passing the detected signal through to the user, boosting the sound such that the user is made aware of the detected sound, rejecting the sound so the user does not hear the detected sound, attenuating the detected sound, and replacing the detected sound with an alternate sound to name.

The Hear & Tell™ **232** application will store a Gaussian mixture model (GMM) for every sonic signature that it has been trained to recognize. Each GMM is completely specified by a mixture of mean vectors, a mixture of covariance matrices, and a mixture of weights.

An example of a warning sound (e.g. siren of emergency vehicle) will be used to further illustrate the Hear & Tell™ **232** learning application. Each GMM provides a model for the distribution of the feature statistics for each warning sound in a multi-dimensional space. Upon presentation of a new feature vector, the likelihood of the presence of each warning sound can be calculated. In at least one exemplary embodiment, each warning sound's GMM is evaluated relative to its anti-model, and a score related to the likelihood of that warning sound is computed in order to determine if a sound is detected. A threshold can be applied directly to this score to decide whether the warning sound is present or absent. Similarly, a sequence of scores can be relayed and used in a more complex rule set to determine absence or presence of the sound. Thus, Hear & Tell™ **232** allows a user to store, model, and train a device for sonic signatures under user control and selection thereby allowing a user to modify their acoustic space through the detection of a sound and response by the device.

In general, personalized sound management applications are customizable point solutions that allow user **202** to handle a variety of tasks associated with managing acoustic space **204**. The selection of personalized sound management applications will depend on the person, the device, and the acoustic space being managed. Examples of devices that can use personalized sound management applications with the appropriate hardware and operating system, but not limited to these examples, are earpieces, media devices, and vehicles.

As described hereinabove, a suite of personalized sound management applications **206** are provided. The user **202** can select the desired applications and have control over their acoustic space **204**. The selected applications are stored in ROM **212**. Under user control, user **202** selects the parameters of personalized sound management applications **206** using operating system **208** for implementing applications **206** in device **210** for managing acoustic space **204** based on the individual's needs and wants.

FIG. **3** is a diagram illustrating a module **300** for implementing personalized sound management in accordance with at least one exemplary embodiment. Module **300** comprises an H-Chip™ **302**, a ROM **304**, an operating system **306**, and user selected personalized sound management applications **308**. In at least one exemplary embodiment, module **300** comprises the H-Chip™ **302** with ROM **304** built into H-Chip™ **302**. Both operating system **306** and user selected personalized sound management applications **308** are stored on ROM **304**. Alternately, H-Chip™ **302** and ROM **304** can be separate chips allowing for a larger block of memory.

H-Chip™ **302** is a microprocessor, DSP, or logic circuit that implements personalized sound management applications **308**. H-Chip™ **302** is optimized for low power dissipation while running user selected personalized sound management applications **308**. In at least one exemplary embodiment, H-Chip™ **302** can be a dedicated engine for running applications **308**.

Module **300** comprises an application engine (firmware) and a dedicated processor that simplifies the integration of user selected and user controlled personalized sound management applications **308** into a device. In an exemplary embodiment, module **300** is integrated into each device in devices **314**. Devices **314** typically, although not always have at least one microphone and at least one speaker. User **310** in conjunction with module **300** can personalize how each device manages sound to the user.

An additional aspect of module **300** is that a third party manufacturer merely builds the acoustic portion of their device around module **300**. In an exemplary embodiment, a manufacturer does not and cannot modify operating system **306** and user selected personalized sound management applications **308**. Thus, the manufacturer saves time, effort, and money through the seamless integration module **300** into a single hardware solution for maximum transportability into 3.sup.rd party form factors.

FIG. **4** is a diagram illustrating a process of providing personalized sound management in accordance with at least one exemplary embodiment. In general, the technology related to providing personalized sound management comprises hardware components and software. In at least one exemplary embodiment, the hardware **402** comprises transducers **404** and H-Chip™ **406**.

Transducers **404** are speaker and microphones for respectively providing and receiving sound in an acoustic space. Depending on the application, the size and form factor of transducers **404** may be a critical design parameter. Transducers **404** can be a speaker for providing high fidelity sound to enhance a user's experience. Similarly, transducers **404** can be a microphone for receiving acoustic information and in some cases can pick up sounds inaudible to the user.

H-Chip™ **406** can be a microprocessor, digital signal processor, logic circuit, or applications specific integrated circuit for use implementing the software applications listed hereinabove and other programs related to personalized sound management. H-Chip™ **406** includes operating system **408** specific for managing the device as it relates to personalized sound management and allows user control to adjust the parameters of each application for their specific needs.

Hardware **402** can be integrated into existing devices or next generation devices for adding utility, providing safety, and allowing personalization of a user's acoustic space.

Providing hardware **402** enables device manufacturers **410** to rapidly integrate the technology into their products. This will quicken the adoption cycle of the technology for the benefit of the general public in health and safety, and for the individual through personalization. Hardware **402** can be a circuit board to which transducers **404** and H-Chip™ **406** are operatively attached.

Device manufacturers **410** can provide their own hardware **414** or use transducers **404** and H-Chip™ **406**. The operating system and application software is incorporated **412** and stored in read only memory (ROM). In at least one exemplary embodiment, operating system **408** and the application software can be stored in ROM. Using read only memory for storage prevents device manufacturers **410** or consumers from tampering with the software code thereby maintaining the integrity of the system and how it performs for the user. Note herein when referring to ROM storage, at least one exemplary embodiment can include RAM or other read/write storage methods.

People in the consumer product industry have stated that any new product and even old products in the hands of consumers may produce liability issues that companies associated with the product have to deal with. This often delays an introduction of a product or adds substantial cost to a product launch. In at least one exemplary embodiment, device manufacturers **410** provide their product for certification **416**. In at least one exemplary embodiment, certification **416** is performed by an independent company from device manufacturers **410**. Certification **416** is a process whereby a new product is exhaustively tested to ensure the device will perform to specification under a variety of conditions and tolerances. In particular, certification **416** tests the operating system and the device performance in implementing the applications related to personal sound management. Should the device not pass certification testing, it can be repaired or redesigned. The repaired or redesigned device can then undergo certification **416** to determine if it meets the test specifications.

In at least one exemplary embodiment, an insurance policy **418** may be provided covering the device manufacturers **410** using operating system **408**, hardware **402**, and the application software. This provides substantial benefits in that the device manufacturers **410** can deploy the technology with less risk. Moreover, each company providing devices will know that the personalized sound management system is similar or identical to others in the market. As mentioned hereinabove, the operating system and application software can be stored in ROM ensuring it cannot be rewritten. Additional measures can be taken to determine if the software is being used out of context and shutting down the operation of the device.

Once tested and certified, device manufacturers **410** can manufacture, market, and sell devices with personalized sound management **424** and be covered from a liability perspective under provisions in the insurance policy **418**. A consumer purchases **420** a certified device and may also be covered under the same insurance policy **418** or may have a second policy directed to the consumer. Consumer protected **422** under insurance policy **418** and having certification **416** will instill consumer confidence to the quality and reliability of the device. This will be discussed in greater detail hereinbelow. Thus, this process is very efficient in creating manufacturer adoption of the personalized sound management technology while protecting both device manufacturers **410** and the consumer.

FIG. **5** is a diagram illustrating application testing in accordance with at least one exemplary embodiment. Device manufacturers **502** develop, prototype, and manufacture products **506**. Products **506** include hardware and operating system **504** for running and performing Personalized Sound Management (PSM™) applications **508**.

In at least one exemplary embodiment, hardware and operating system **504** is capable of running all personalized sound management applications **508**. Products **506** may or may not use all of personalized sound management applications **508**. Similarly, an end user or customer of products **506** may or may not use all of personalized sound management applications **508**.

Hardware and operating system **504** of products **506** are loaded with all personalized sound management applications **508**. The hardware and operating system **504** are exercised in testing **510** to ensure that it can run personalized sound management applications **508** per specification. Product **506** may or may not be tested for all personalized sound management applications **508** depending on the capability of the device. Products **506** will be tested **510** for the implementation of all personalized sound management applications it is capable of running to meet user specifications for the product (i.e. product certification **512**). In at least one exemplary embodiment, should a product run only a subset of personalized sound management applications, the product can be designed and tested to lock out the running of applications not specified for the particular product (even though hardware and operating system **504** can run personalized sound management applications **508**).

A scale of economy for developing and manufacturing hardware and operating system **504** is achieved by having a number of different device manufacturers **502** standardize on the engine (hardware and operating system **504**) for implementing personalized sound management. Consistency of product, ability to expand scope of product, and lowering cost through volume manufacturing are all achieved by this methodology. In the example where liability insurance may be purchased by the product certification company (or other company) testing **510** all personalized sound management applications **508** in products **506** ensures that hardware and operating system **504** perform per specification independent of the product it is placed in. Furthermore, having little or no variation on the operation of hardware and operating system **504** and personalized sound management applications **508** minimizes risk. An added benefit is that a large statistical database of diversified products is generated that will allow improvements to managing personalized sound management devices to improved device performance as well as lower liability risk. The consumer benefits by having a highly exercised and stable product that improves their health, safety, and quality of life.

FIG. **6** is a diagram illustrating testing of a personalized sound management product **602** in accordance with at least one exemplary embodiment. In at least one exemplary embodiment, testing of all products incorporating personalized sound management applications are tested by a single entity. Personalized sound management product **602** can be developed and manufactured by a number of different companies. As previously mentioned, this allows the single entity, should it so desire to purchase a liability insurance policy that covers the manufacturers making personalized sound management product **602** and consumers using personalized sound management product **602**.

Personalized sound management product **602** is tested by certification laboratory **604**. In at least one exemplary embodiment, certification laboratory **604** is independent from the different manufacturers that develop personalized sound management product **602** thereby providing unbiased

testing. Testing **614** of hardware **606**, software **608**, and manufacturing **610** are performed on personalized sound management product **602**. Testing **614** is related to operation and performance of personalized sound management applications. Hardware testing includes the processor, transducers, and other elements, which implement the personalized sound management of the device. Testing of software **608** includes the operating system and personalized sound management applications. Testing **614** includes reliability **616** and quality **618** testing. Testing **614** can also include human testing because in some embodiments, personalized sound management product may be worn in an orifice of the human body.

Upon passing testing **614** the personalized sound management product **602** is certified by certification laboratory **604**. A product certified **620** can include some form of warranty, indemnification, and liability insurance for the device. Problems or issues with personalized sound management product **602** are reported to the company should it fail for any reasons. Corrective action can be taken and the device retested to determine if the issue has been resolved.

In at least one exemplary embodiment, certification laboratory **604** or its parent company will brand the product by placing a trademark or company logo on the approved personalized sound management product **602**. The trademark or logo represents the incorporation of its personalized sound management technology in the product (hardware, software, and intellectual property) as well as the certification indicating the quality and reliability of the device.

FIG. **7***a* is a diagram illustrating a consumer purchase process for a personalized sound management product in accordance with at least one exemplary embodiment. A consumer reviews a number of different devices such as earpieces in a display area. The display area allows the consumer to look, feel, and touch the hardware. The consumer focuses on an earpiece **701** in the display area and wants to learn more about the device.

A sales person can demonstrate earpiece **701** allowing the consumer to test the functions and performance of the device. A demonstration room or test environment can be used to illustrate performance differentiation when compared to other earpieces. In at least one exemplary embodiment, earpiece **701** is an in-ear device that has a removable and disposable sealing section. This allows consumer testing using earpiece **701** while maintaining a sanitary condition by disposing of the sealing section after each test session.

The consumer indicates to the sales person that he or she would like to purchase earpiece **701**. The sales person accesses a computer that is coupled to server system for completing a purchase transaction. The consumer provides registration information that is entered into a database in a step **703**. The consumer or the sales person can enter the information into the database. The registration information **705** comprises information such as personal, financial, business, and preference. Personal information relates to data such as age, sex, home address, telephone number, email address, which identifies the consumer. Financial information relates the consumer's ability to pay such as credit history, credit card for billing, job information, and banking institutions. Business information relates to a business purchase of the component. Similar to above, business identification and business financial information would be entered with the purchase. Preferences relate to things such as how the consumer wants to be contacted (email, phone), do they want to be made aware of upgrades for their device, and other products they might have interest in. Other information or user experiences could also be collected during the registration process.

The unit selected by the consumer is then entered into the system in a step **707**. In at least one exemplary embodiment, the system identifies the unit and provides a listing of the hardware, software, and subsystems that work with the earpiece **701**. The consumer can look at the options available in a step **709** and select among these options. In at least one exemplary embodiment, the selection process can be facilitated by topic, occupation, or other characteristics by providing a list of typical user features applied to earpiece **701**. For example, the consumer is purchasing earpiece **701** for a multimedia player. Earpiece **701** provides a substantial improvement in noise attenuation, musical clarity, definition, and dynamic range. The consumer is interested in the safety aspect (and selects this topic on the kiosk) provided by earpiece **701** in alerting the user to an emergency event. For example, earpiece **701** can detect and make the user aware of sirens, emergency sounds, alarms, etc. that would be difficult to hear when listening to music. The consumer by looking at the "safety" topic can select appropriate applications software and sonic signatures to alert the user of a safety situation for their specific device. The consumer can continue to look through other topics and select hardware and software that supplements and implements the desired user experience.

As mentioned previously, the options can be provided in a step **709** in many different formats. Another format of providing options is by occupation. For example, a dentist may be provided a package to outfit the entire office and patients. The earpieces substantially reduce noise related to drilling which are both a psychological barrier for dentistry and a hearing loss mechanism for people working in a dentist office. The occupation package can provide wireless communication between devices allowing dentists, assistants, and patients to talk or listen to music even in a noisy environment. Replaceable in-ear sealing sections would allow the staff to maintain sanitary conditions with each new patient. After reviewing the consumer selects the appropriate hardware and software to personalize their sound space (on the kiosk or computer) in a step **711**.

In at least one exemplary embodiment, the selected application software provided to earpiece **701** is a subscription whereby the consumer pays a monthly fee to use the applications. Alternately, the selected application software can be a license or purchase associated with the earpiece **701** whereby a one time fee is paid. The consumer can discuss the terms of the agreement with the sales person or review the contract on the computer. If the consumer approves, the contract is signed or accepted and an initial payment is made for the hardware and software in a step **713**. The consumer can have the subscription payment automatically billed to a credit card or other automatic payment methodology to ensure uninterrupted service.

The purchased earpiece is then connected via a wired or wireless connection to download the selected software and enabled in a step **715**. A wired connection **717** such as a mini USB cable is shown in the illustration. Wired connection **717** connects to a server system having the appropriate software. In at least one exemplary embodiment, the selections from step **711** can be stored in memory and used to direct the downloading of software to the device, verify the software can be used in the device, and that the appropriate software has been loaded.

FIG. **7***b* is a diagram illustrating a consumer purchase process for a personalized sound management product in

accordance with at least one exemplary embodiment. A consumer **702** can purchase personalized sound management product hardware **704** and personalized sound management application software **706**. Software **706** works with hardware **704** to personalize and manage sound for consumer **702** for providing utility, safety, and health benefits.

Hardware **704** can range from a complete product such as a communication device, sound device, or earpiece to ancillary components for enhancing the look of the device or adding further hardware features. In the event that consumer **702** purchases hardware **704**, customer information **708** is required to continue with purchase. In general, customer name, address, personal information and financial information are typically required to purchase hardware **704**. Other information can also be collected through a questionnaire or some form of incentive. Should customer information **708** exist and consumer **702** is in good stead then generating customer information **708** is not required and the order process continues. Any issue that occurs in checking customer information **708** will be brought up for rectification to consumer **702**. The hardware **710** that consumer **702** selected is verified and place order **712** is performed.

Personalized sound management application software **706** is a suite of applications that consumer **702** selects for their specific needs. Once customer information **708** has been established, consumer **708** is provided subscription information **714**. Use of personalized sound management application software **706** is based on a periodic fee or subscription (for example, a monthly fee). Subscription information **714** informs consumer **702** as to the periodic subscription fees associated with the software **706**, terms of use, bundled packages of applications, and other pertinent information to complete the decision process. Should consumer **702** approve of the terms, a contract **716** is provided to the customer to accept or decline. A bill **718** is generated when an order is placed either through purchasing hardware **710**, accepting a subscription to personalized sound management application software **706**, or both. In at least one exemplary embodiment, the purchase process is web based allowing consumer **702** to purchase from a client system via a communication path such as the internet or communication network.

FIG. **8** is a diagram illustrating registering a new device that includes personalized sound management applications in accordance with at least one exemplary embodiment. A new device or unit is received by a customer **802**. The manufacturer has loaded the preordered personalized sound management applications chosen by customer **802** into the unit. Typically, customer **802** is going to want to try the unit shortly after receiving it (e.g., the unit is operational for a predetermined period of time **804**, for example 1 week). In at least one exemplary embodiment, customer **802** can try the unit for a predetermined time period **804**. In general, the time period **804** (for example, 15-30 minutes) is long enough for customer **802** to get a feel for the unit but is not a useful length of time. Furthermore, the unit will notify customer **802** through a transducer on the unit that they must register the device and that the unit will not be usable after the predetermined time period. Note that in at least one exemplary embodiment the customer would have the opportunity to demo certain features (**806**) not purchased.

In at least one exemplary embodiment, a website **810** is a portal to register the unit purchased by customer **802**. Website **810** can have a web page devoted to registering new devices to simplify the process. Customer **802** may log in (if an account already exists) or provide personal information related to the registering process. A database **812** is accessed that includes both customer and order information in one or more databases. In at least one exemplary embodiment, the unit is in communication with the device manufacturer server **814** through website **810** or coupled thereto through a customer system.

Verification **816** verifies that all information is correct and that there are no pending issues **818** with a corrective action identified. The verification process **816** can checks things like device serial number, features and applications ordered by customer **802**, personal/financial information. Once verification **816** is complete, it is determined at step **817** whether to register the device. Step **817** proceeds to step **820** if there are no pending issues **818**. Registration **820** registers the unit to the owner after verification **816** is complete.

FIG. **9** is a diagram illustrating enabling a device in accordance with at least one exemplary embodiment. In general, registration **902** links customer information to a device as disclosed in FIG. **8**. The registration process includes a comprehensive legally-compliant, state-specific Informed Consent system for collecting and storing the Informed Consent of users.

In at least one exemplary embodiment, the device will have at least one microphone. In the process of registration **902** the customer voice is recorded by the device. The customer voice can be stored on the device and to the customer database. The customer voice can be used for a number of applications including voice verification to use the device thereby acting as a deterrent for others to use or steal the device.

As mentioned in FIG. **8** the unit or device can be enabled for a predetermined time period. Prior to enabling the new device, a contract **904** is provided for customer review. The contract outlines the terms of use for using the device and that the product had been certified through a rigorous testing process. In at least one exemplary embodiment, liability coverage may be provided to the customer if the contract is accepted. Having the liability coverage in conjunction with the certification process is powerful in building consumer confidence about personalized sound management.

At step **905**, it is determined whether the contract is accepted. The new device is not enabled **906** if the customer does not accept contract **904**. The new device is enabled **908** if the customer accepts contract **904**. In at least one exemplary embodiment, the registration and enabling of the unit is through a client system coupled to the business server. Thus, the unit is enabled remotely.

FIG. **10** is a diagram illustrating updating a unit or device in accordance with at least one exemplary embodiment. The device is enabled for use. In general, instructions or an operational video may be provided for a user to get acquainted with the device. Even with this help, a device may have so many features and applications that the user could get overwhelmed or frustrated trying to understand and operate the unit. In some cases, a user may not want even try to learn how to operate all the features and initially start out with the application that provides significant utility to the user. The user may never address learning how to use all the capabilities of the device. Moreover, misuse of a device can lead to dangerous operating conditions that could cause a serious or harmful situation.

Acclimation **1002** is a process to ensure that a user of a device appreciates the entire feature set but more particularly, enables the features in a way that ensures proper usage in a safe and structured manner. In at least one exemplary embodiment, acclimation **1002** comprises two components, a learning module and a reminder module. The learning module is a system that "teaches" the user of the device the

functional components of the device. In at least one exemplary embodiment, the learning module may incorporate a learning system that evaluates the user in the domain of a particular competency before certain functions or features of the device are enabled. For example, features of the device are enabled sequentially, either automatically after the user has utilized a particular function for a predetermined time (thereby showing competency in using the feature) or following a user competency evaluation, which can be invoked either manually (when the user is confident in using the feature) or automatically.

The reminder module may or may not be used in acclimation **1002**. The reminder serves to remind the user of the correct usage of features and also serves to remind the user of other features on the device which the user may not have used in a given time period. In at least one exemplary embodiment an evaluation is invoked based on a predetermined time period. The evaluation may be for a single function or feature or a combination of different functions. The predetermined time period before evaluating can vary depending on the function. Alternatively, the evaluation may be invoked based on a particular recent event, such as a software upgrade, or a detected change in user usage. If the user passes the evaluation, the evaluation may be repeated, the relevant learning mode for the corresponding function to which the users competency was evaluated may be repeated, or the corresponding function to which the users competency was evaluated may be disabled. Thus, acclimation **1002** allows a user to learn the device at a friendly pace, ensures that the user does know how to use a feature, and protects the user from using the device in a manner that could be harmful.

The device can be coupled **1004** from a client system to the business server to update software on the device or purchase new hardware and software. As disclosed hereinabove, the user information is checked to determine if there are any issues that need to be rectified (e.g. missed payment, user does not match to device, etc. . . .). In the event that the issues cannot be rectified the device is disabled in a step **1006**. In at least one exemplary embodiment, the user information, device, and personalized sound management applications correlate. The server checks for updates **1008** for the device and downloads updates to the device. The user goes through a similar process as described in FIGS. **7**A-**7**B for a hardware or software purchase **1010** including agreeing to a contract to subscribe and contract to terms of use as described in FIG. **9**.

FIG. **11** is a diagram illustrating a device for implementing personalized sound management in accordance with at least one exemplary embodiment. The device is generally indicated as an earpiece that partially seals or seals a user's ear canal **1124** and is constructed and operates in accordance with at least one exemplary embodiment of the invention. As illustrated, the earpiece comprises an electronic housing unit **1100** and a sealing unit **1108**. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal **1124** of a user **1130**. The earpiece can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude ear canal **1124**, and is suitable for use with users having healthy or abnormal auditory functioning.

The earpiece includes an Ambient Sound Microphone (ASM) **1120** to capture ambient sound, an Ear Canal Receiver (ECR) **1114** to deliver audio to an ear canal **1124**, and an Ear Canal Microphone (ECM) **1106** to capture and assess a sound exposure level within the ear canal **1124**. The earpiece can partially or fully occlude the ear canal **1124** to provide various degrees of acoustic isolation. In at least one exemplary embodiment, the assembly is designed to be inserted into the user's ear canal **1124**, and to form an acoustic seal with the walls of the ear canal **1124** at a location between the entrance to the ear canal **1124** and the tympanic membrane **1126** (or ear drum). In general, such a seal is typically achieved by means of a soft and compliant housing of sealing unit **1108**.

Sealing unit **1108** is an acoustic barrier having a first side corresponding to ear canal **1124** and a second side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit **1108** includes an ear canal microphone tube **1112** and an ear canal receiver tube **1110**. Sealing unit **1108** creates a closed cavity of approximately 5 cc between the first side of sealing unit **1108** and the tympanic membrane **1126** in ear canal **1124**. As a result of this sealing, the ECR (speaker) **1114** 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 **1126** resulting from the sound field at the entrance to the ear canal **1124**. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

In at least one exemplary embodiment and in broader context, the second side of sealing unit **1108** corresponds to the earpiece, electronic housing unit **1100**, and ambient sound microphone **1120** that is exposed to the ambient environment. Ambient sound microphone **1120** receives ambient sound from the ambient environment around the user.

Electronic housing unit **1100** houses system components such as a microprocessor **1116**, memory **1104**, battery **1102**, ECM **1106**, ASM **1120**, ECR **1114**, and user interface **1122**. Microprocessor **1116** (or processor **1116**) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor **1116** is operatively coupled to memory **1104**, ECM **1106**, ASM **1120**, ECR **1114**, and user interface **1122**. A wire **1118** provides an external connection to the earpiece. Battery **1102** powers the circuits and transducers of the earpiece. Battery **1102** can be a rechargeable or replaceable battery.

In at least one exemplary embodiment, electronic housing unit **1100** is adjacent to sealing unit **1108**. Openings in electronic housing unit **1100** receive ECM tube **1112** and ECR tube **1110** to respectively couple to ECM **1106** and ECR **1114**. ECR tube **1110** and ECM tube **1112** acoustically couple signals to and from ear canal **1124**. For example, ECR **1114** outputs an acoustic signal through ECR tube **1110** and into ear canal **1124** where it is received by the tympanic membrane **1126** of the user of the earpiece. Conversely, ECM **1106** receives an acoustic signal present in ear canal **1124** though ECM tube **1112**. All transducers shown can receive or transmit audio signals to a processor **1116** that undertakes audio signal processing and provides a transceiver for audio via the wired (wire **1118**) or a wireless communication path.

The earpiece can actively monitor a sound pressure level both inside and outside an ear canal **1124** and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warn-

ing sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal **1124** using ECR **1114** and ECM **1106**, as well as an Outer Ear Canal Transfer function (OETF) using ASM **1120**. For instance, the ECR **1114** can deliver an impulse within the ear canal **1124** and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal **1124**. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

In at least one exemplary embodiment, the earpiece has a number of sonic signatures stored in memory. ASM **1120** is providing acoustic information from the ambient environment to processor **1116**. Processor **1116** analyses the acoustic information for a sound similar to the sonic signature. Once identified, the earpiece will provide a response to the sound based on the application. In a first exemplary embodiment, the earpiece will reduce music or telephone call (or the dominant source of sound being provided by the earpiece) and amplify the identified signal (ambulance or police car) thereby notifying the user of the approaching vehicle. In a second exemplary embodiment, the earpiece will tell the user (through a synthesized voice) that an ambulance or police car is approaching including the direction of the vehicle. The earpiece can also provide the identified signal with the voice warning. Other variations are possible.

Conversely, the earpiece can perform the opposite operation. The earpiece can identify a signal similar to the sonic signature and then attenuate it before providing it through ECR **1114**. For example, the user of the earpiece is a gun enthusiast. The user downloads a sonic signature related to a gun shot. The earpiece upon identifying the sound of the gun shot would attenuate the portion of the acoustic information provided by ASM **1120** similar to the sonic signature of the gun shot while allowing other signals to come through. Thus, the user could engage in a conversation at the gun range with the gun shot sounds attenuated while passing the conversation through the earpiece thereby protecting his ear from the loud sounds in this environment and being able to hear the conversation with more clarity.

In at least one exemplary embodiment, the earpiece can manually or automatically record, measure sound pressure levels, attach metadata (including time stamp and geocode), and upload the information when a communication path is present to a sound database. The earpiece is capable of running personalized sound management software disclosed herein. Moreover, hardware ingredients such as the transducers (ECR **1114**, ASM **1120**, and ECM **1106**), processor **1116**, and sealing unit **1108** are provided to manufacturers as disclosed in FIG. **3** for allowing a variety of companies to manufacture devices of their own earpiece designs. It should be noted that although an earpiece is used as an example of a personalized sound management device the components can be used in other communication devices and sound devices to personalize control of the sound.

In at least one exemplary embodiment, sealing section **1108** is a replaceable unit. Sealing section **1108** can pull out and be replaced as a unit. Sealing section **1108** performance can degrade over time due to particulate build up. There may also be sanitary or health reasons for replacing sealing section **1108** periodically. In at least one exemplary embodiment, sealing section **1108** replacement parts can be provided as part of a periodic subscription fee to a user, purchased over the counter in a store, or purchased through a web environment.

FIG. **12** is a block diagram of a device for implementing personalized sound management in accordance with at least one exemplary embodiment. A power supply **1205** powers components of the earpiece including microprocessor/DSP **1206** (or processor **1206**) and a data communication system **1216**. As illustrated, the earpiece can include the processor **1206** operatively coupled through a data communication system **1216** to an ASM **1210**, an ECR **1212**, and ECM **1208**. Data communication system **1216** can include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor **1206** can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) **1202** and Read Only Memory **1204**. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor **1206**. The processor **1206** can also include a clock to record a time stamp.

In general, a data communication system **1216** is a communication pathway to components of the earpiece and components external to the earpiece. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system **1216** is configured to communicate with ECM assembly **1208**, ASM assembly **1210**, visual display **1218**, and user control interface **1214** of the earpiece. As shown, user control interface **1214** can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system **1216** is capable of communication to devices exterior to the earpiece such as the user's mobile phone **1232**, a second earpiece **1222**, and a portable media player **1228**. Portable media player **1228** can be controlled by a manual user control **1230**.

The user's mobile phone includes a mobile phone **1232** communication system **1224**. A microprocessor **1226** is operatively coupled to mobile phone communication system **1224**. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece **1220** worn by another person to the user's mobile phone **1232**. Similarly, the user's mobile phone **1232** can be connected to the data communication system **1216** of the earpiece as well as the second earpiece **1222**. This connection would allow one or more people to listen and respond to a call on the user's mobile phone **1232** through their respective earpieces.

As illustrated, a data communication system **1216** can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor **1206** can provide functionality of the VOX by way of software, such as program code, assembly language, or machine language.

ROM **1204** can be used to store personalized sound management applications to minimize the possibility of modification and tampering of the code. The RAM **1202** can also store program instructions for execution on the processor **1206** as well as captured audio processing data. For instance, memory RAM **1202** and ROM **1204** can be off-

chip and external to the processor **1206** and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor **1206** to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data. The non-volatile memory could also be used to store sonic signatures.

Data communication system **1216** can include an audio interface operatively coupled to the processor **1206** and the VOX to receive audio content, for example from portable media player **1228**, cell phone **1232**, or any other communication device, and deliver the audio content to the processor **1206**. The processor **1206** responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor **1206** (or the VOX of data communication system **1216**) can lower a volume of the audio content responsive to detecting an event such as a sonic signature for transmitting the acute sound to the ear canal of the user. The processor **1206** by way of the ECM **1208** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system **1216**.

The earpiece and data communication system **1216** can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading and uploading over-the-air to the earpiece. It should be noted also that next generation access technologies also can be applied to the present disclosure.

Data communication system **1216** can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece and provide a geocode as an identifier for a recording or measurement such as sound pressure level.

The power supply **1205** 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 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply **1205** to improve sensory input via haptic vibration. As an example, the processor **1206** 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 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 can be reused in different form factors for the master and slave devices.

FIG. **13** is a diagram of a communication device **1302** or earpiece **1310** configured to provide sonic signatures to sonic signature database **1308** in accordance with at least one exemplary embodiment. Collecting a large number of sounds around the world is a daunting task. As mentioned previously, no group or business entity would have the ability to acoustically map the world on a continuous basis. In at least one exemplary embodiment, the collection of sonic signatures is achieved by mobilizing as many people as possible by making it simple to capture and provide a sound for sonic signature database **1308**. This is achieved by adapting a common device having a potential of reaching billions of people for manually or automatically capturing sonic signatures and providing them to database **1308**.

In at least one exemplary embodiment, communication device **1302** is a mobile communication device having a microphone for receiving sound. Examples of a communication device **1302** are a phone, cell phone, PDA, computer, two way radio, smart phone, and earpiece. Earpiece **1310** includes a microphone for capturing sound and the circuitry disclosed in FIG. **12**. Earpiece **1310** is operably configured for recording sound and measuring sound pressure levels. The acoustic information received from the microphone of earpiece **1310** is recorded to memory residing in earpiece **1310** and tagged with metadata including a time stamp and geocode. The sound pressure level (SPL) is measured from the microphone signal through analog or digital signal processing typically before any audio processing occurs within earpiece **1310**. The SPL measurement of this recorded event is associated to the stored recording and metadata. Similarly, communication device **1302** is adapted for capturing sound **1312** including recording, measuring sound pressure level, storing, and tagging with metadata (including a time stamp and geocode). In at least one embodiment, communication device **1302** could have additional circuitry similar to that in FIG. **12** in the device specifically for generating a sonic signature database. Alternately, the circuitry within communication device **1302** can be adapted for recording sound and measuring sound pressure level. In particular, mechanical aspects of the communication device such as microphone placement, microphone porting, as well as electronic audio processing (automatic gain control, equalization, etc. . . .) is taken into account for ensuring an accurate sound pressure level measurement and recording.

In at least one exemplary embodiment, a user having communication device **1302** manually or automatically captures and provides a sonic signature to a sonic signature database **1308**. For example, a microphone on communication device **1302** is always enabled for receiving sound. The sound can be stored in a buffer. The sound in the buffer is analyzed and based on a variety of criteria can be configured to be provided to database **1308**. For example, criteria such as sound pressure level, time, frequency of sound, geographic location, or recognition of a sound (sonic signature detection) are but a few of the parameters that could be used to determine that the sound in the buffer is worthy of saving. Metadata is automatically attached such as a time stamp and geocode but the user can also add information. In at least one exemplary embodiment, a communication path **1304** is opened and a link is made to website **1306** and more particularly to database **1308** where the stored sounds can be automatically uploaded. In at least one exemplary embodiment, the sonic signature, sound pressure level, and metadata could be immediately sent if a communication path **1304** is available to save memory. Further communication between website **1308** and the user of communication device **1302** or earpiece **1310** can take place to edit, identify, describe, and format the provided sound **1312** at a more convenient time. It should be noted that video information that includes audio information can also be provided in

similar fashion as disclosed hereinabove. The audio information from the video can be used for sonic signature database **1308**.

Earpiece **1310** and communication device **1302** can be operably coupled together. A priority could be set up such that earpiece **1310** is the primary recorder of sound **1312** when enabled by the user. Earpiece **1310** can be used with other devices for example a portable media player. Earpiece **1310** can automatically or manually record, measure SPL, and tag metadata of sound **1312** as described hereinabove. The sonic signatures stored in earpiece **1310** could be sent to website **1306** and sonic signature database **1308** if earpiece **1310** is coupled to a communication path **1304** or through another device to which it is operably coupled that has a communication path. Alternately, the sonic signatures could be uploaded via website **1306** and sonic signature database **1308** at a more convenient time, for example a wired or wireless link to the user's personal computer at home, allowing the user to also provide additional metadata before providing the information. Thus, a common device has been provided that is adapted for capturing, storing, measuring SPL, adding metadata including a time stamp and geocode, and uploading the acoustic information to a database thereby including the broadest number of people across the largest geographic area for sound collection on a continuous basis.

FIG. **14** is a block diagram illustrating a cell phone **1404** capturing a sonic signature and providing the sonic signature to a database of sounds **1418** in accordance with at least one exemplary embodiment. Cell phone **1404** is enabled for capturing a sonic signature via an exclusive button or through an automatic process. In at least one exemplary embodiment, the user can select whether a sound is recorded, a sound pressure level measurement is recorded, or both are acquired. A default can be that both are acquired automatically.

A sound **1402** is received by a microphone on cell phone **1404** and stored in a buffer in a record sounds step **1406**. The sound in the buffer is analyzed and determined to be saved. The sound pressure level (SPL) of the sonic signature is measured or calculated in a measure SPL step **1408**. The user can also manually enter metadata **1410** via a keyboard to a metadata table or can enter vocal description in an attached audio stream. Metadata **1410** includes a time stamp and a geocode corresponding to the sonic signature.

In at least one exemplary embodiment, the sonic signature, sound pressure level, and metadata can be stored in memory **1412** that resides on cell phone **1404**. A queue of sonic signatures **1414** can be stored in memory **1412** for uploading at an appropriate time. The user can initiate uploading of the queue of sonic signatures **1414** to database of sounds **1418** when a communication path is completed. In at least one exemplary embodiment, cell phone **1404** can automatically connect **1416** to servers in database of sounds **1418** and upload queue of sonic signatures **1414** when a communication path is enabled. Although stored on database of sounds **1418**, there may be an iterative process to determine if the sonic signatures are in the correct format or are unique enough to be permanently stored. Thus, a large database of sounds **1418** can be collected world wide by an automatic process using a common device such as a cell phone that could not be accomplished by direct means. In at least one exemplary embodiment, the database of sounds **1418** is used in conjunction with personalized sound management applications configured to provide sonic signatures for identifying and providing a response to the identified sound.

To encourage retailers to actively engage and participate in selling Hearium Labs branded products, a business model has been developed based on an annuity revenue sharing methodology. Through this patent pending approach, for the first time, retailers, operators and other suppliers will be able to carry a single audio earpiece or other device in their stores that can be sold to a wide and deep range of consumers. Utilizing an "iPod/iTunes" approach, suppliers will earn profit off the sale of the hardware in store and Personalized Sound Management applications at and after the point of sale. As the consumer continues to personalize their earpiece or device over time, the supplier will earn residuals of up to 50% of the application revenues through a co-operative sharing program between them and Hearium Labs.

To accomplish this, Hearium Labs has enabled the hardware solution to work off of a webified environment that is being architected with commercial flexibility for consumer driven point of sale purchase and activation. This approach allows for the Personalized Sound Management applications to be remotely downloaded and activated on the hardware through any Internet equipped PC and web-based browser. Consumers will thus be able to "enhance" their devices by adding both purchased and subscription applications and personalize their experience with technology designed to acclimate devices to the wearer and their environment.

In general, a process has been provided that eliminates barriers for mass adoption of a new technology. The result is that the technology can be provided by a number of manufacturers thereby having ubiquitous availability at release. Universal compatibility is also provided by using the same hardware and software in each new product having personalized sound management. Certification ensures that each device is manufacturable and performs to specification. Manufacturers and consumers benefit because liability coverage may be provided. Consumers can adapt to the technology rapidly through an acclimation process.

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 such modifications and equivalent structures and functions.

We claim:

1. A system comprising:

a user interface; and

an audio device comprising:

a microphone configured to measure ambient sound and generate a microphone signal;

a speaker configured to emit an audio signal;

a memory configured to store instructions;

an audio buffer configured to store at least a portion of the microphone signal;

a data buffer;

a processor operatively coupled to the memory, the processor operatively coupled to the audio buffer, the processor operatively coupled to the data buffer, wherein the processor is configured to execute the instructions to perform operations comprising:

receiving the microphone signal;

sending a portion of the microphone signal to the audio buffer;

analyzing the audio buffer for detecting a sonic signature;

notifying a user when the sonic signature is detected;

sending a modified microphone signal to the data buffer replacing or adding to previous data stored in the data buffer; and

sending a portion of the data stored in the data buffer, spanning a time period, to the speaker in response to a user request, wherein the user request is one of the sonic signature that is a voice command or a manual input from the user interface.

2. The system according to claim **1**, wherein the sonic signature is a voice command, and where the voice command is to replay audio.

3. The system according to claim **1**, wherein the time period is less than 30 minutes.

4. The system according to claim **1**, wherein the time period is less than 15 minutes.

5. The system according to claim **1**, wherein the audio device is wirelessly connected to an earpiece.

6. The system according to claim **5**, wherein the earpiece includes:

an ambient sound microphone; and

an ear canal receiver.

7. The system according to claim **6**, wherein the earpiece further includes:

an ear canal microphone.

8. The system according to claim **7**, wherein the operation of notifying the user sends a notification signal to the ear canal receiver.

9. The system according to claim **5** wherein the earpiece further includes:

a sealing section.

10. The system according to claim **1**, wherein the sonic signature detected is one of a siren or an alarm.

11. The system according to claim **10** wherein the user interface is a button.

12. The system according to claim **1**, wherein the time period is less than 1 minute.

* * * * *

Exhibit I

US011521632B2

(12) **United States Patent**
Goldstein

(10) **Patent No.: US 11,521,632 B2**
(45) **Date of Patent: Dec. 6, 2022**

(54) **PERSONAL AUDIO ASSISTANT DEVICE AND METHOD**

(71) Applicant: **Staton Techiya LLC**, Delray Beach, FL (US)

(72) Inventor: **Steven Wayne Goldstein**, Delray Beach, FL (US)

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 180 days.

(21) Appl. No.: **16/785,822**

(22) Filed: **Feb. 10, 2020**

(65) **Prior Publication Data**

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(51) **Int. Cl.**
H04R 1/00 (2006.01)
G10L 21/00 (2013.01)
(Continued)

(52) **U.S. Cl.**
CPC ***G10L 21/00*** (2013.01); ***G06F 3/165*** (2013.01); ***G06F 3/167*** (2013.01); ***G06F 16/60*** (2019.01);
(Continued)

(58) **Field of Classification Search**
CPC G10L 21/00; G10L 13/00; G10L 15/26; G10L 21/06; G10L 25/48; G06F 3/165;
(Continued)

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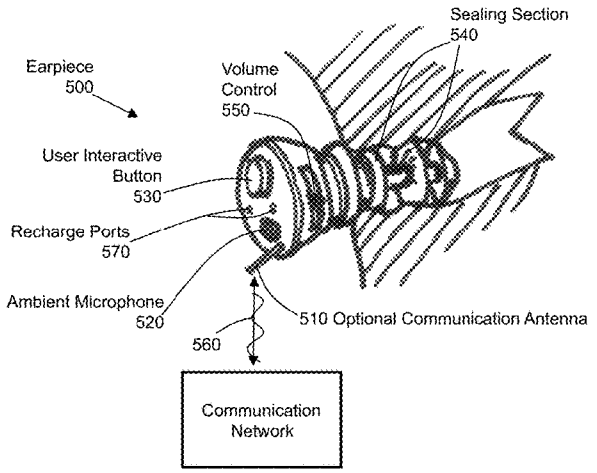
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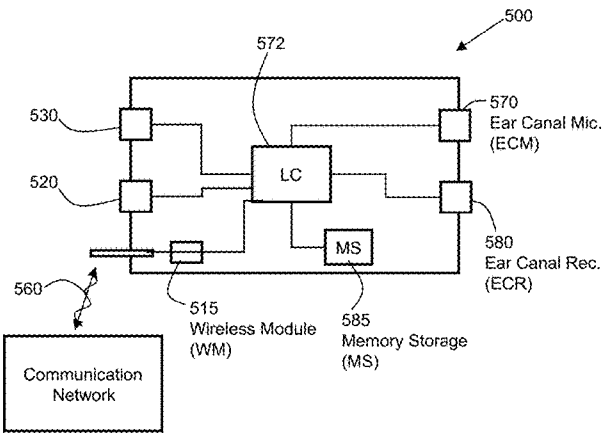
Primary Examiner — Matthew A Eason
(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti

(57) **ABSTRACT**

A system includes a first microphone that captures audio, a communication module communicatively coupled to the first microphone, a logic circuit communicatively coupled to the first microphone and communication module, a speaker operatively coupled to the logic circuit, and an interaction element. The interaction element and logic circuit are configured to initiate control of audio content for output from the speaker in response to at least one voice command detected in captured audio. Other embodiments are disclosed.

17 Claims, 5 Drawing Sheets





Related U.S. Application Data

continuation of application No. 11/774,965, filed on Jul. 9, 2007, now abandoned.

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(51) **Int. Cl.**

H04W 4/80	(2018.01)
G06F 16/60	(2019.01)
H04M 1/05	(2006.01)
H04R 1/10	(2006.01)
G10L 21/06	(2013.01)
H04R 3/00	(2006.01)
G10L 13/00	(2006.01)
G10L 15/26	(2006.01)
G06F 3/16	(2006.01)
G10L 25/48	(2013.01)
H04M 1/247	(2021.01)
H04M 1/60	(2006.01)
H04M 1/72442	(2021.01)

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

CPC ***G10L 13/00*** (2013.01); ***G10L 15/26*** (2013.01); ***G10L 21/06*** (2013.01); ***G10L 25/48*** (2013.01); ***H04M 1/05*** (2013.01); ***H04M 1/2471*** (2013.01); ***H04R 1/1016*** (2013.01); ***H04R 3/00*** (2013.01); ***H04W 4/80*** (2018.02); *H04M 1/6066* (2013.01); *H04M 1/72442* (2021.01)

(58) **Field of Classification Search**

CPC G06F 3/167; G06F 16/60; H04M 1/05; H04M 1/2471; H04M 1/6066; H04M 1/72442; H04R 1/1016; H04R 3/00; H04W 4/80

See application file for complete search history.

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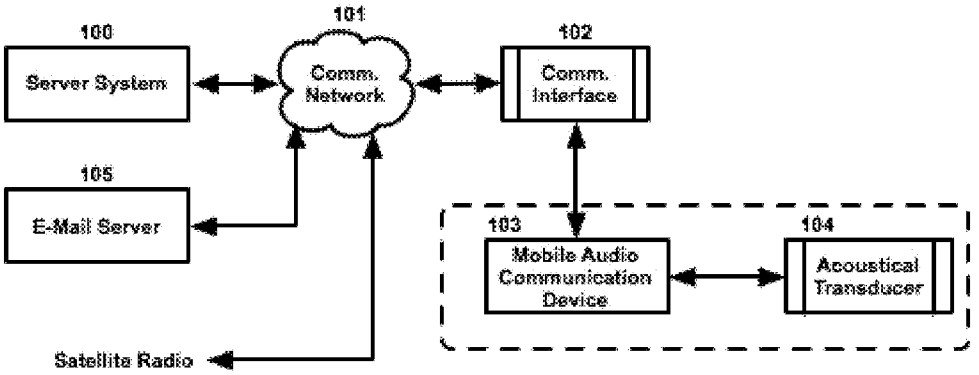

100
Server System
101
Comm. Network
102
Comm. Interface
105
E-Mail Server
103
Mobile Audio Communication Device
104
Acoustical Transducer
Satellite Radio

Fig. 1

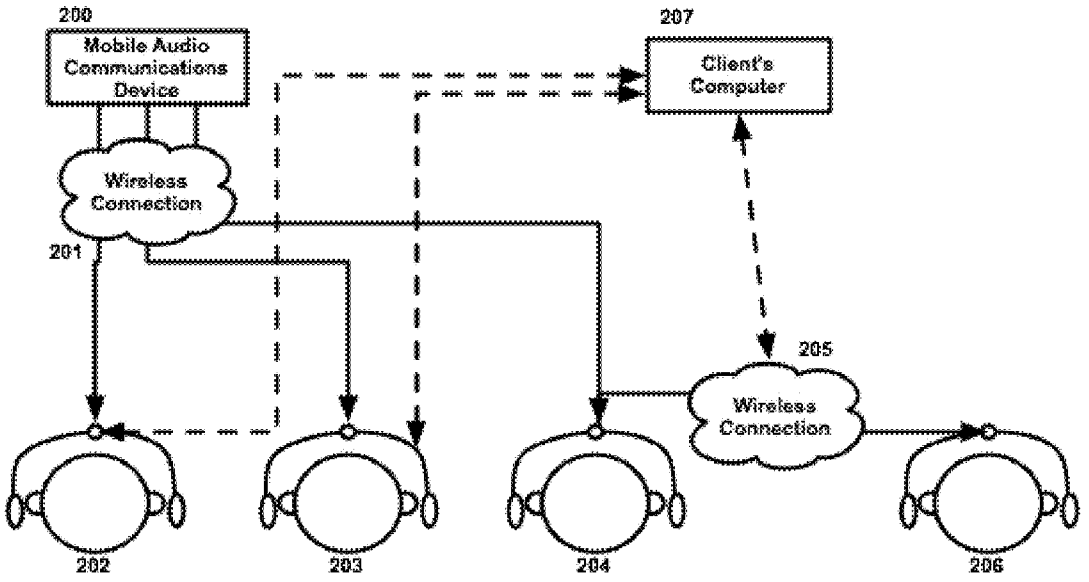

200
Mobile Audio Communications Device
207
Client's Computer
Wireless Connection
201
205
Wireless Connection
202
203
204
206

Fig. 2

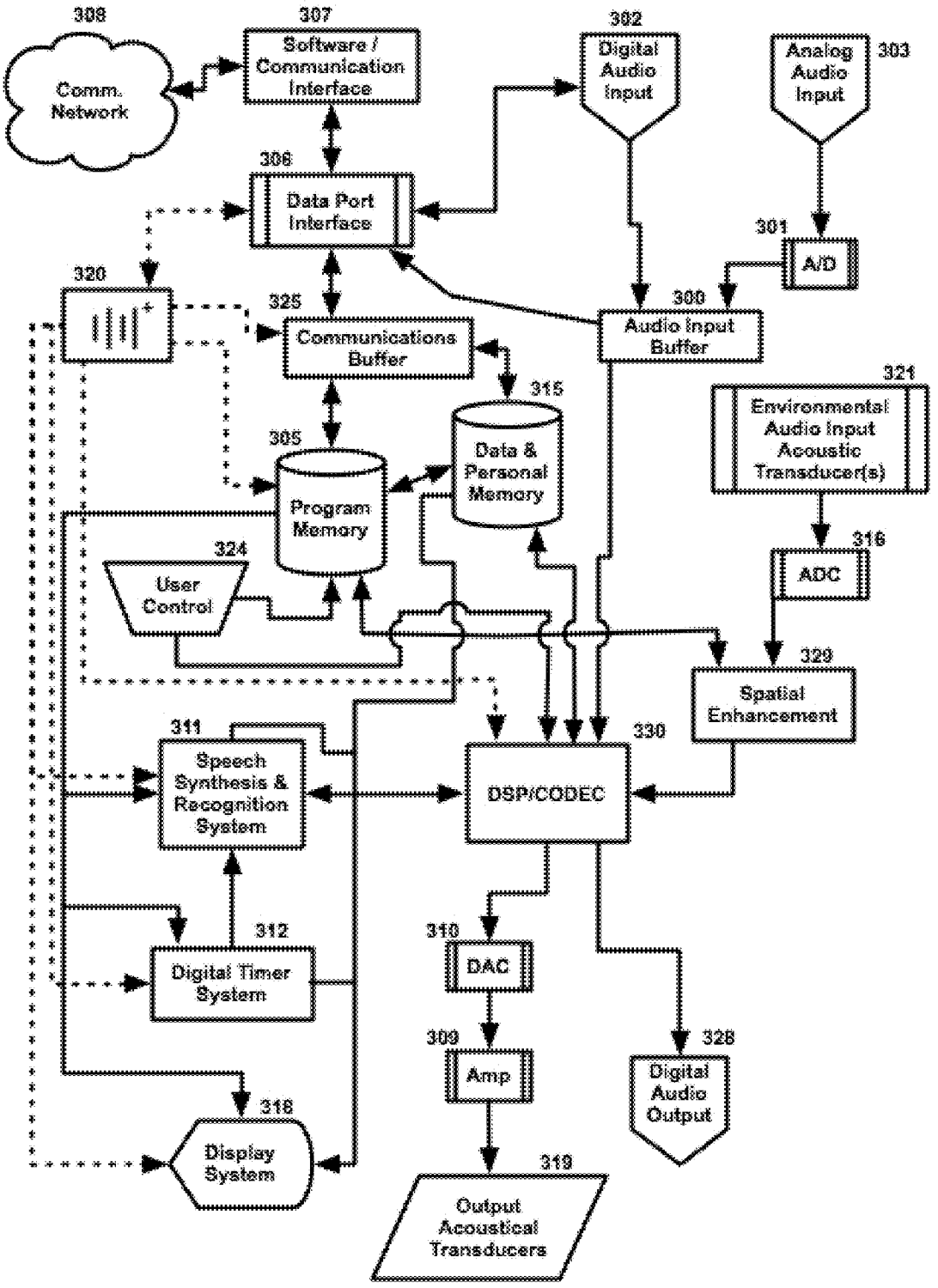

308
Comm. Network
307
Software / Communication Interface
302
Digital Audio Input
303
Analog Audio Input
306
Data Port Interface
301
A/D
320
325
Communications Buffer
300
Audio Input Buffer
321
Environmental Audio Input Acoustic Transducer(s)
315
Data & Personal Memory
305
Program Memory
316
ADC
324
User Control
329
Spatial Enhancement
311
Speech Synthesis & Recognition System
330
DSP/CODEC
312
Digital Timer System
310
DAC
309
Amp
328
Digital Audio Output
318
Display System
319
Output Acoustical Transducers

Fig. 3

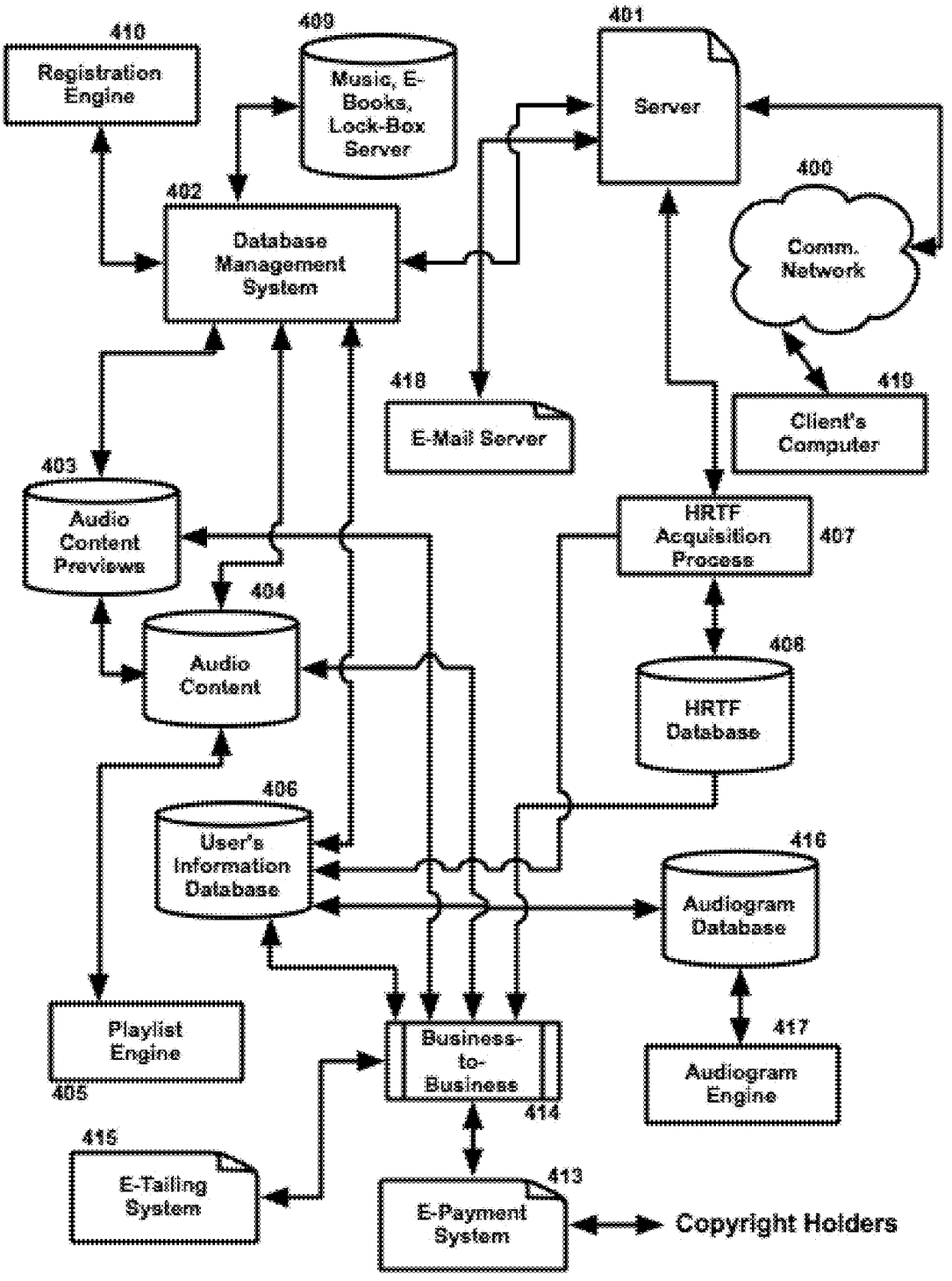

410
Registration Engine
409
Music, E-Books, Lock-Box Server
401
Server
400
Comm. Network
402
Database Management System
418
E-Mail Server
419
Client's Computer
403
Audio Content Previews
HRTF Acquisition Process
407
404
Audio Content
408
HRTF Database
406
User's Information Database
416
Audiogram Database
Playlist Engine
405
Business-to-Business
414
417
Audiogram Engine
415
E-Tailing System
413
E-Payment System
Copyright Holders

Fig. 4

FIG. 5A

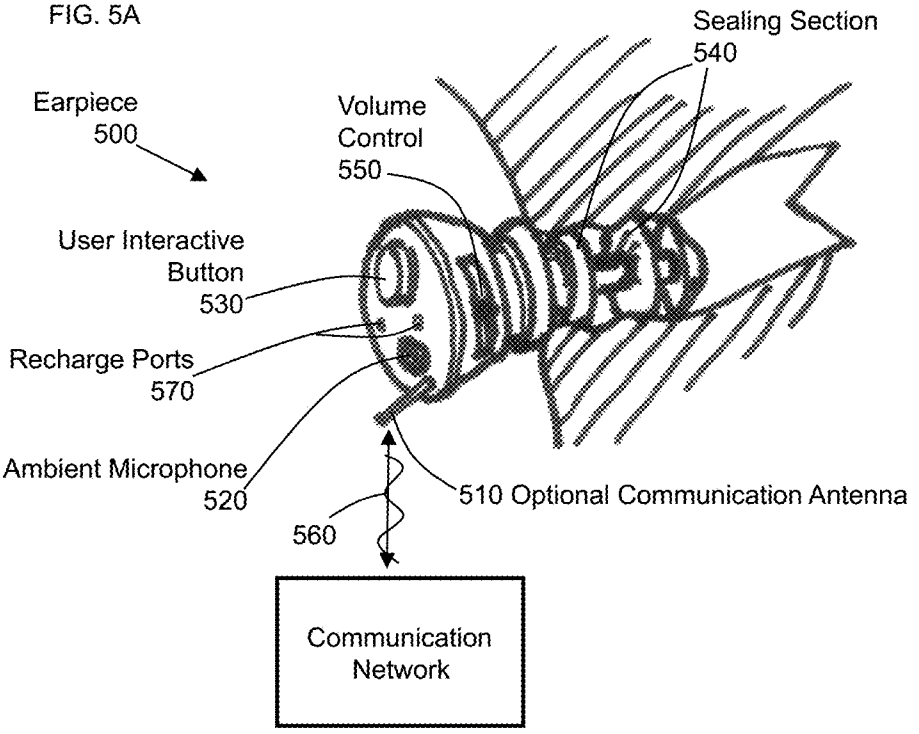

Sealing Section
540
Earpiece
500
Volume
Control
550
User Interactive
Button
530
Recharge Ports
570
Ambient Microphone
520
560
510 Optional Communication Antenna
Communication
Network

FIG. 5B

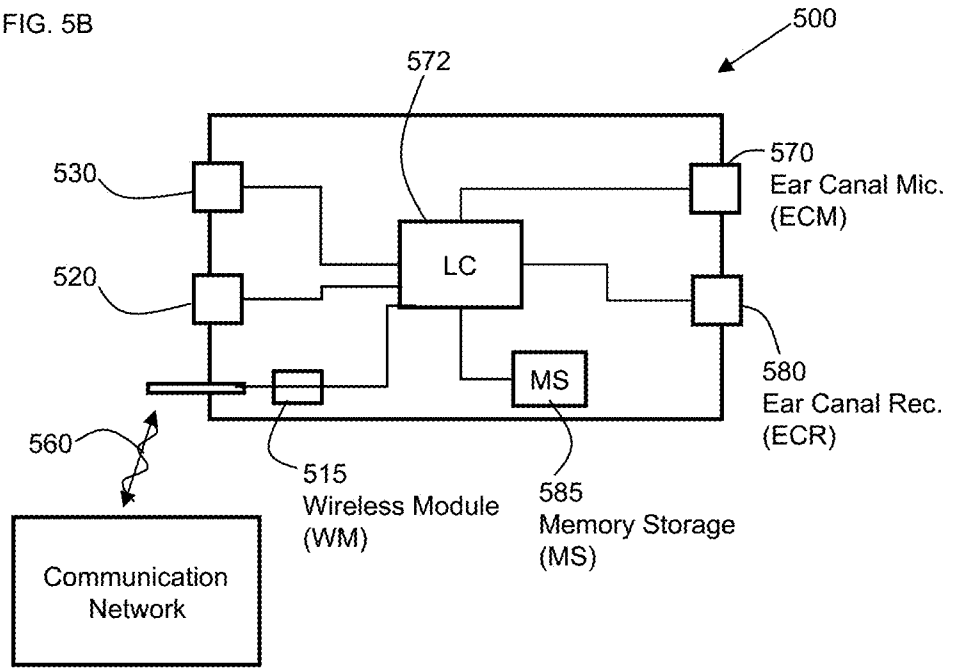
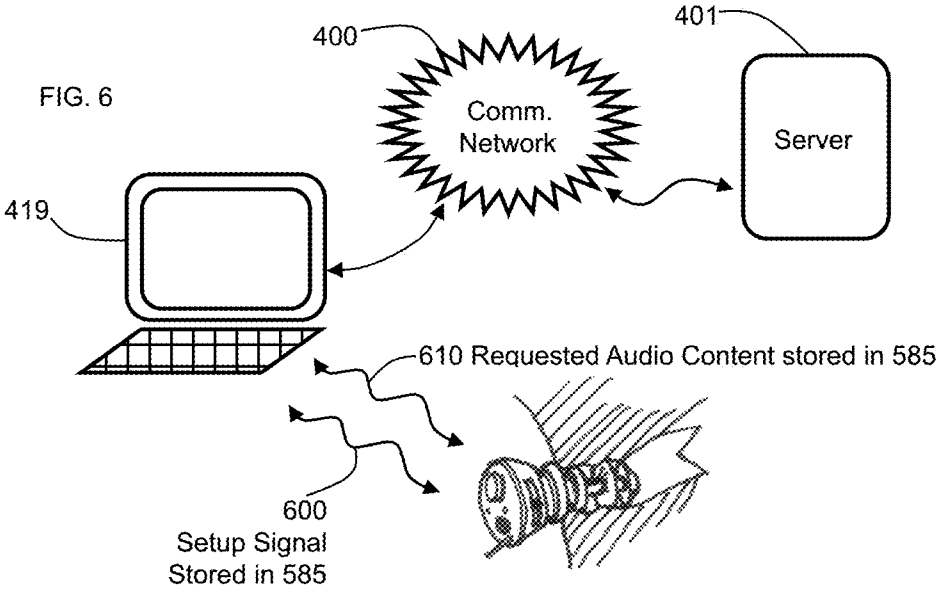

500
572
530
570
Ear Canal Mic.
(ECM)
520
LC
MS
580
Ear Canal Rec.
(ECR)
560
515
Wireless Module
(WM)
585
Memory Storage
(MS)
Communication
Network

FIG. 6


400
401
Comm. Network
Server
419
610 Requested Audio Content stored in 585
600
Setup Signal
Stored in 585

PERSONAL AUDIO ASSISTANT DEVICE AND METHOD

CROSS-REFERENCE TO RELATED APPLICATIONS

This application is a continuation of and claims the priority benefit of U.S. patent application Ser. No. 14/148,749, filed 7 Jan. 2014, which claims the priority benefit of U.S. Provisional Application No. 60/806,769, under 35 U.S.C. § 119(e), filed 8 Jul. 2006, and U.S. Non-Provisional application Ser. No. 11/774,965 filed 9 Jul. 2007, all of which are incorporated herein by reference in their entireties.

FIELD

The claimed embodiments relate in general to methods and devices for the storage and recall of audio content via an earpiece, and in particular, though not exclusively, for the storage and playing of music or verbal content on a system that is built into a headphone.

BACKGROUND

Present audio content playing devices are separated from the headphone system that normally contains the speakers (also referred to as receivers). The reason for this has typically been that audio content has been stored on disks that require a separate playing system. However, even with the advent of storing audio content on non-disk RAM (Random Access Memory) storage systems, the audio content player has been separated from the earpiece system (e.g., plug in headphones or earbuds). Combining the capacity for audio download and playing in an earpiece system is not obvious over related art since the user interaction system (e.g., play button, keyboard system) does not readily appear compatible with the size of an earpiece device and the difficulty of user interaction.

Additionally, no system currently exists for registration and download of audio content into an earpiece.

BRIEF DESCRIPTION OF THE DRAWINGS

Examples of the claimed embodiments will become apparent from the following detailed description, taken in conjunction with the drawings in which:

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network;

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity);

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment;

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer;

FIG. **5**A illustrates an earpiece that can store and download audio content in accordance with at least one exemplary embodiment;

FIG. **5**B illustrates a block diagram of the earpiece of FIG. **5**A; and

FIG. **6** illustrates a user interface for setting the parameters of the Personal Audio Assistant.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS

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

Processes, methods, materials and devices known by one of ordinary skill in the relevant arts can not be discussed in detail but are intended to be part of the enabling discussion where appropriate for example the generation and use of transfer functions.

Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it can not be discussed for following figures.

Note that herein when referring to correcting or corrections of an error (e.g., noise), a reduction of the error and/or a correction of the error is intended.

Summary of Exemplary Embodiments

At least one exemplary embodiment is directed to a system for Personalized Services delivered to a Personal Audio Assistant incorporated within an earpiece (e.g., earbuds, headphones). Personalized Services include content such as music files (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech, speech-to-text processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction, academic instruction, time and date information, speech-to-speech delivery, instructions from a GPS system and others. A Personal Audio Assistant can be an audio playback platform for providing the user with Personalized Services.

At least one exemplary embodiment is directed to a Personal Audio Assistant system that is included as part of an earpiece (e.g., Headphone system). The Personal Audio Assistant is capable of digital audio playback, mitigating the need to carry a personal music player. Furthermore, a subscription-based service provides audio content to the user through the Personal Audio Assistant. The type of audio content, which is automatically provided to the user, is based on the user's preferences, which are obtained through a registration process.

The audio content, which is seamlessly downloaded to the Personal Audio Assistant in the background, is managed from a Server system and is only available on the Personal Audio Assistant for a predetermined period of time or for a fixed number of playback counts. However, the user can purchase any music file or electronic book directly from the Personal Audio Assistant with a simple one-click control interface, storing the purchased audio content on the Personal Audio Assistant as well as storing the content permanently in a user storage lock-box location on the Server system.

The system provides for audio content to be new and "fresh" each time the user auditions the content. As such, the content is typically auditioned in a first-in:first-out scenario. In one such example, the user has turned on the Personal Audio Assistant at 8:00 am and by 10:00 am has auditioned 2 hours of content that were created for the user as a manifestation of the user's choices of their preferences of genre, artist, their demographics, day of the week, time of day and purchase history. The system also provides for the elimination of a particular song or playlist in situ.

As the user's Listening History Envelope is updated based on experience, subsequent downloads will only contain content incorporating these revised preferences. The Personal Audio Assistant provides for ample memory, thus permitting hours of uninterrupted playback without the need to download additional content from the server. When in need, the Personal Audio Assistant automatically interrogates various communication platforms as it searches for connections. Once a connection is made, the Listener History Envelope file is uploaded to the server, and a new set of personalized playlist content is downloaded to the Personal Audio Assistant. Accordingly, as the Personal Audio Assistant content is auditioned and thus depleted, the communications system provides for constant replenishment.

In another embodiment, the Personal Audio Assistant also provides for a new set of business solutions to be offered to the music industry. As the personalized audio content is only available for audition for a limited period of time, and can not be sent to the user again from for weeks to months, the user's purchasing behavior can be demonstrated as spontaneous. The basic model of "Try before you buy" is the expected outcome. In another iteration, the distributor of the music can choose to offer discounts, which can be time-sensitive or quantity-sensitive in nature, in effect promoting greater purchase activity from the user.

In another iteration, while in audition a user can wish to place the desired content in a hold status. The hold status forms the basis of a "wish list," thus allowing the user to hold for future consideration audio content while it is being auditioned. This content resides in the memory of the Personal Audio Assistant for a defined period, and is automatically erased, or the user can do so manually. The selected content will also appear on the user's computer via a URL address; here it resides on the server ready for audition or purchase and download.

The system is designed to operate as simply as possible. Using a single button, which has multiple contacts, the interface allows the user to purchase, delete, skip to next, and add to a wish list and even control a listening level.

In another iteration, the user can download their own music to the Personal Audio Assistant for audition. The Personal Audio Assistant system is capable of text-to-speech processing and can interface with personal scheduling software to provide auditory schedule reminders for the user. Auditory reminders relating to the user's medication schedule are also generated by the system.

At least one exemplary embodiment includes input Acoustic Transducers (microphones) for capturing user's speech as well as Environmental Audio. In further embodiments, stereo input Acoustic Transducers capture Environmental Audio, and, mixing it with the audio signal path, present the ambient sound field to the user, mitigating the need to remove the Headphone apparatus for normal conversation.

Additional exemplary embodiments are directed to various scenarios for the delivery and consumption of audio content. The Personal Audio Assistant can store and play back audio content in compressed digital audio formats. In one embodiment, the storage memory of the Personal Audio Assistant is completely closed to the end-user and controlled from the Server. This allows for audio content to be distributed on a temporary basis, as part of a subscription service. In another iteration of the present embodiments, the storage memory of the Personal Audio Assistant is not completely closed to the end-user, allowing the user to transfer audio content to the Personal Audio Assistant from any capable device such as a Personal Computer or a Personal Music Player.

In at least one exemplary embodiment the Personal Audio Assistant automatically scans for other Bluetooth®-enabled audio playback systems and notifies the user that additional devices are available. These additional devices can include a Bluetooth® video system, television system, personal video player, video camera, cell phone, another Personal Audio Assistant and others.

In another iteration, the Personal Audio Assistant can be directly connected to a Terrestrial Radio receiver, or have such a receiver built in to the system.

In another exemplary embodiment, a technique known as Sonification can be used to convey statistical or other numerical information to a headphone. For example, the user would be able to receive information about the growth or decline of a particular stock, groups of stocks or even sectors of the markets though the Personal Audio Assistant. Many different components can be altered to change the user's perception of the sound, and in turn, their perception of the underlying information being portrayed. An increase or decrease in some level of share price or trading levels can be presented to the user. A stock market price can be portrayed by an increase in the frequency of a sine tone as the stock price rose, and a decline in frequency as it fell. To allow the user to determine that more than one stock was being portrayed, different timbres and spatial locations might be used for the different stocks, or they can be played to the user from different points in space, for example, through different sides of their headphones. The user can act upon this auditory information and use the controls built-in to the headphone to either purchase or sell a particular stock position.

Furthermore, specific sonification techniques and preferences can be presented to the user as "themes" from which the user can select. For example, one theme might auralize the current trading price of one stock with an ambient sine tone in the left ear, the price of another stock in the right ear, their respective trade volumes as perceived elevation using personalized head-related transfer function binauralization, and the current global index or other market indicator as the combined perceptual loudness of both tones. Such a scheme affords ambient auditory display in this example of five dimensions of financial data without compromising the user's ability to converse or work on other tasks. In another embodiment, the system affords users the ability to customize themes to their liking and to rapidly switch among them using simple speech commands. Additionally, the user can search the web from voice commands and receive results via a text to speech synthesizer.

In yet another exemplary embodiment the Personal Audio Assistant (PAA) functions as a dictation device for medical professionals for dictating clinical information to a patient's medical record, or write prescriptions for medication or devices. Conversely, the PAA can function as text-to-speech allowing the clinician to audition information from a medical record, rather than reading. The PAA can save considerable time preparing clinician interaction with a patient.

In another iteration, the Personal Audio Assistant can function as a tool to locate other users of Personal Audio Assistant who share common interests, or who are searching for particular attributes of other users. Whereas the first user has stored specific personal information in the Public Data memory of the Personal Audio Assistant, an example of which might be related to schools attended, marital status, profession etc, or the first user can be in search of another

user with these attributes and whereas a second user of a Personal Audio Assistant comes within communication range of the first user, the individual Personal Audio Assistants communicate with each other, access the personal information stored in each of their respective Public Data memories to ascertain if these users have common interests. If a match occurs, each unit can contain both audible and visual indicators announcing that a match has been made and thus each user can start dialog either physically or electronically via the environmental microphones.

Examples of Terminology

Note that the following non-limiting examples of terminology are solely intended to aid in understanding various exemplary embodiments and are not intended to be restrictive of the meaning of terms nor all inclusive.

Acoustic Isolation Cushion: An "Acoustic Isolation Cushion" shall be defined as a circum-aural or intra-aural device that provides acoustic isolation from Environmental Noise. Acoustic Isolation Cushions can be included as part of a Headphones system, allowing the output of the acoustical transducers to reach the ear unimpeded, but still providing acoustic isolation from Environmental Noise.

Acoustic Transducer: An "Acoustic Transducer" shall be defined as a device that converts sound pressure level variations into electronic voltages or vice versa. Acoustic Transducers include microphones, loudspeakers, Headphones, and other devices.

Audio Playback: "Audio Playback" shall be defined as the auditory stimuli generated when Playback Hardware reproduces audio content (music, speech, etc) for a listener or a group of listeners listening to Headphones.

Audition: "Audition" shall be defined as the process of detecting sound stimulus using the human auditory system. This includes the physical, psychophysical, psychoacoustic, and cognitive processes associated with the perception of acoustic stimuli.

Client: A "Client" shall be defined as a system that communicates with a Server, usually over a communications network, and directly interfaces with a user. Examples of Client systems include personal computers and mobile phones.

Communications Port: A Communication Port shall be defined as an interface port supporting bidirectional transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others) via a communications network (e.g., the Internet, cellular networks).

Control Data: "Control Data" shall be defined as information that dictates the operating parameters for a system or a set of systems.

Earcon: An Earcon shall be defined as a Personalized Audio signal that informs the User of a pending event typically inserted in advance of the upcoming audio content.

Ear Mold Style: "Ear Mold Style" shall be defined as a description of the form factor for an intra-aural device (e.g., hearing aids). Ear Mold Styles include completely in the canal (CIC), in the canal (ITC), in the ear (ITE), and behind the ear (BTE).

Environmental Audio: "Environmental Audio" shall be defined as auditory stimuli of interest to the user in the environment where the user is present. Environmental Audio includes speech and music in the environment.

Environmental Noise: "Environmental Noise" shall be defined as the auditory stimuli inherent to a particular environment where the user is present and which the user does not wish to audition. The drone of highway traffic is a common example of Environmental Noise. Note that Environmental Noise and Audio Playback are two distinct types of auditory stimuli. Environmental Noise does not typically include Music or other audio content.

E-Tailing System: An "E-tailing System" shall be defined as a web-based solution through which a user can search, preview and acquire some available product or service. Short for "electronic retailing," E-tailing is the offering of retail goods or services on the Internet. Used in Internet discussions as early as 1995, the term E-tailing seems an almost inevitable addition to e-mail, e-business, and e-commerce. E-tailing is synonymous with business-to-consumer (B2C) transactions. Accordingly, the user can be required to register by submitting personal information, and the user can be required to provide payment in the form of Currency or other consideration in exchange for the product or service. Optionally, a sponsor can bear the cost of compensating the E-tailer, while the user would receive the product or service.

Generic HRTF: A "Generic HRTF" shall be defined as a set of HRTF data that is intended for use by any Member. A Generic HRTF can provide a generalized model of the parts of the human anatomy relevant to audition and localization, or simply a model of the anatomy of an individual other than the Member. The application of Generic HRTF data to Audio Content provides the least convincing Spatial Image for the Member, relative to Semi-Personalized and Personalized HRTF data. Generic HRTF data is generally retrieved from publicly available databases such as the CIPIC HRTF database.

Headphones: "Headphones" (also known as earphones, earbuds, stereophones, headsets, Canalphones, or the slang term "cans") are a pair of transducers that receive an electrical signal from a media player, communication receivers and transceivers, and use speakers placed in close proximity to the ears (hence the name earphone) to convert the signal into audible sound waves. Headphones are intended as personal listening devices that are placed either circum-aural or intra-aural according to one of the Ear Mold Styles, as well as other devices that meet the above definition such as advanced eyewear that includes Acoustical Transducers (i.e. Dataview). Headphones can also include stereo input Acoustic Transducers (microphones) included as part of the Ear Mold Style form factor.

HRTF: "HRTF" is an acronym for head-related transfer function—a set of data that describes the acoustical reflection characteristics of an individual's anatomy relevant to audition. Although in practice they are distinct (but directly related), this definition of HRTF encompasses the head-related impulse response (HRIR) or any other set of data that describes some aspects of an individual's anatomy relevant to audition.

Informed Consent: "Informed Consent" shall be defined as a legal condition whereby a person can provide formal consent based upon an appreciation and understanding of the facts and implications associated with a specific action. For minors or individuals without complete possession of their faculties, Informed Consent includes the formal consent of a parent or guardian.

Listening History Envelope: "Listening History Envelope" shall be defined as a record of a user's listening habits over time. The envelope includes system data, time system was turned off, time the system is presenting content, when the system doesn't audition, system transducers, when the user auditions content, time stamp of content being auditioned, content which is: skipped, deleted, played multiple times, saved in the Wish List, and time between listening sessions.

Music: "Music" shall be defined as a form of expression in the medium of time using the structures of tones and silence to create complex forms in time through construction of patterns and combinations of natural stimuli, principally sound. Music can also be referred to as audio media or audio content.

Playback Hardware: Any device used to play previously recorded or live streaming audio. Playback Hardware includes Headphones, loudspeakers, personal music players, mobile phones, and other devices.

Personal Audio Assistant: A "Personal Audio Assistant" shall be defined as a portable system capable of interfacing with a communications network, directly or through an intermediate, to transmit and receive audio signals and other data.

Personal Computer: "Personal Computer" shall be defined as any piece of hardware that is an open system capable of compiling, linking, and executing a programming language (such as C/C++, Java™, etc.).

Personal Music Player: "Personal Music Player" shall be defined as any portable device that implements perceptual audio decoder technology but is a closed system in that users are not generally allowed or able to write software for the device.

Personalized HRTF: A "Personalized HRTF" shall be defined as a set of HRTF data that is measured for a specific Member and unique to that Member. The application of Personalized HRTF data to Audio Content creates, by far, the most convincing Spatial Image for the Member (Begault et. al. 2001, D. Zotkin, R. Duraiswami, and L. Davis 2002).

Personalized Services: "Personalized Services" shall be defined as services customized to better meet the needs of an individual. Personalized Services include media content (for preview or purchase) related to a user's preferences, reminders from personal scheduling software, delivery and text-to-speech processing of email, marketing messages, delivery and text-to-speech of stock market information, medication reminders, foreign language instruction [real-time foreign language translation], academic instruction, time and date information, and others.

Public Data: "Public Data" shall be defined as data which contains specific and personal information about the registered user of the Personal Audio Assistant. The registered user chooses which portions of their complete Registration Process data they wish to include in this subset. This data becomes distributed to other users who have compliant devices thus allowing other users to know specific details of the registered user.

Registration Process: "Registration Process" includes the acquisition of the user's preference via a web page. Typically, the process would include the items to be captured: Age, demographics, email, gender, Relative Audiogram, Personal Preferences, banking information, credit card information, wake-up and sleep times, music preferences by genre, artist, preferences for writers and authors, desire to receive advertising, turn-on listening level, equalization, email preferences, parental control setup as well as other user-controlled settings.

Relative Audiogram: A "Relative Audiogram" shall be defined as a measured set of data describing a specific individual's hearing threshold level as a function of frequency. A Relative Audiogram is only an approximate Audiogram, leaving more complete Audiogram analysis to qualified audiologists.

Semi-Personalized HRTF: A "Semi-Personalized HRTF" shall be defined as a set of HRTF data that is selected from a database of known HRTF data as the "best-fit" for a specific user. Semi-Personalized HRTF data is not necessarily unique to one user; however, interpolation and matching algorithms can be employed to modify HRTF data from the database to improve the accuracy of a Semi-Personalized HRTF. The application of Semi-Personalized HRTF data to Audio Content provides a Spatial Image that is improved compared to that of Generic HRTF data, but less effective than that of Personalized HRTF data. The embodiments within speak to a variety of methods for determining the best-fit HRTF data for a particular Member including anthropometrical measurements extracted from photographs and deduction.

Server: A "Server" shall be defined as a system that controls centrally held data and communicates with Clients.

Sonification: "Sonification" shall be defined as the use of non-speech audio to convey information or to aurally perceptualize non-acoustic data (auralize). Due to a variety of phenomena involving human cognition, certain types of information can be better or more efficiently conveyed using auditory means than, for example, visual means.

Exemplary Embodiments

FIG. **1** illustrates the connection between an earpiece device (**103** and **104**) and a communication network (**101**) via communication interface **102**, which can be operatively connected (via a wired or wireless connection) to a server system (**100**) and/or an e-mail server (**105**). Additionally a radio signal (e.g., satellite radio) can be input into the earpiece **500** (FIG. **5**B) via a communication module (e.g., Bluetooth® wireless module **515**).

FIG. **2** illustrates at least one exemplary embodiment where earpiece devices share information with other earpiece devices within range (e.g., GPS location and identity). For example multiple users (e.g., **202**, **203**, **204**, and **206**) can send signals to each individual earpiece (e.g., **500**) when in range (e.g., via a wireless connection **205**) or to a mobile audio communications device **200** via a wireless connection (**201**) with each earpiece (**500**). Additionally information (e.g., audio content, software download) can be sent via a client's computer **207** to each earpiece, either directly (e.g., **205**), or via **200**. For example audio content can be retrieved on a user's computer and sent to the earpieces that have authorization to use it.

FIG. **3** illustrates an example of various elements that can be part of an earpiece device in accordance with at least one exemplary embodiment. The earpiece can include all or some of the elements illustrated in FIG. **5**B. For example the logic circuit **572** or the operatively connected memory storage device **585**, can include spatial enhancement software **329**, a DSP codec **330**, a speech synthesis and recognition system **311**, and a digital timer system **312**. Additional elements can be connected to the logic circuit **572** as needed, for example a power supply **320**, a software communication interface **307** (e.g., wireless module **515**) (which may be connected to communication network **308**), data port interface **306**, audio input buffers **300** connected to digital audio input **302** and/or analog audio input **303** converted to digital via an ADC **301**, environmental audio input acoustic transducer(s) **321** converted to digital via an ADC **316**, user control **324**, digital audio output **328**, output acoustic transducers **319** (which receive signals converted to analog via a DAC **310** and amplified via amplifier **309**), display system **318**, communication buffers **325**, program memory **305**, data and personal memory **315**, as well as other electronic devices as known by one of ordinary skill in the relevant arts.

FIG. **4** illustrates an example of a communication system in accordance with at least one exemplary embodiment that a user can use to register via his/her computer **419**, via a communication network **400** (e.g., Internet connection) connected to many various database and registration systems as illustrated and labeled in FIG. **4**. For example, server **401**, database management system **402**, audio content preview database **403**, audio content database **404**, playlist engine **405**, user's information database **406**, HRTF acquisition process module **407**, HRTF database **408**, lock-box server **409**, registration engine **410**, e-payment system **413**, business-to-business module **414**, e-tailing system **415**, audiogram database **416**, audiogram engine **417** and/or e-mail server **418**.

FIG. **5**A illustrates an earpiece **500** having sealing section **540** in an ear canal that can store and download audio content **560** in accordance with at least one exemplary embodiment. The earpiece **500**, can include a first user interaction element **530** (e.g., a button), that can be used to turn the earpiece **500** on, or if on then activate an audio play command to start playing saved audio content. The earpiece **500** can also include a second user interaction element **550** (e.g., a slide control) that can be used for example to control the volume. The earpiece **500** can also include recharge ports **570**, that can accept two wires of varying voltage that can be inserted into the recharge ports **570** to recharge any batteries in the earpiece **500**. The earpiece **500** can include an ambient microphone **520** and an optional communication antenna **510**, that if needed can aid in the communication between the earpiece **500** and a communication network.

FIG. **5**B illustrates a block diagram of the earpiece of FIG. **5**A, illustrating the first user interaction element **530**, the ambient microphone (AM) **520**, that can be used to pick up ambient audio content, an ear canal microphone (ECM) **590** that can pick up audio in the ear canal region, an ear canal receiver (ECR) **580** that can direct audio content to the ear drum, all of which can be connected operatively to a logic circuit **572**. A memory storage device **585** can be operatively connected to the logic circuit (LC) **572**, and can store data such as registration, preference, and audio content data. The optional communication antenna **510** can be connected to a communication module (e.g., wireless module **515**), and can receive or transmit information **560** to a communication network.

FIG. **6** illustrates a user interface for setting the parameters stored in the memory storage device **585**. For example a user can use his/her computer **419** to communicate with a server **401** (e.g., via a communication network **400**) to start the user's registration (e.g., with an audio content provider). The registration information can then be transmitted **600** to set the stored parameters in the memory storage device **585** of the earpiece **500**. Additionally a requested (e.g., bought) audio content can be downloaded **610** into the memory storage device **585** of the earpiece **500**.

At least one exemplary embodiment is directed to an earpiece comprising: an ambient microphone; an ear canal microphone; an ear canal receiver; a sealing section; a logic circuit; a communication module; a memory storage unit; and a user interaction element, where the user interaction element is configured to send a play command to the logic circuit when activated by a user where the logic circuit reads registration parameters stored on the memory storage unit and sends audio content to the ear canal receiver according to the registration parameters.

In at least one exemplary embodiment the audio content is stored in the memory storage unit. The earpiece according to claim **2**, where the communications module is a wireless communications module. Additionally the earpiece can include a second user interaction element configured to alter the volume of the audio content that is emitted from the ear canal receiver.

Upon a play command being received by the logic circuit the logic circuit can check registration parameters stored in the memory storage device for example one of the registration parameters can be whether the audio content is a sample audio content or a fully purchased audio content, or the allowed number of times an audio content can be played, and a counter value that keeps track of the number of times the audio content has been played.

The earpiece can send an auditory warning to be emitted by the ear canal receiver when the counter value is greater than or equal to the allowed number of times the audio content can be played, and where the logic circuit does not send the audio content to the ear canal receiver.

Further Exemplary Embodiments

At least one exemplary embodiment is directed to a system for the delivery of Personalized Services to Personal Audio Assistants, the system comprising: a Personal Audio Assistant system for presenting Personalized Services to the user as Audio Playback; a Server system for user registration, Personalized Service management, and communication; a Registration Process for collecting detailed registration information from users, including the information necessary for creating Personalized Services; a communications protocol (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to the Server or connecting the Personal Audio Assistant to other Personal Audio Assistants (peer-to-peer behavior).

In at least one exemplary embodiment a Personal Computer acts as an intermediate, connecting to the Server system over a communications network and connecting to the Personal Audio Assistant over a local connection. At least one exemplary embodiment includes a Personal Hearing Damage Intervention System (e.g., USPTO—60/805,985—Goldstein).

In at least one exemplary embodiment a Personal Audio Assistant system included as part of a Headphone system, the system comprising: a Communications Port supporting a communications protocol enabling communication with the Server system, peer devices, and other capable devices; a non-volatile program memory storage system for storing Control Data, dictating system behavior; a data memory storage system for storing data and audio content; an analog audio input/output and corresponding ADC/DAC; a digital audio input/output and a digital audio signal path; a user control system allowing the user to adjust the level of the audio output and control the behavior of the system; a user control system allowing the user to purchase the content being auditioned in real time; a user control system allowing the user to control, delete, fast forward, output level control, scan, advance, the data stored both stored in memory as well as new streaming data emails and reminders; a display system for presenting information to the user(s) visually using any method familiar to those skilled in the art (LED, LCD, or other); a display system for presenting information to the user(s) (e.g., using Earcons and other sound files); a speech synthesis system for converting text-to-speech and generating speech signals; a speech recognition system for converting speech to-text to respond and send emails and to interface with the control language as to provide naviga-

tional commands; a digital timer system; a power supply system in the form of a battery; a unique identification number for each Personal Audio Assistant; Input Acoustic Transducers; an Output Acoustic Transducer; an Audio amplification system; Acoustic Isolation Cushions conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE; see definitions) and other elements common to Headphone systems; a digital signal processor (DSP) system; and a CODEC processor capable of improving the perceptual sound quality of the content to be auditioned while governed by delivering the correct SPL dose.

In at least one exemplary embodiment the system is independent of a Headphone array or can be included and imbedded as part of a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the various processing needed to derive the intended functions are distributed among any combination of a Server system, a Personal Computer system, a Personal Music Player system, a personal monitoring system, an automotive audio system, a home audio system, an avionics audio system, a personal video system, a mobile cell phone system, a personal digital assistant system, a standalone accessory, or an advanced eye-wear system with acoustical transducers.

In at least one exemplary embodiment the Personal Audio Assistant system can exchange audio signals with a mobile phone via the Communications Port, allowing the Personal Audio Assistant to function as a mobile phone accessory.

In at least one exemplary embodiment a communications buffer is included. For example when a network connection is available, the communications buffer uploads stored content (e.g., Listening Habits Envelope) and stores incoming transmissions (e.g., music, electronic books, and updates to the firmware or operating system) from the Communications Port; The contents of the communications buffer are then transmitted whenever a network connection becomes available. At least one exemplary embodiment includes a perceptual audio codec decoding technology in the DSP, enabling the storage and playback of compressed digital audio formats (e.g., MP3, AAC, FLAG, etc.). At least one exemplary embodiment is compliant and compatible with DRM, FairPlay® and other forms of digital content governance.

At least one exemplary embodiment includes a user control system for selecting and playing back audio content stored in memory that operates using any combination of the following methods: a button or tactile interface which upon auditioning a song can be pressed to order content; a button, tactile and/or voice controlled interface which, when pressed once, commanded to, activates playback of short audio clips or audio thumbnails of the audio content stored in memory; When the button is pressed again during audio thumbnail playback, the current audio content selection is played in its entirety; The behavior of this interface is similar to the "scan" button interface common in FM/AM radio devices; a button, tactile and/or voice controlled interface that, when pressed or commanded to, skips to the next piece of audio content, which is selected randomly from all available audio content that has a play count equal to or less than the play count of the piece of audio content currently playing; The behavior of this interface is similar to the "shuffle" behavior found in some personal music players; an interface for browsing audio content storage devices familiar to those skilled in the art; and a process to allow for increased data memory storage capacity for storing audio content.

In at least one exemplary embodiment the contents of the data memory are encrypted and controlled by the Server system only, prohibiting the end-user from loading unauthorized audio content into the data memory. Further the contents of the data memory can be manipulated by the end-user, allowing the user to transfer audio content to the Personal Audio Assistant system from any device capable of interfacing with the communications port; For example, audio content can be transferred to the system from a Personal Music Player or a Personal Computer. According to at least one exemplary embodiment, audio content (or other media content) updates are retrieved from the Server system any time a connection is detected by the communications port. Furthermore, an exemplary embodiment can include an acoustical and/or visual indicator informing the user when a transfer of data is activated.

In at least one exemplary embodiment radio wave transmissions are used to implement some communications protocol and the communications port acts as a radio receiver. Additionally the Personal Audio Assistant can include: interfaces with some personal scheduling software through the communications port; a speech synthesis system which generates speech-signal reminders corresponding to information from the scheduling software, where the digital timer system triggers the presentation of the speech-signal reminders at the appropriate time.

Additionally the Personal Audio Assistant can interface with an email platform through the communications port; The speech synthesis system converts the email in text to speech and provides email to the user in aural presentation format. The system further comprising: a process in the Registration engine allowing the user to optimize their personalization process of incoming emails by associating a specific Earcon with the importance of the incoming email. As such, normal priority email contains an introduction sound announcing to the user the level of importance the sender associated with their email; a speech recognition system for converting speech-to-text which interfaces with the control language as to provide navigational commands allowing the user to respond and send emails.

In at least one exemplary embodiment the communications port system makes use of some wireless communications protocol (802.11, Bluetooth®, A2DP, or other) to transmit and receive digital audio data for playback, the system further comprising: an audio codec to encode and decode digital audio transmissions; a wireless communications system (802.11, Bluetooth®, A2DP, etc.) for transmitting and receiving data (digital audio transmissions, Control Data, etc.); a method for pairing two or more Personal Audio Assistants through a wireless communications protocol to provide a secure exchange of audio content, data such as the user's Public Data; an audio warning signal or visual display system output that notifies the user anytime a compatible transmission becomes available; and a user control system enabling the user to switch between available compatible transmissions.

In at least one exemplary embodiment the system enables listeners to share digital audio transmissions, the system further comprising: a method for scanning for available digital audio transmissions within range; a user control interface for specifying digital audio transmission behavior; a method for employing the system as a relay to other compliant devices; re-broadcasting digital audio transmissions to increase wireless range. In at least one exemplary

embodiment multiple systems are capable of sharing the contents of their program and data memory using the wireless communications protocol.

In at least one exemplary embodiment of the system, the input Acoustic Transducer is used to record audio content to the data memory storage system, the system further comprising: an implementation of some perceptual audio codec technology in the DSP, enabling the storage of compressed audio formats (e.g., MP3, AAC, FLAG, etc); and an increased data memory storage capacity for storing recorded audio content.

In at least one exemplary embodiment, the stereo input Acoustic Transducers are ultimately connected to the audio signal path at the DSP, allowing the user to audition Environmental Audio (e.g., speech or music) and mitigating the need for the user to remove the Headphone apparatus to audition Environmental Audio, the system further comprising: a stereo pair of input Acoustic Transducers placed close to the user's ear canal input, conforming to one of the Ear Mold Styles (CIC, ITC, ITE, or BTE, see definitions); and by mounting the input Acoustic Transducers in a CIC or ITC configuration, spatial-acoustic cues are preserved, creating a spatially-accurate Environmental Audio input signal—essentially a personal binaural recording; a method for acoustically compensating for the non-linear frequency response characteristics of the Acoustical Isolation Cushions of a given Headphone system by applying corresponding inverse filters to the Environmental Audio input signal at the DSP; With this method, the system acts as a linear-frequency-response hearing protection apparatus (e.g., USPTO-60805985-Goldstein).

At least one exemplary embodiment includes a system for first attenuating Audio Playback and then mixing the Environmental Audio input signals, at a louder listening level, with the audio signal path using the DSP, where the system is activated by any combination of the following methods: a manual switch to activate/deactivate the system; a speech-detection apparatus to activate the system when speech is detected as the principal component of the Environmental Audio input; and a music-detection apparatus to activate the system when music is detected as the principal component of the Environmental Audio input.

At least one exemplary embodiment can include active noise reduction, echo cancellation and signal conditioning that can be environmentally customized through the registration process to better meet the user's specific needs (i.e., occupation-related noise cancellation); A typical application would be a special set of noise cancellation parameters tuned to the drilling equipment used by a dentist.

In at least one exemplary embodiment the input Acoustic Transducers are instead mounted within circum-aural, intra-aural BTE, or intra-aural ITE molds (see Ear Mold Style), the system further comprising: a spatial audio enhancement system for supplementing the spatial-acoustic cues captured by the stereo pair of input Acoustical Transducers to provide improved spatial perception of Environmental Audio using any combination of the following methods: the application of Generic, Semi-Personalized, or Personalized HRTF data to the Environmental Audio input signal; the application of binaural enhancement algorithms, familiar to those skilled in the art, to the Environmental Audio input signals; the application of a pinna simulation algorithm to the Environmental Audio input signal; and a synthetic pinna apparatus placed just before the stereo input Acoustic Transducers.

At least one exemplary embodiment includes a Server system for the creation, Registration, management, and delivery of Personalized Services, the system comprising: a communications system for interfacing with public communication networks to exchange data with Personal Audio Assistants, a Client's computer, mobile phones, PDAs or other capable devices; a database and database management system for storing and retrieving information relating to user Registration, Personalized Services, audio content, Control Data, and other data; a Registration interface system for collecting, storing, and applying information provided by users; a method for creating Personalized Services based on user Registration information; an end-user audio content Lock-Box storage system, providing every registered user access to their purchased media content; a business-to-business interface system for acquiring audio content with record labels, copyright holders, and other businesses; an E-tailing system including an electronic transactions system enabling users to purchase content, items offered for sale or pay subscription fees electronically; an E-Payment system compensating the various copyholders upon purchase of content by user; a Playlist engine, which acquires the user's Registration information, Listening History Envelope and then creates audio playlists, which is optimized for the user preferences and further refinements; and an Email server, which distributes communications to the user and others, regarding marketing data, the status of the user weekly SPL dose, and other information.

At least one exemplary embodiment includes machine-learning techniques employed to better optimize the user's preferences relating to audio content and other media content, the system further comprising: a method for tracking the purchase history of each user, relating the purchase history to media content preferences, and using the purchase history to make media content recommendations; a method for examining a user's digital media library, stored on a Personal Computer, Personal Music Player, or Personal Audio Assistant, from the Server system, and relating media content preferences and media content recommendations to the user's digital media library; and a method for examining a user's Listening History Profile.

At least one exemplary embodiment includes a Registration system for collecting a wide variety of information from users, including information necessary for creating Personalized Services, the system comprising: a Server system; an interface system for querying the user to collect registration information including demographics (age, gender), Playback Hardware information, Headphone information, occupational information, home and work locations, medication information, music-related preferences, video-related preferences, and other information; a method for customizing Control Data based on registration information; and a method for creating Personalized Services based on registration information.

In at least one exemplary embodiment a fast HRTF acquisition process is included as part of the Registration process, the system further comprising a method for the fast acquisition of Semi-Personalized HRTF data via a deduction process, the method comprising: a database system containing indexed, clustered HRTF data sets; an auditory test signal with distinctive spatial characteristics, where two or more distinct sound source locations exist; a system for the application of potential HRTF matches to the auditory test signal; and a feedback system, allowing the user to select the best listening experience from a number of candidate listening experiences, based on the spatial quality perceived in the HRTF-processed auditory test signal.

In at least one exemplary embodiment Personalized HRTF data is measured and used instead of Semi-Personalized HRTF data, by any method familiar to those skilled in the art.

In at least one exemplary embodiment the user is provided some Personal Audio Assistant free-of-charge or at a discount, given the user agrees to a subscription service commitment to receive Personalized Services for a certain amount of time.

In at least one exemplary embodiment, as part of the Personalized Services, the user is provided with temporary audio content corresponding to the preferences indicated during the registration process; Further, the user is given the option to purchase the audio content permanently; Otherwise, the audio content is replaced with new audio content from the Server, after a predetermined amount of time or a predetermined number of playback counts, the system comprising: a Personal Audio Assistant with an enhanced user control system, enabling a registered user to purchase media content directly from the Personal Audio Assistant with a button; and a Personal Audio Assistant with an enhanced user control system, enabling a registered user to store a reference to media content that can be purchased by the user at a later time.

In at least one exemplary embodiment, video or gaming content is included as well as audio content, the system further comprising: a Personal Audio Assistant with an enhanced visual display system, capable of playing video and/or gaming content.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives medication reminders in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; Medication reminders are generated by the Server system based on the user's registration information.

In at least one exemplary embodiment, as part of the Personalized Services, the user receives stock market information in the form of speech signals, audio signals, text, or graphics on the user's Personal Audio Assistant; The stock market information is selected by the Server system based on the user's registration information, the system further comprising: the user having successfully registered their Personal Audio Assistant with a brokerage firm, or other stock trading engines, the user can then purchase or sell a stock by use of a user button or a speech command.

Further in at least one exemplary embodiment, the user is able to request specific media content to be transferred temporarily or permanently to the user's Personal Audio Assistant, the system further comprising: an interface system operating on the Server allowing users to request specific media content by artist, title, genre, format, keyword search, or other methods familiar to those skilled in the art; and a media content search engine system.

In at least one exemplary embodiment a Relative Audiogram compensation filter is applied to audio signal path by the digital signal processor, the system either (e.g., USPTO-60805985-Goldstein): (a) Retrieves Relative Audiogram compensation information from a remote Server after a registration process (during transmission, the information can include HIPAA compliant encoding); or (b) calculates a compensation filter from Relative Audiogram information obtained by the system locally. For example U.S. Pat. No. 6,840,908-Edwards, and U.S. Pat. No. 6,379,314-Horn, discuss methods for the acquisition of an individual's Relative Audiogram.

In at least one exemplary embodiment a Satellite Radio transmitter/receiver (transceiver) is incorporated within the Headphone proper, allowing the user to at least: receive XM®, Sirius® and other broadcasts for playback over the system; select radio stations for playback over the system via the control system, the control system comprising either a single-click tactile interface or the speech-controlled circuitry; store selected portions of such broadcasts in memory for later recall and playback via the control systems; engage a novel commercial-skip feature for attenuating the playback level of suspected sales commercials broadcasts; and engage a speech-skip feature for attenuating the playback of speech (e.g., news, announcements, etc.).

At least one exemplary embodiment includes a Walkie-Talkie mode, which broadcasts input to the system's built-in microphone, whereby the user's speech can be detected by the input acoustic transducer and remotely broadcast where at least one of the following occurs: the Walkie-Talkie mode receives input via AM/FM broadcasts (as well as digital communications protocols) from a nearby user; the Walkie-Talkie mode allows nearby users to engage in conversation with increased perceptual clarity in noisy environments (e.g., aircraft cockpits), using for example a noise-cancellation system; selectively engage and disengage the Walkie-Talkie mode using the control system; detect other users of the system within a given range; and alert the user of the system when other detected systems contain certain Public Data and contain a predefined Public Message Key (e.g., "If the detected system belongs to a single male between the ages of 25 and 30 and whose favorite sport is tennis, then broadcast the message, 'I like tennis also; would you like to have coffee?'" or "If the detected system belongs to a user who attended Princeton University, then broadcast the message, 'Go Tigers!'").

At least one exemplary embodiment can use other communications to accomplish this service rather than AM/FM; as such the system can incorporate communications transmission protocols (TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or other protocols) and a communications network (i.e. the Internet, cellular networks) connecting the Personal Audio Assistant to other Personal Audio Assistants. At least one exemplary embodiment can selectively control the broadcast of public data and public message keys via the control system.

At least one exemplary embodiment includes a Sonification algorithm within the Headphone, which enables auditory display of digitally received data, including for example financial data, news, GPS data, the system further containing a variety of sonification "themes" selected during the registration process that map requested data (e.g., current average trading price of AAPL stock, the Dow Jones Industrial Index, and the Nasdaq Composite®) to corresponding audio content (e.g., the frequency of a sine tone presented in the left ear, the frequency of a sine tone presented in the right ear, and the global amplitude of both sine tones, respectively).

At least one exemplary embodiment includes an auditory display, which is synthesized by the onboard Digital Signal Processor. In at least one exemplary embodiment the auditory display is created through the digital audio signal processing effects applied to any other acoustic data the system is capable of reproducing (e.g., terrestrial radio, prepurchased audio content in the user's digital library, electronic books, etc.); For example, a sudden listening level increase in the playback level of a song to which the user was listening can be triggered by a predefined alert condition (e.g., Nasdaq Composite® has exceeded 2200 points).

At least one exemplary embodiment includes the ability to create themes using a computer program and uploading a file to the Headphone system.

At least one exemplary embodiment includes a speech recognition system for converting speech to HTML (Voice Browser), whereby the user can access the Internet, provide navigational commands, perform searches and receive results via the Headphones through a text (HTML)-speech synthesize.

Additionally, the Personal Audio Assistant can be totally incorporated with a mobile cell phone, or any portable technology which incorporates any of the following protocols, TCP/IP, USB, IEEE 1394, IEEE 802.11, Bluetooth®, A2DP, GSM, CDMA, or others known to those of ordinary skill in the arts via a communications network (e.g., the Internet, cellular networks), the system further comprising: an Acoustic Transducer constructed as part of the mobile cell phone or a series of Acoustic Transducers, which are constructed as part of mobile cell phone; a commutations path incorporated into the mobile cell phone providing for bi-directional communication with a Headphone array; the incorporation of the mobile cell phone's microphone(s) to act as the Environmental Audio Acoustical Transducer(s); and the incorporation of the mobile cell phone's keyboard or touch sensitive screen to function as a manual input or to complement speech commands and that can act in a way to respond to Personalized Services offered to a user. While the present embodiments have been described with reference to exemplary embodiments, it is to be understood that the claimed embodiments are 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 such modifications and equivalent structures and functions.

I claim:

1. An earpiece, comprising:

a sealing section configured to be inserted into and seal an ear canal of a user;

an ambient microphone configured to measure ambient audio content in the ambient environment near the user's ear;

an ear canal microphone configured to measure internal audio in the ear canal of the user;

a speaker;

an interaction element, wherein the interaction element is a tactile interface;

a location unit configured to determine the location of the earpiece;

a memory that stores instructions; and

a logic circuit configured to execute the instructions to perform operations, the operations

comprising:

generating active noise reduction using at least one of the measured ambient audio, the internal audio or a combination of both;

activating a voice controlled interface when a manual activation of the interaction element is detected;

detecting a user's voice from measured ambient audio content;

extracting an audio signal of the user's voice; and

analyzing the audio signal to determine a voice command.

2. An earpiece according to claim **1**, further including the operations of:

analyzing the audio signal to determine a voice command; and

initiating a response in response to the voice command.

3. An earpiece according to claim **2**, where the voice command is to create an audio content wish list.

4. An earpiece, comprising:

a sealing section;

an ambient microphone;

a speaker;

an interaction element;

a location unit;

a memory that stores instructions; and

a logic circuit configured to execute the instructions to perform operations, the operations

comprising:

detecting the manual activation of the interaction element;

switching the mode of the earpiece to voice command mode when manual activation of the interaction element is detected;

detecting a user's voice;

extracting an audio signal of the user's voice;

analyzing the audio signal to determine a voice command; and

initiating a response in response to the voice command, where the voice command is perform at least one of the following actions with respect to an audio content list stored in the memory, purchase a song in the audio content list, delete a song from the audio content list, skip to the next song in the audio content list, add a song to the audio content list, and delete the audio content list.

5. An earpiece according to claim **2**, where the voice command is to search the internet.

6. An earpiece according to claim **5**, where the internet search results are received via a text to speech analyzer resulting in an audio result and the user receives the audio result.

7. An earpiece according to claim **2**, where the voice command is to play audio from the internet.

8. An earpiece according to claim **7**, where a second voice command is to scan audio from the internet.

9. An earpiece according to claim **2**, where the voice command is to play audio from a radio station.

10. An earpiece according to claim **2**, where the voice command is to search for a particular stock value.

11. An earpiece according to claim **2**, where the voice command is to link to a user's investment account.

12. An earpiece according to claim **11**, where a second voice command is to search for a particular stock value.

13. An earpiece according to claim **12**, where a third voice command is to buy the particular stock.

14. An earpiece according to claim **12**, where a third voice command is to sell the particular stock.

15. The earpiece according to claim **2** including an ear canal microphone.

16. An earpiece according to claim **15**, further including the operations of:

measuring the sound pressure level using the ear canal microphone.

17. An earpiece according to claim **15**, further including the operations of:

calculating the sound pressure level dosage of a user using the measured sound pressure level.

* * * * *

JS 44 (Rev. 09/19)

CIVIL COVER SHEET

The JS 44 civil cover sheet and the information contained herein neither replace nor supplement the filing and service of pleadings or other papers as required by law, except as provided by local rules of court. This form, approved by the Judicial Conference of the United States in September 1974, is required for the use of the Clerk of Court for the purpose of initiating the civil docket sheet. *(SEE INSTRUCTIONS ON NEXT PAGE OF THIS FORM.)*

I. (a) PLAINTIFFS

STATON TECHIYA, LLC

(b) County of Residence of First Listed Plaintiff _____
(EXCEPT IN U.S. PLAINTIFF CASES)

(c) Attorneys *(Firm Name, Address, and Telephone Number)*
Steven J. Balick, ASHBY & GEDDES
500 Delaware Avenue, 8th Floor, P.O. Box 1150
Wilmington, DE 19801, Telephone: (302) 654-1888

DEFENDANTS

HARMAN INTERNATIONAL INDUSTRIES, INCORPORATED and HARMAN-KARDON, INC.

County of Residence of First Listed Defendant _____
(IN U.S. PLAINTIFF CASES ONLY)

NOTE: IN LAND CONDEMNATION CASES, USE THE LOCATION OF THE TRACT OF LAND INVOLVED.

Attorneys *(If Known)*

II. BASIS OF JURISDICTION *(Place an "X" in One Box Only)*

- ❒ 1 U.S. Government Plaintiff
- ☒ 3 Federal Question *(U.S. Government Not a Party)*
- ❒ 2 U.S. Government Defendant
- ❒ 4 Diversity *(Indicate Citizenship of Parties in Item III)*

III. CITIZENSHIP OF PRINCIPAL PARTIES *(Place an "X" in One Box for Plaintiff and One Box for Defendant)*

(For Diversity Cases Only)

	PTF	DEF		PTF	DEF
Citizen of This State	❒ 1	❒ 1	Incorporated *or* Principal Place of Business In This State	❒ 4	❒ 4
Citizen of Another State	❒ 2	❒ 2	Incorporated *and* Principal Place of Business In Another State	❒ 5	❒ 5
Citizen or Subject of a Foreign Country	❒ 3	❒ 3	Foreign Nation	❒ 6	❒ 6

IV. NATURE OF SUIT *(Place an "X" in One Box Only)*

Click here for: Nature of Suit Code Descriptions.

CONTRACT
- ❒ 110 Insurance
- ❒ 120 Marine
- ❒ 130 Miller Act
- ❒ 140 Negotiable Instrument
- ❒ 150 Recovery of Overpayment & Enforcement of Judgment
- ❒ 151 Medicare Act
- ❒ 152 Recovery of Defaulted Student Loans (Excludes Veterans)
- ❒ 153 Recovery of Overpayment of Veteran's Benefits
- ❒ 160 Stockholders' Suits
- ❒ 190 Other Contract
- ❒ 195 Contract Product Liability
- ❒ 196 Franchise

REAL PROPERTY
- ❒ 210 Land Condemnation
- ❒ 220 Foreclosure
- ❒ 230 Rent Lease & Ejectment
- ❒ 240 Torts to Land
- ❒ 245 Tort Product Liability
- ❒ 290 All Other Real Property

TORTS

PERSONAL INJURY
- ❒ 310 Airplane
- ❒ 315 Airplane Product Liability
- ❒ 320 Assault, Libel & Slander
- ❒ 330 Federal Employers' Liability
- ❒ 340 Marine
- ❒ 345 Marine Product Liability
- ❒ 350 Motor Vehicle
- ❒ 355 Motor Vehicle Product Liability
- ❒ 360 Other Personal Injury
- ❒ 362 Personal Injury - Medical Malpractice

PERSONAL INJURY
- ❒ 365 Personal Injury - Product Liability
- ❒ 367 Health Care/ Pharmaceutical Personal Injury Product Liability
- ❒ 368 Asbestos Personal Injury Product Liability

PERSONAL PROPERTY
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- ❒ 371 Truth in Lending
- ❒ 380 Other Personal Property Damage
- ❒ 385 Property Damage Product Liability

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- ❒ 443 Housing/ Accommodations
- ❒ 445 Amer. w/Disabilities - Employment
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- ❒ 448 Education

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- ❒ 555 Prison Condition
- ❒ 560 Civil Detainee - Conditions of Confinement

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- ❒ 690 Other

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- ❒ 751 Family and Medical Leave Act
- ❒ 790 Other Labor Litigation
- ❒ 791 Employee Retirement Income Security Act

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- ❒ 422 Appeal 28 USC 158
- ❒ 423 Withdrawal 28 USC 157

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- ❒ 820 Copyrights
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- ❒ 835 Patent - Abbreviated New Drug Application
- ❒ 840 Trademark

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- ❒ 862 Black Lung (923)
- ❒ 863 DIWC/DIWW (405(g))
- ❒ 864 SSID Title XVI
- ❒ 865 RSI (405(g))

FEDERAL TAX SUITS
- ❒ 870 Taxes (U.S. Plaintiff or Defendant)
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- ❒ 376 Qui Tam (31 USC 3729(a))
- ❒ 400 State Reapportionment
- ❒ 410 Antitrust
- ❒ 430 Banks and Banking
- ❒ 450 Commerce
- ❒ 460 Deportation
- ❒ 470 Racketeer Influenced and Corrupt Organizations
- ❒ 480 Consumer Credit (15 USC 1681 or 1692)
- ❒ 485 Telephone Consumer Protection Act
- ❒ 490 Cable/Sat TV
- ❒ 850 Securities/Commodities/ Exchange
- ❒ 890 Other Statutory Actions
- ❒ 891 Agricultural Acts
- ❒ 893 Environmental Matters
- ❒ 895 Freedom of Information Act
- ❒ 896 Arbitration
- ❒ 899 Administrative Procedure Act/Review or Appeal of Agency Decision
- ❒ 950 Constitutionality of State Statutes

V. ORIGIN *(Place an "X" in One Box Only)*

☒ 1 Original Proceeding ❒ 2 Removed from State Court ❒ 3 Remanded from Appellate Court ❒ 4 Reinstated or Reopened ❒ 5 Transferred from Another District *(specify)* ❒ 6 Multidistrict Litigation - Transfer ❒ 8 Multidistrict Litigation - Direct File

VI. CAUSE OF ACTION

Cite the U.S. Civil Statute under which you are filing ***(Do not cite jurisdictional statutes unless diversity)***:
35 U.S.C. § 271

Brief description of cause:
Patent Infringement

VII. REQUESTED IN COMPLAINT:

❒ CHECK IF THIS IS A **CLASS ACTION** UNDER RULE 23, F.R.Cv.P.

DEMAND $

CHECK YES only if demanded in complaint:
JURY DEMAND: ☒ Yes ❒ No

VIII. RELATED CASE(S) IF ANY

(See instructions):

JUDGE _____ DOCKET NUMBER See attached

DATE 7/25/2023

SIGNATURE OF ATTORNEY OF RECORD /s/ Steven J. Balick

FOR OFFICE USE ONLY

RECEIPT # _____ AMOUNT _____ APPLYING IFP _____ JUDGE _____ MAG. JUDGE _____

CIVIL COVER SHEET ATTACHMENT

Related Cases

Case Name	Case Number	Court
Staton Techiya, LLC v. HARMAN International Industries, Incorporated	[TBD]	Delaware
Staton Techiya, LLC v. Samsung Electronics Co., Ltd. et al	23-0319-JRG-RSP	EDTX
Staton Techiya, LLC et al v. Samsung Electronics Co., Ltd. et al	21-0413-JRG-RSP	EDTX