UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SAMSUNG ELECTRONICS CO., LTD., and
SAMSUNG ELECTRONICS, AMERICA, INC.

Petitioners

v.

STATON TECHIYA, LLC,

Patent Owner

Case IPR2024-01003

U.S. Patent No. 11,710,473

DECLARATION OF CHRIS KYRIAKAKIS, PH.D. IN SUPPORT OF PETITION FOR INTER PARTES REVIEW OF U.S. PATENT NO. 11,710,473

Samsung v. Techiya
IPR2024-01003
Exhibit 1002

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A. GROUND 1A: *PATEL* IN COMBINATION WITH *TOPHOLM* RENDERS OBVIOUS CLAIMS 1 and 3-7 125

1. Claim 1 125

a. [1.pre]: A device that can control ambient passthrough in an earpiece comprising: 125

b. [1.1]: a first memory that stores instructions; and [1.2] a processor that executes the instructions to perform operations, the operations comprising 131

c. [1.3]: communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor; 136

d. [1.5]: sending an audio content signal to the earpiece, ... 143

e. [1.4]: sending an audio content gain value and an ambient sound gain value to the earpiece; and 145

(i) [1.4.1]: sending an audio content gain value … to the earpiece 145

(ii) [1.4.2]: sending … an ambient sound gain value to the earpiece 149

f. [1.6]: wherein the earpiece processor applies the audio content gain value to the audio content signal to generate a modified audio content signal, ... 156

g. [1.7]: wherein the earpiece processor applies the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal, and 157

h. [1.8]: wherein the earpiece processor mixes the modified ambient sound signal with the modified audio content signal to generate a mixed signal. 158

2. Claim 3 ("The device according to claim 1, wherein the earpiece processor generates the ambient sound signal from a first microphone signal from the first microphone"); Claim 5 ("The device according to claim 1, wherein the earpiece processor applies an mbient gain to a first microphone signal

from the first microphone to generate the ambient sound signal.") 161

3. Claim 4 ("The device according to claim 1 further including a user interface.") 166

4. Claim 6 ("The device according to claim 1, wherein the device is a portable communication device"); Claim 7 ("The device according to claim 6, where the device portable communication device is at least one of a phone, a watch, a tablet, earpiece, a lapto computer or a combination thereof.") 167

B. Ground 1B: *Patel*, *Topholm*, and *Williamson* Render Obvious Claim 2 170

1. Claim 2 170

a. [2.1] The device according to claim 1, wherein the earpiece processor calculates a noise level based upon a first microphone signal from the first microphone 170

b. [2.2] wherein the earpiece processor modifies the mixed signal based upon the noise level. 177

C. Ground 2A: *Melanson* and *Rosenberg* Render Obvious Claims 8, 19, and 20 179

1. Claim 8 179

a. [8.pre]: A device that can control ambient sound passthrough comprising: 179

b. [8.1]: a first microphone that measures a first acoustic environment to generate a first microphone signal; 181

c. [8.2]: a second microphone that is positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone generates a second microphone signal; 182

d. [8.3]: a speaker; 187

e. [8.4] and [8.5]: a memory that stores instructions; and a processor that executes the instructions to perform operations, the operations comprising: 188

f. [8.6]: receiving an audio content signal; 190

g. [8.7]: modifying the audio content signal to generate a modified audio content signal; 192

h. [8.8]: modifying the first microphone signal to generate a modified first microphone signal; [8.9]: modifying the second microphone signal to generate a modified second microphone signal; 193

i. [8.11]: detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof; 196

j. [8.12]: modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal; 198

k. [8.10]: generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof; 201

l. [8.13]: generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal; and [8.14] sending the mixed signal to the speaker 204

2. Claim 19: The device according to claim 8, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a user's voice is detected. 206

3. Claim 20: The device according to claim 8, wherein the acute sound is a voice. 212

D. Ground 2B: *Melanson*, *Rosenberg*, *Samadani*, and *Dijkstra* Render Obvious Claims 9, 13-18 212

1. Claim 9: The device according to claim 8, wherein the operations further comprise: determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics of at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof. 212

a. Two-step acute sound detection 213

b. "determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics" of one of the modified microphone signals 220

2. Claim 13: The device according to claim 9, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the mixed signal if the user's voice is detected. 226

3. Claim 14: The device according to claim 9, wherein the operations further comprise: detecting whether the acute sound is at least one of a warning sound, siren, voice, specific noise type, or a combination, by analyzing the spectrum of at least one of th modified first microphone signal, the modified second microphone signal, the modified ambient sound signal, or a combination. 229

4. Claim 15: The device according to claim 14, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a warning or siren is detected. 231

5. Claim 16: The device according to claim 14, where the operations further comprise: sending a notification signal to the speaker if the acute sound is a warning sound or siren. 232

6. Claim 17: The device according to claim 16, wherein the warning sound is at least one of a bell, an emergency vehicle, the sound of a security system or a combination thereof. 234

7. Claim 18: The device according to claim 14, wherein if an alarm is detected it is at least one of a police siren, an ambulance siren, a car honking or a combination thereof. ...235

E. Ground 2C: *Melanson*, *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* Render Obvious Claims 10-12 236

1. Claim 10: The device according to claim 9, wherein the operations further comprise: generating a noise signal from at least one of the modified first microphone signal, the modified second microphone signal, or both; generating a noise reduction signal. 236

2. Claim 11: The device according to claim 10, wherein the operations further comprise: mixing the noise reduction signal

with the mixed signal to generate a new mixed signal, and sending the new mixed signal to the speaker instead of the mixed signal.....246

3. Claim 12: The device according to claim 11, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the new mixed signal if the user's voice is detected. 247

XI. SECONDARY CONSIDERATIONS.....247

XII. CONCLUSION 248

I, Chris Kyriakakis declare that:

I. ASSIGNMENT

1. I have been retained on behalf of Samsung Electronics Co., Ltd. to offer technical opinions related to U.S. Patent No. 11,710,473 ("the '473 patent") (EX1001). I understand that Samsung is requesting that the Patent Trial and Appeal Board ("PTAB" or "Board") institute an *inter partes* review ("IPR") proceeding of the '473 patent.

2. I have been asked to provide my independent analysis of the '473 patent based on the prior art publications cited in this declaration.

3. I am not and never have been, an employee of Samsung. I received no compensation for this declaration beyond my normal hourly compensation based on my time actually spent analyzing the '473 patent, the prior art publications cited below, and issues related thereto, and I will not receive any added compensation based on the outcome of any IPR or other proceeding involving the '473 patent.

II. QUALIFICATIONS

4. I am an independent consultant. All of my opinions stated in this declaration are based on my own personal knowledge and professional judgment. In forming my opinions, I have relied on my education, experience, and knowledge regarding electrical engineering, computer science, and audio and acoustic sciences.

5. I am over 18 years of age and, if I am called upon to do so, I would be competent to testify as to the matters set forth herein. A copy of my current curriculum vitae, which details my education and professional and academic experience, as well as a list of all publications I have authored in the past ten years, is included as EX1003 in this proceeding. The following provides an overview of some of my experience that is relevant to the matters set forth in this declaration.

6. I earned my Bachelor of Science degree in Engineering and Applied Science from the California Institute of Technology (Caltech) in 1985. I received my Master of Science degree in Electrical Engineering in 1987 and my Ph.D. in Electrical Engineering in 1993, both from the University of Southern California ("USC").

7. Since 2002, I am a tenured Associate Professor in the Electrical Engineering Department at USC. My research interests lie at the intersection of acoustics, psychoacoustics (the science that studies human perception of sound), and audio signal processing. My recent research has focused on the study of audio systems in challenging environments including automobiles and mobile devices, as well as algorithms for enhancing the performance of voice recognition engines. I have published nearly 100 technical papers, including several peer reviewed papers. I have published a book entitled *Immersive Audio*

Signal Processing, and hold several patents in acoustic measurement of loudspeakers in rooms and cars, loudspeaker crossover optimization, and loudspeaker response correction using signal processing. My publications examine various aspects of sound measurement, how sound interacts with the acoustical elements of the environment, novel methods for surround sound recording and reproduction, and the perception of sound by human listeners.

8. A focus of my research has been the study of microphone arrays for applications in enhanced far field voice recognition. This includes methods for Direction of Arrival finding and beam forming from linear and circular microphone arrays. I have also investigated methods for noise removal and reverberation reduction from microphone signals. In addition, over the past year, I have worked on methods for echo cancellation for applications in wearable devices. Previous work includes loudspeakers and how they interact with their acoustical environment.

9. In particular, I have researched the role of sound reflections, absorption, and diffusion in the performance of loudspeakers. I have also published several technical papers on acoustical measurement methods in rooms and device enclosures and developed novel signal processing algorithms for optimizing sound system performance. Other topics I have researched include multichannel audio acquisition and rendering, virtual microphones and virtual

speakers, hybrid headphone- loudspeaker rendering methods, and advanced signal processing techniques for optimizing sound quality from small portable devices.

10. I am the founding Director of the USC Immersive Audio Laboratory with facilities for experimental work in room acoustics, multichannel audio, and psychoacoustics. This laboratory also serves as a unique teaching facility for my undergraduate course in Introduction to Digital Audio and my graduate course in Immersive Audio Signal Processing. Both courses have a major acoustics component that examines the interaction of sound with the acoustical environment (e.g., home, movie theater, car). The graduate course was developed through a two- year grant I received from the National Science Foundation entitled "Collaborative Learning in Engineering Using Immersive Environments," and was the first of its kind to assess the impact of audio immersion in student learning. In addition to the courses I teach, I have also supervised and served on Ph.D. dissertation committees for more than 30 students.

11. In 2019, I joined Syng as the Chief Audio Scientist. Syng creates innovative products for next generation spatial audio and my role there includes research and development of audio algorithms for sound capture using

microphone arrays to measure the acoustics of the room and loudspeaker beamforming for rendering sound in various directions.

12. From 2003 to 2018, I was the co-founder and Chief Technology Officer of Audyssey Laboratories, a USC spin-off company that develops and licenses audio technology to leading automotive, professional and consumer electronic companies around the world. A major project that I led and worked on was focused on the measurement and characterization of in-ear and over ear earbuds and headphones. This work included the development of methods for measuring earbuds and headphones using a mannequin with artificial ears and, in particular, methods for testing for proper seal prior to taking acoustical measurements. It also included the use of ear canal microphones and comparisons of the in-ear microphone response with a reference microphone in the room. A database of over 500 in-ear earbuds and headphones was created using these methods. The acoustical correction filters that were generated for each headphone were implemented as an embedded solution that could run on DSP processors in mobile devices and laptop computers.

13. I also developed audio algorithms and designed speakers. These speakers were novel acoustical designs that used a combination of unique enclosures, and audio signal processing to optimize their performance and overcome limitations that arise from small drivers and enclosures. For example,

we used signal processing technologies combined with novel acoustical design to extend the bass response of small woofers and passive radiators beyond what was previously possible in small speaker enclosures. The innovations in these designs have received awards, including Popular Science's "Best of What's New."

14. In April 2018 I was elected as Senior Member of the Institute of Electrical and Electronic Engineers (IEEE) in recognition of my contributions to the field of engineering. I am also a member of the Audio Engineering Society (AES), an association for professionals in the audio industry.

15. In 2006, I received a World Technology Network Award. This organization presents awards to innovators in several areas in which technology can foster a paradigm change. My award was for innovations in immersive audio that enable new capabilities in media and journalism. Other award recipients at that event included Vice President Al Gore, Google, and Space-X. In addition, my work has been featured in multiple news articles, including pieces by The Atlantic, the BBC World Service, the New York Times, and National Public Radio.

16. In the late 1990s and early 2000s, I was a faculty researcher and later Deputy Director of the National Science Foundation's engineering research center established at USC. I was studying the fundamental and technological

limitations of immersive audio and the role of acoustics on the performance of loudspeakers and audio systems in homes and cars. In 2003, together with one of my graduate students, I received the award for Best Paper at the Institute of Electrical and Electronics Engineers ("IEEE") Conference on Signals, Systems and Computers.

III. SUMMARY OF CONCLUSIONS FORMED

17. This Declaration explains the conclusions that I have formed based on my analysis. To summarize those conclusions:

- Ground 1A: *Patel* and *Topholm* render obvious claims 1, and 3-7
- Ground 1B: *Patel*, *Topholm*, and *Williamson* render obvious claim 2
- Ground 2A: *Melanson* and *Rosenberg* render obvious claims 8, and 19-20
- Ground 2B: *Melanson*, *Rosenberg*, *Samadani*, and *Dijkstra* render obvious claims 9, 13-18
- Ground 2C: *Melanson*, *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* render obvious claims 10-12

IV. LEVEL OF ORDINARY SKILL IN THE ART

18. I have been informed by Petitioners' counsel that a person having ordinary skill in the art is a hypothetical person who is presumed to have the skill and experience of an ordinary worker in the field at the time of the priority date of the challenged patent. I have been asked to assume for the purposes of this Petition that the priority date is January 22, 2007. I have filed a separate declaration in connection with IPR2024-01004 in which I was asked to provide

my opinion that the Challenged Claims are not entitled to a priority date before February 3, 2022.

19. Based on my knowledge and experience in the field and my review of the '473 patent and file history, I believe that a person having ordinary skill in the art in this matter would have had at least a Bachelor's of Science Degree (or equivalent) in electrical engineering, computer science, or a similar field and two years of experience in the design of digital audio systems and associated signal processing such as voice or speech processing and/or acoustic pattern recognition. A greater amount of education could compensate for fewer years of work experience, and vice versa. This is the appropriate level of skill whether the claims are entitled to priority as of January 22, 2007 or February 3, 2022.

20. Based on my experiences, I have at least the skills of a person having ordinary skill in the art as well as a good understanding of the capabilities of a person having ordinary skill in the art. To be clear, all of my opinions in this declaration are from the perspective of one of ordinary skill in the art as I have defined it here during the relevant timeframe.

V. LEGAL PRINCIPLES

21. I am not an attorney and offer no legal opinions, but in the course of my work, I have had experience studying and analyzing patents and patent claims from the perspective of a person skilled in the art.

22. For the purposes of this declaration, I have been informed about certain aspects of the law that are relevant to forming my opinions. My understanding of the law is as follows.

23. Petitioners' counsel has informed me that a published nonprovisional application is entitled to claim priority to the filing date of its provisional application (as opposed to the filing date of the published non-provisional application) if (1) the provisional application discloses the portions of the published non-provisional application relied upon as prior art, and (2) at least one claim of the published non-provisional claim is supported by the provisional application, i.e., the provisional application must disclose the subject matter of the claim and enable a person of ordinary skill in the art to make and use the claim.

24. Petitioners' counsel has informed me that for the prior art to inherently disclose a claimed limitation, the prior art need not expressly disclose the limitation, so long as the claimed limitation necessarily flows from a disclosure in the prior art. I also understand that it is acceptable to examine evidence outside the prior art reference (extrinsic evidence) in determining whether a feature, while not expressly discussed in the reference, is necessarily present in that reference.

25. Petitioners' counsel has informed me that a patent claim can be considered to have been obvious to a person having ordinary skill in the art at the time the application was filed. I am informed that this means that, even if all of the requirements of a claim are not found in a single prior art reference, the claim is not patentable if the differences between the subject matter in the prior art and the subject matter in the claim would have been obvious to a person having ordinary skill in the art at the time of the invention.

26. I have been informed by Petitioners' counsel that a determination of whether a claim would have been obvious should be based upon several factors, including, among others:

- the level of ordinary skill in the art at the time the application was filed;
- the scope and content of the prior art; and
- what differences, if any, existed between the claimed invention and the prior art.

27. I have been informed by Petitioners' counsel that a single reference can render a patent claim obvious if any differences between that reference and the claims would have been obvious to a person having ordinary skill in the art. Alternatively, I understand that the teachings of two or more references may be combined in the same way as disclosed in the claims, if such a combination would have been obvious to one having ordinary skill in the art. In determining

whether a combination based on either a single reference or multiple references would have been obvious, it is appropriate to consider, among other factors:

- whether the teachings of the prior art references disclose known concepts combined in familiar ways, and when combined, would yield predictable results;
- whether a person having ordinary skill in the art could implement a predictable variation, and would see the benefit of doing so;
- whether the claimed elements represent one of a limited number of known design choices, and would have a reasonable expectation of success by those skilled in the art;
- whether a person having ordinary skill would have recognized a reason to combine known elements in the manner described in the claim;
- whether there is some teaching or suggestion in the prior art to make the modification or combination of elements claimed in the patent; and
- whether the innovation applies a known technique that had been used to improve a similar device or method in a similar way.

28. I have been informed by Petitioners' counsel that one of ordinary skill in the art has ordinary creativity and is not an automaton. Petitioners' counsel has also informed me that in considering obviousness, obviousness may not be determined using the benefit of hindsight, including hindsight derived from the patent being considered.

29. I further understand that a hypothetical person of ordinary skill in the art is assumed to know and to have all relevant prior art in the field of endeavor covered by the patent at issue. I have been informed that there are two criteria for determining whether prior art is analogous and thus can be

considered prior art: (1) whether the art is from the same field of endeavor, regardless of the problem addressed, and (2) if the reference is not within the field of the patentee's endeavor, whether the reference still is reasonably pertinent to the particular problem with which the patentee is involved. I have also been informed that the field of endeavor of a patent is not limited to the specific point of novelty, the narrowest possible conception of the field, or the particular focus within a given field. I have also been informed that a reference is reasonably pertinent if, even though it may be in a different field from that of the patentee's endeavor, it is one which, because of the matter with which it deals, logically would have commended itself to a patentee's attention in considering their problem.

30. I further understand that, in making a determination as to whether the claimed invention would have been obvious to a person of ordinary skill, the Board may consider certain objective factors if they are present, such as: commercial success of products practicing the claimed invention; long-felt but unsolved need; teaching away; unexpected results; copying; and praise by others in the field. These factors are generally referred to as "secondary considerations" or "objective indicia" of nonobviousness. I understand, however, that for such objective evidence to be relevant to the obviousness of a claim, there must be a causal relationship (called a "nexus") between the claim and the evidence and

that this nexus must be based on a novel element of the claim rather than something in the prior art. I also understand that even when they are present, secondary considerations may be unable to overcome primary evidence of obviousness (such as motivation to combine with predictable results) that is sufficiently strong.

31. I have been informed by Petitioners' counsel that for a patent claim in a later-filed application to be entitled to the priority date of an earlier-filed application, the specification of the earlier-filed application must adequately support the patent claim of the later-filed application. This means that the specification of the earlier-filed application must provide an adequate written description of the invention claimed in the later-filed application.

32. I have been informed by Petitioners' counsel that the written description requirement is met if the applicant conveys in the specification with reasonable clarity to those skilled in the art that he or she possessed the claimed invention as of the filing date of the application. An applicant shows that one is "in possession" of the invention by describing the invention, with all its claimed limitations, not merely that which makes it obvious. A description that merely renders the later-filed claim obvious is not sufficient. One may satisfy the written description requirement with words, structures, figures, diagrams, and/or formulas that fully set forth the claimed invention. Although the exact terms of

the claim need not be recited verbatim in the specification, the specification must contain an equivalent description of the claimed subject matter.

33. I am also informed that the failure to include in the written description aspects of the claimed invention that would be routinely known and understood by one of ordinary skill in the art when the application was filed will not be fatal to the validity of the claim if such aspects are inherent to what was disclosed. To be inherent, the missing descriptive matter must necessarily be present in the specification such that one skilled in the art would recognize such a disclosure.

VI. MATERIALS CONSIDERED

34. My analysis and conclusions set forth in this declaration are based on my educational background and experiences in the field (see Section II). Based on my knowledge and experience, I believe that I am considered to be an expert in the field. Also, based on my knowledge and experience, I understand and know of the capabilities of persons of ordinary skill in the field since the early 2000s, and I taught, participated in organizations, and worked closely with many such persons in the field during that time frame.

35. As part of my independent analysis for this declaration, I have considered the following: the background knowledge/technologies that were commonly known to persons of ordinary skill in this art during the time before

the earliest claimed priority date for the ’473 patent; my own knowledge and experiences gained from my work experience in the field of the ’473 patent and related disciplines; and my experience in working with others involved in this field and related disciplines.

36. In addition, I have analyzed the following publications and materials:

- U.S. Patent 11,710,473 (EX1001)
- File History of U.S. Patent 11,710,473 (EX1004)
- U.S. Patent 11,244,666 (“’666 patent”) (EX1007)
- File History of U.S. Patent 11,244,666 (EX1008)
- File History of U.S. Appl. No. 90/019,169, Samsung Electronics Co., Ltd. and Samsung Electronics, America, Inc., Request for *Ex Parte Reexamination* of U.S. Pat. No. 11,244,666, Date: Feb. 24, 2023 (EX1009)
- U.S. Publication 2007/0041589A1 (“*Patel*”) (EX1014)
- Publication WO2006/097099A1 (“*Topholm*”) (EX1015)
- U.S. Patent 5,027,410 (“*Williamson*”) (EX1016)
- U.S. Patent 7,903,825 (“*Melanson*”) (EX1017)
- U.S. Publication 2007/0189544A1 (“*Rosenberg*”) (EX1018)

- U.S. Provisional Application Number 60/841,990 ("'990PROV") (EX1019)
- U.S. Publication 2006/0182295A1 ("*Dijkstra*") (EX1020)
- U.S. Publication 2008/0079571A1 ("*Samadani*") (EX1021)
- Publication WO2006/036262A1 ("*Engle*") (EX1022)
- U.S. Patent 4,455,677 ("*Fox*") (EX1023)
- U.S. Patent 4,827,458 ("*d'Arc*") (EX1024)
- EP0495653A ("*Smiedt*") (EX1025)
- U.S. Patent 5,647,011 ("*Garvis*") (EX1026)
- U.S. Patent 6,782,106 ("*Kong*") (EX1027)
- Sumit Basu et al., *Smart Headphones: Enhancing Auditory Awareness Through Robust Speech Detection and Source Localization*, 5 IEEE INT'L CONF. ON ACOUSTICS SPEECH & SIGNAL PROCESSING PROC. 3361 (2001) (EX1028)
- Florian Mueller & Matthew Karau, *Transparent Hearing*, CHI '02 EXTENDED ABSTRACTS ON HUM. FACTORS IN COMPUTING SYS. 730 (2002) (EX1029)
- U.S. Publication 2001/0046304A1 ("*Rast*") (EX1030)
- U.S. Patent 4,947,432 ("*Topholm432*") (EX1031)
- U.S. Patent 5,202,927 ("*Topholm927*") (EX1032)

- E.P. Publication 1519625A2 ("*Victorian*") (EX1033)
- Excerpt from COMPUTER DICTIONARY (2d ed. 1994) ("*Computer Dictionary*") (EX1034)
- Excerpt from ANDREW S. TANENBAUM, STRUCTURED COMPUTER ORGANIZATION (2006) ("*Tanenbaum*") (EX1035)
- Excerpts from ROBERT OSHANA, DSP SOFTWARE DEVELOPMENT TECHNIQUES FOR EMBEDDED AND REAL-TIME SYSTEMS (2006) ("*Oshana*") (EX1036)
- U.S. Patent 5,748,754 ("*Maag*") (EX1037)
- Excerpt from GLEN BALLOU, HANDBOOK FOR SOUND ENGINEERS: THE NEW AUDIO CYCLOPEDIA (Glen Ballou ed., 1987) ("*AudioCyclopedia*") (EX1038)
- U.S. Publication 2007/0201705 ("*Dorogusker*") (EX1039)
- WIPO Publication WO2008067454A2 ("*Bongiovi*") (EX1040)
- U.S. Patent 4,891,841 ("*Bohn*") (EX1041)
- U.S. Publication 2008/0129520A1 ("*Lee*") (EX1042)
- U.S. Publication 2007/0021148A1 ("*Mahini*") (EX1043)
- U.S. Publication 2008/0165988A1 ("*Terlizzi*") (EX1044)
- Excerpt from File History of U.S. Appl. No. 17/483,190 (EX1045)
- U.S. Publication 2008/0089530A1 ("*Bostick*") (EX1046)

- U.S. Publication 2004/0179694A1 ("*Alley*") (EX1047)
- U.S. Publication 2006/0262944A1 ("*Rasmussen*") (EX1048)
- U.S. Publication 2003/0165246A1 ("*Kvaløy*") (EX1049)
- U.S. Publication 2007/0177743A1 ("*Mertens*") (EX1050)
- U.S. Patent 7,903,826 ("*Boersma*") (EX1051)

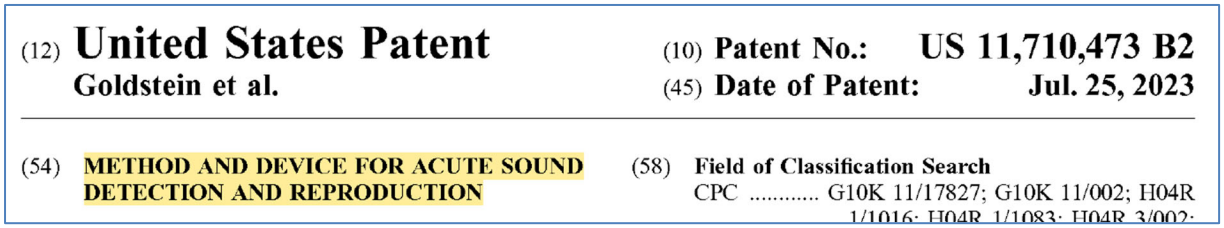
and any other materials I refer to in this declaration in support of my opinions.

37. My analysis and conclusions set forth in this declaration are based on the perspective of a person having ordinary skill in the art.

VII. OVERVIEW OF THE '473 PATENT

A. THE SPECIFICATION

38. The '473 patent is titled "Method and Device for Acute Sound Detection and Reproduction."



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

(10) **Patent No.: US 11,710,473 B2**
(45) **Date of Patent: Jul. 25, 2023**

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

(58) **Field of Classification Search**
CPC G10K 11/17827; G10K 11/002; H04R 1/1016; H04R 1/1083; H04R 3/002;

EX1001, Cover (annotated)[1]

39. The '473 patent explains that "[o]n a daily basis, people are exposed to various environmental sounds and noises within their environment, such as the sounds from traffic, construction, and industry." *'473Pat.*, 1:55-58.

[1] All emphases and annotations throughout are added unless otherwise noted.

"To combat the undesired cacophony of annoying sounds, people are arming themselves with portable audio playback devices to drown out intrusive noise." *Id.*, 1:59-61. "With such devices, the person is immersed in the audio experience and generally less likely to hear warning sounds within their environment. In some cases, the user may even turn up the volume to hear their personal audio over environmental noises. It also puts them at high sound exposure risk which can potentially cause long term hearing damage." *Id*., 2:3-9. "A need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties." *Id*., 2:23-25.

40. The alleged invention of the '473 patent "provide[s] a method and device for acute sound detection and reproduction." *Id*., 2:29-31. "At least one exemplary embodiment of the invention is directed to an earpiece for ambient sound monitoring and warning detection." *Id*., 4:56-58. With reference to Figure 1, copied below, "[e]arpiece 100 includes an Ambient Sound Microphone (ASM) Ill [*sic*] 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." *Id*., 5:3-7.

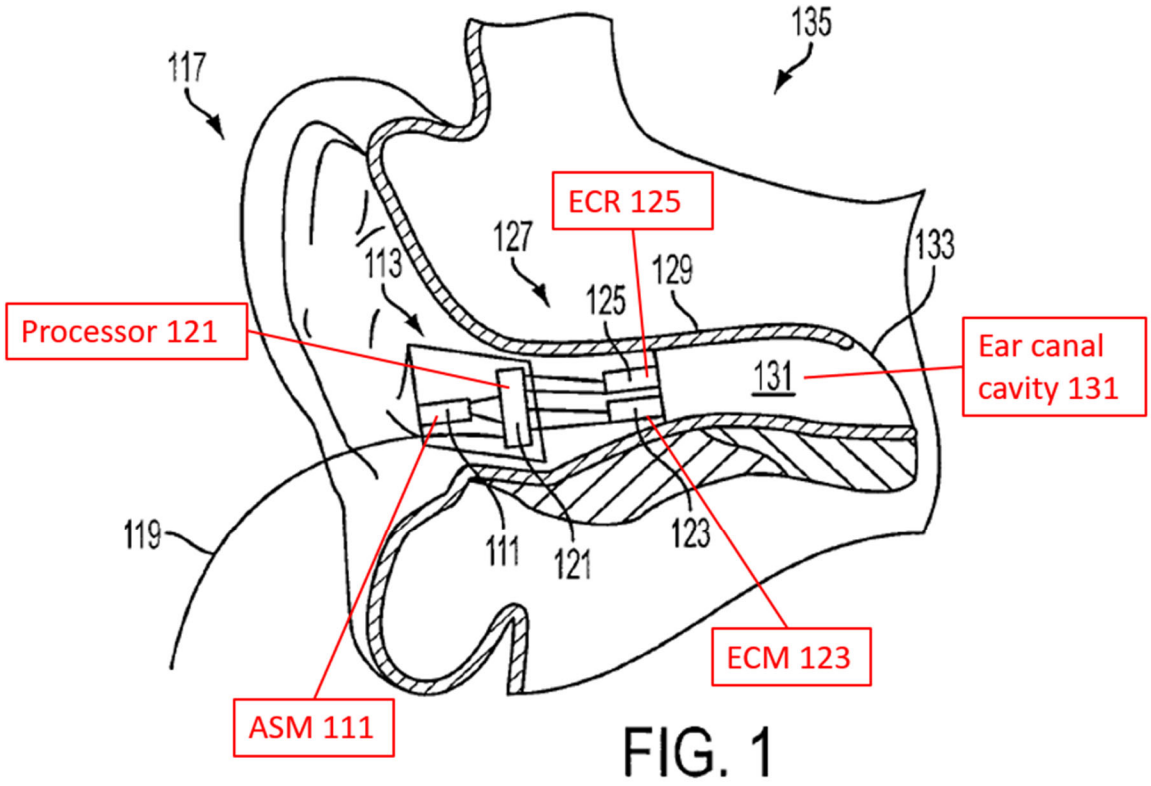
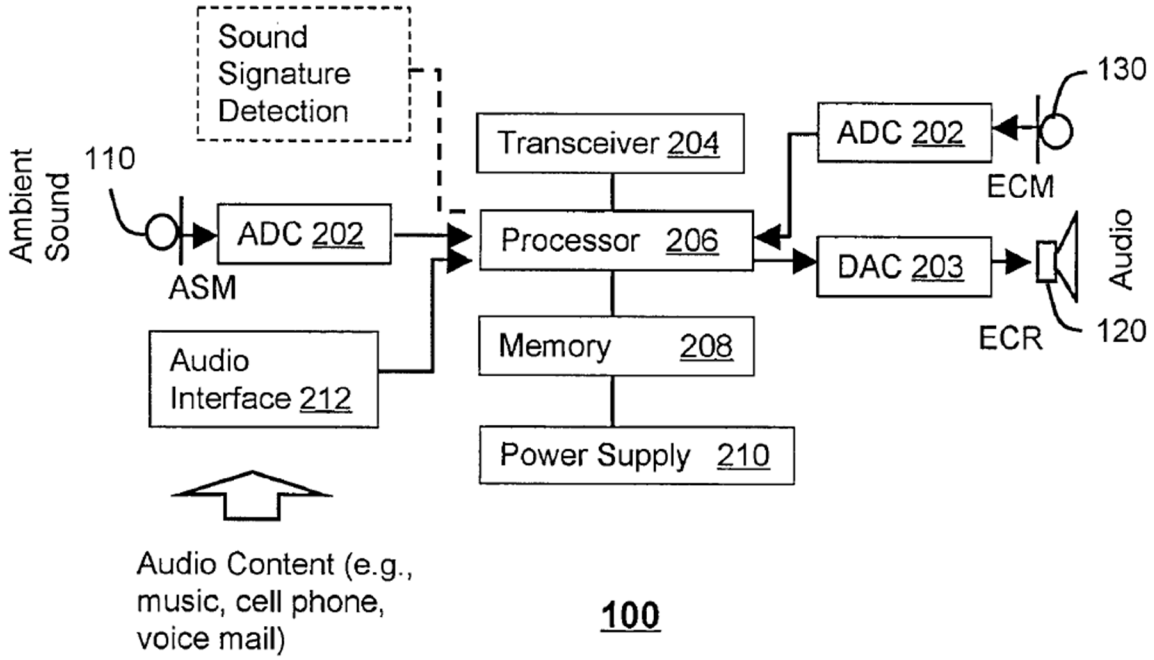


FIG. 1

'473pat., FIG. 1 (annotated)

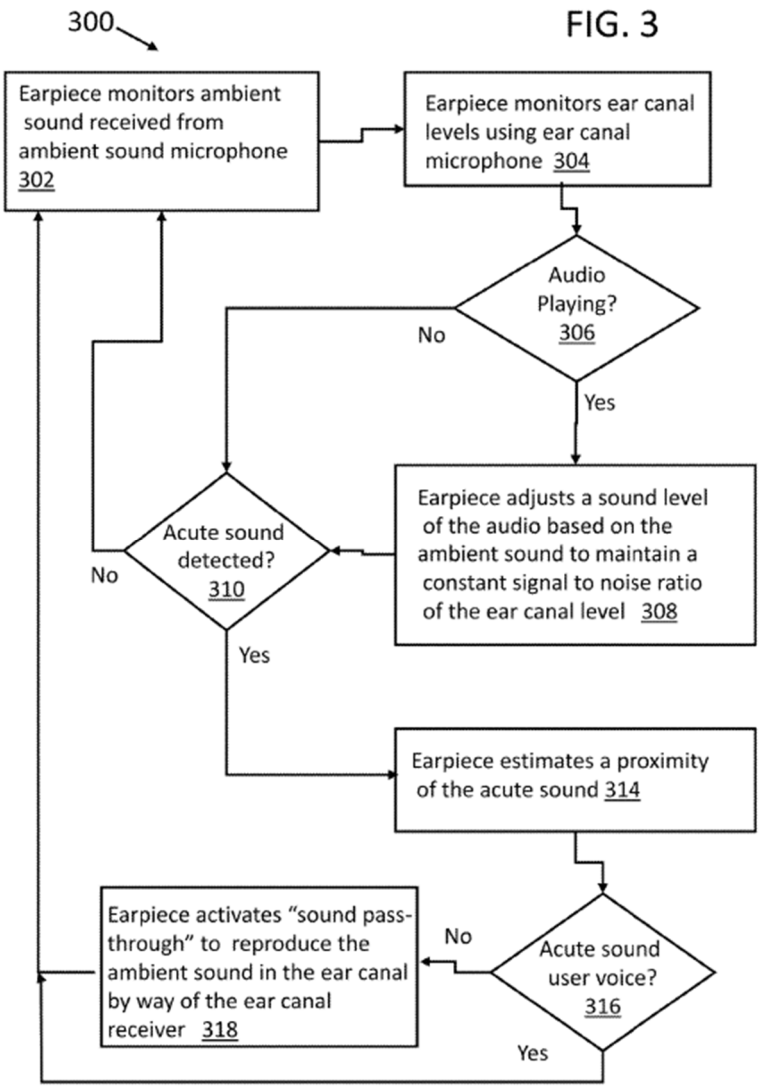
41. With reference to Figure 2, copied below, "the earpiece 100 can include a processor 206 operatively coupled to the ASM 111, ECR 125, and ECM 123 via one or more Analog to Digital Converters (ADC) 202 and Digital to Analog Converters (DAC) 203." *Id.*, 5:38-43. "The processor 206 can monitor the ambient sound captured by the ASM 111 for acute sounds in the environment, such as an abrupt high energy sound corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., "help", "stop", "police", etc.), or specific noise type (e.g., breaking glass, gunshot, etc.)." *Id.*, 5:43-50.



'473pat., FIG. 2

42. "The earpiece 100 can include an audio interface 212 operatively coupled to the processor 206 to receive audio content, for example from a media player or cell phone, and deliver the audio content to the processor 206. The processor 206 responsive to detecting acute sounds can adjust the audio content and pass the acute sounds directly to the ear canal." *Id.*, 5:59-65.

43. Figure 3 provides "a flowchart of a method 300 for acute sound detection and reproduction in accordance with an exemplary embodiment." *Id.*, 6:30-32. "The method 300 can start in a state wherein the earpiece 100 has been inserted and powered on. As shown in step 302, the earpiece 100 can monitor the environment for ambient sounds received at the ASM 111." *Id.*, 6:40-43.



'473pat., FIG. 3

44. "If at step 306, audio is playing (e.g., music, cell phone, etc.), the earpiece 100 adjusts a sound level of the audio based on the ambient sound to maintain a constant signal to noise ratio with respect to the ear canal level at step 308. For instance, the processor 206 can selectively amplify or attenuate audio content received from the audio interface 212 before it is delivered to the ECR 125. The processor 206 estimates a background noise level from the ambient

sound received at the ASM 111, and adjusts the audio level of delivered audio content (e.g., music, cell phone audio) to maintain a constant signal (e.g., audio content) to noise level (e.g., ambient sound)." *Id*., 7:10-21.

45. "If at step 310, an acute sound is detected within the ambient sound, the earpiece 100 activates 'sound pass-through' to reproduce the ambient sound in the ear canal by way of the ECR 125." *Id*., 7:28-31. "The processor 206 permits the ambient sound to pass through the ECR 125 to the ear canal 131 directly for example by replicating the ambient sound external to the ear canal within the ear canal." *Id*., 7:31-34.

46. "In one embodiment, the earpiece 100 looks for temporal and spectral characteristics in the ambient sound for detecting acute sounds. For instance, as will be explained ahead, the processor 206 looks for an abrupt change in the Sound Pressure Level (SPL) of an ambient sound across a small time period. The processor 206 can also detect abrupt magnitude changes across frequency sub-bands (e.g. filter-bank, FFT, etc.). Notably, the processor 206 can search for on-sets (e.g., fast rising amplitude wave-front) of an acute sound or other abrupt feature characteristics without initially attempting to initially identify or recognize the sound source. That is, the processor 206 is actively listening for a presence of acute sounds before identifying the type of sound source." *Id*., 7:48-61.

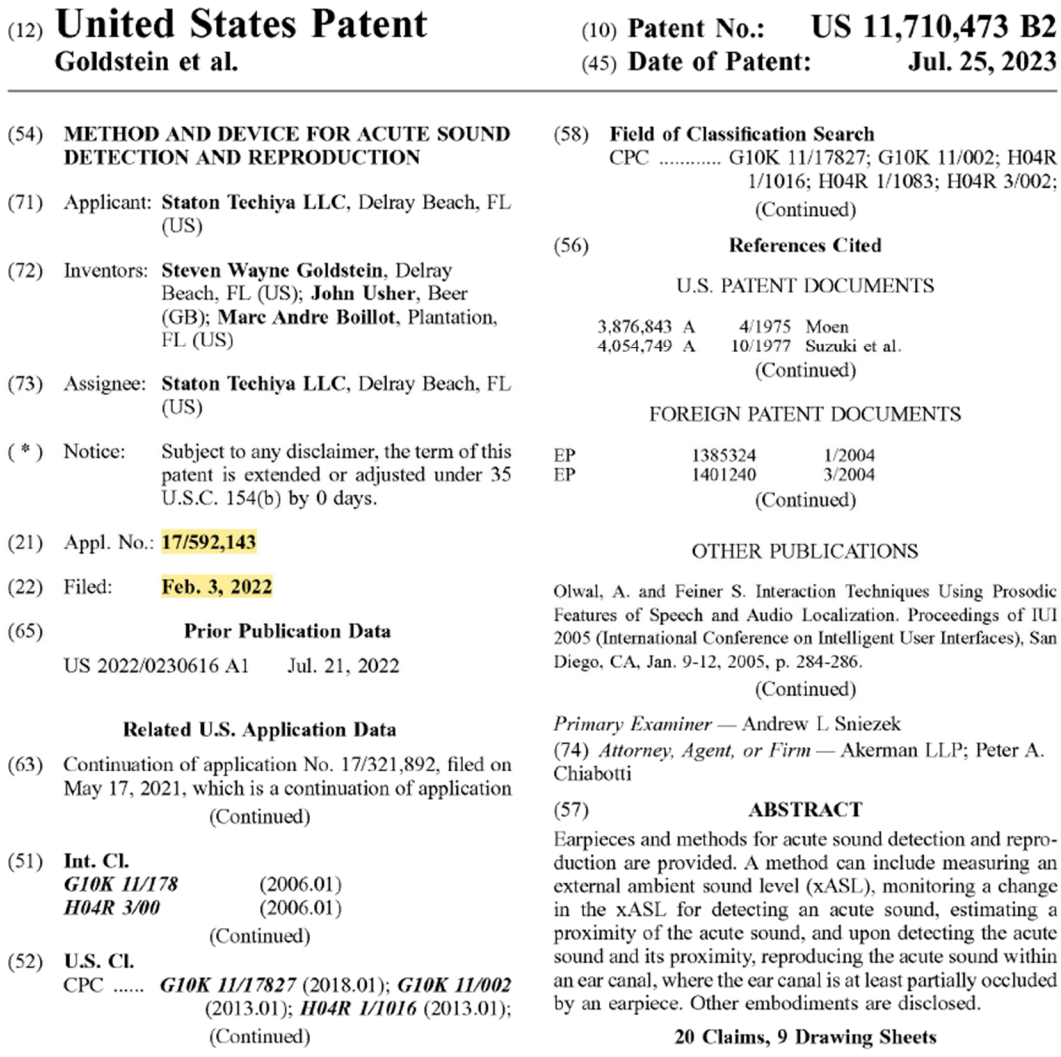
47. "At step 316, the earpiece 100 determines whether it is the user's voice that generates the acute sound when the user speaks, or whether it is an external sound such as a vehicle approaching the user. If at step 316, the processor 206 determines that the acute sound is a result of the user speaking, the processor 206 does not activate a pass-through mode, since this is not considered an external warning sound. The pass-through mode permits ambient sound detected at the ASM 111 to be transmitted directly to the ear canal. If however, the acute sound corresponds to an external sound source, such as an on-set of a warning sound, the earpiece at step 318 activates 'sound pass-through' to reproduce the ambient sound in the ear canal by way of the ECR 125." *Id*., 8:44-57.

48. "The earpiece 100 can also present an audible notification to the user indicating that an external sound source generating the acute sound has been detected. The method 300 can proceed back to step 302 to continually monitor for acute sounds in the environment." *Id*., 8:57-61.

B. PROSECUTION HISTORY

49. The '473 patent issued from U.S. Application No. 17/592,143 ("'143 application"), filed on February 3, 2022. *'473Pat.*, Cover. The '473 patent claims priority through a chain of non-provisional applications to a provisional application filed on January 22, 2007. *'473Pat.*, Cover. The '473

patent's parent application is U.S. 17/321,892 (the "'892 application" or "'892App."). *Id*.



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

(10) **Patent No.: US 11,710,473 B2**
(45) **Date of Patent: Jul. 25, 2023**

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

(71) Applicant: **Staton Techiya LLC**, Delray Beach, FL (US)

(72) Inventors: **Steven Wayne Goldstein**, Delray Beach, FL (US); **John Usher**, Beer (GB); **Marc Andre Boillot**, Plantation, FL (US)

(73) Assignee: **Staton Techiya LLC**, Delray Beach, FL (US)

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

(21) Appl. No.: **17/592,143**

(22) Filed: **Feb. 3, 2022**

(65) **Prior Publication Data**

US 2022/0230616 A1 Jul. 21, 2022

Related U.S. Application Data

(63) Continuation of application No. 17/321,892, filed on May 17, 2021, which is a continuation of application (Continued)

(51) **Int. Cl.**
G10K 11/178 (2006.01)
H04R 3/00 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***G10K 11/17827*** (2018.01); ***G10K 11/002*** (2013.01); ***H04R 1/1016*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC G10K 11/17827; G10K 11/002; H04R 1/1016; H04R 1/1083; H04R 3/002;
(Continued)

(56) **References Cited**

U.S. PATENT DOCUMENTS

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4,054,749 A 10/1977 Suzuki et al.
(Continued)

FOREIGN PATENT DOCUMENTS

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(Continued)

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Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan. 9-12, 2005, p. 284-286.
(Continued)

Primary Examiner — Andrew L Sniezek
(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti

(57) **ABSTRACT**

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

20 Claims, 9 Drawing Sheets

EX1001, Cover (annotated)

50. The snip below shows the text of claim 1 as originally filed in the '143 application.

1. A wearable device that can control ambient passthrough in an earpiece comprising:

a display;

a first memory that stores instructions; and

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

communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;

sending a mixing value to the earpiece; and

sending an audio content signal to the earpiece, wherein the earpiece processor is configured to mix an ambient sound signal with the audio content signal based upon the mixing value.

EX1004 at 48

51. The snip below shows the text of claim 6 as originally filed in the '143 application. After various amendments, claim 6 issued as independent claim 8.

6. A wearable device that can control ambient sound passthrough comprising:

a first microphone configured to measure a first acoustic environment to generate a first microphone signal;

a second microphone configured to be positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone is configured to generate a second microphone signal;

a speaker;

a memory that stores instructions; and

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

receiving an audio content signal;

modifying the audio content signal to generate a modified audio content signal;

modifying the first microphone signal to generate a modified first microphone signal;

modifying the second microphone signal to generate a modified second microphone signal;

generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination of both;

detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or both, and if an acute sound is detected, determining whether the acute sound is a user's voice;

generating a mixed signal by mixing the ambient sound signal with the modified audio content signal, and if a user's voice is detected modifying the modified audio content signal before forming the mixed signal; and

sending the mixed signal to the speaker.

EX1004 at 48-49

52. On December 22, 2022, the Patent Office mailed a non-final rejection of claims 1-17.

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/592,143	02/03/2022	Steven Wayne Goldstein	CONT-17321892	3471

150139 7590 12/22/2022

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

EXAMINER	
SNIEZEK, ANDREW L	

ART UNIT	PAPER NUMBER
2688	

NOTIFICATION DATE	DELIVERY MODE
12/22/2022	ELECTRONIC

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

EX1004 at 786 (annotated)

Office Action Summary	Application No. 17/592,143	Applicant(s) Goldstein et al.	
	Examiner ANDREW SNIEZEK	Art Unit 2688	AIA (FITF) Status No

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

Period for Reply

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

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

Status

1)☑ Responsive to communication(s) filed on 2/3/22.

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

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

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

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

Disposition of Claims*

5) ☑ Claim(s) 1-17 is/are pending in the application.

5a) Of the above claim(s) ____ is/are withdrawn from consideration.

6) ☐ Claim(s) ____ is/are allowed.

7) ☑ Claim(s) 1-17 is/are rejected.

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

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

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

EX1004 at 787 (annotated)

53. The Patent Office rejected claims 1-17 "as failing to comply with the enablement requirement." With respect to claim 1, the Patent Office stated:

> The claim(s) contains subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. Independent claim 1 sets forth a "mixing value" that is sent to the earpiece, and configured to mix based upon a "mixing value". The written disclosure as filed does not even mention a "mixing value", what structure is used to

provide such a value, nor how such a value is used to mix the ambient sound signal and the audio content signal as set forth. Claims 2-5 inherit this language of claim 1.

EX1004 at 788-789.

54. With respect to claim 6, the Patent Office stated

> Independent claim 6 sets forth that a mixed signal is generated by mixing ambient sound signal with a modified audio content signal and "if a user's voice is detected modifying the modified audio content signal before forming the mixed signal". The written disclosure as filed does not teach such an arrangement as set forth. The limitations of claims 7-17 inherit the language of claim 6.

EX1004 at 788-789.

55. On March 22, 2023, the applicant mailed a response to the December 22, 2022 office action that included amendments to the claims and arguments.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/592,143	CONFIRMATION NO. 3471
APPLICATION OF	:	Steven Wayne Goldstein	
DATE FILED	:	3 Feb. 2022	
EXAMINER	:	SNIEZEK, ANDREW L	
TC/A.U.	:	2688	
DOCKET	:	CONT-173218892	
TITLE	:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION	

RESPONSE TO OFFICE ACTION

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

Date: March 22, 2023

Sir:

These remarks are filed in response to the Office Action dated 22 Dec. 2022.

Please amend the above-identified patent application as follows:

- **Amendments to the Claims** begin on page 2 of this paper; and
- **Remarks/Arguments** begin on page 5 of this paper.

EX1004 at 887 (annotated)

56. The snip below shows the amendments that the applicant made to claim 1.

Listing of Claims:

1.(currently amended) A ~~wearable~~ device that can control ambient passthrough in an earpiece comprising:

a display;

a first memory that stores instructions; and

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

communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;

sending an audio content gain value and an ambient sound gain value ~~a mixing value~~ to the earpiece; and

sending an audio content signal to the earpiece, wherein the earpiece processor is configured to:

apply the audio content gain value to the audio content signal to generate a modified audio content signal;

apply the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal; and

mix ~~an~~ the modified ambient sound signal with the modified audio content signal to generate a mixed signal ~~based upon the mixing value~~.

EX1004 at 888

57. The snip below shows the amendments that the applicant made to claim 6.

6.(currently amended) A ~~wearable~~ device that can control ambient sound passthrough comprising:

a first microphone configured to measure a first acoustic environment to generate a first microphone signal;

a second microphone configured to be positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone is configured to generate a second microphone signal;

a speaker;

a memory that stores instructions; and

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

receiving an audio content signal;

modifying the audio content signal to generate a modified audio content signal;

modifying the first microphone signal to generate a modified first microphone signal;

modifying the second microphone signal to generate a modified second microphone signal;

generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination of both;

detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or both~~, and if an acute sound is detected, determining whether the acute sound is a user's voice~~;

modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal;

generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal~~, and if a user's voice is detected modifying the modified audio content signal before forming the mixed signal~~; and

sending the mixed signal to the speaker.

EX1004 at 889-890

58. The applicant also added new claims 18-20:

18.(new) The device according to claim 1, wherein the device is a portable communication device.

19.(new) The device according to claim 18, where the device portable communication device is at least one of a phone, a watch, a tablet, earpiece, a laptop computer or a combination thereof.

20.(new) The device according to claim 6, wherein the acute sound is a voice.

EX1004 at 892

59. The applicant stated that the written description support in the '143 application for amendments to claim 1 "can at least be found in the specification [0069] and in Figure 9." EX1004 at 894.

60. Paragraph [0069] of the '143 application and Figure 9 are coped below.

[0069] At step 922 the BNL is estimated using inputs from either or both the ASM signal at step 902, and/or the ECM signal at step 906. The BNL may be adjusted by the earpiece noise reduction rating 924. These signals are selected using the BNL input switch at step 918, which may be controlled automatically or with a specific user-generated manual operation at step 926. The Ear-Canal SNR is calculated at step 920 by differencing the ACL from step 914 and the BNL from step 922 and the resulting SNR 930 is passed to the method step 932 for AGC coefficient calculation. The AGC coefficient calculation 932 calculates gains for the Audio Content signal and ASM signal from the Automatic Gain Control steps 928 and 936 (for the Audio Content and ASM signals, respectively). AGC coefficient calculation 932 may use a default preferred SNR 938 or a user- preferred SNR 934 in its calculation. After the ASM signal and Audio content signal have been processed by the AGCs 928 and 936, the two signals are mixed at step 940.

EX1004 at 45

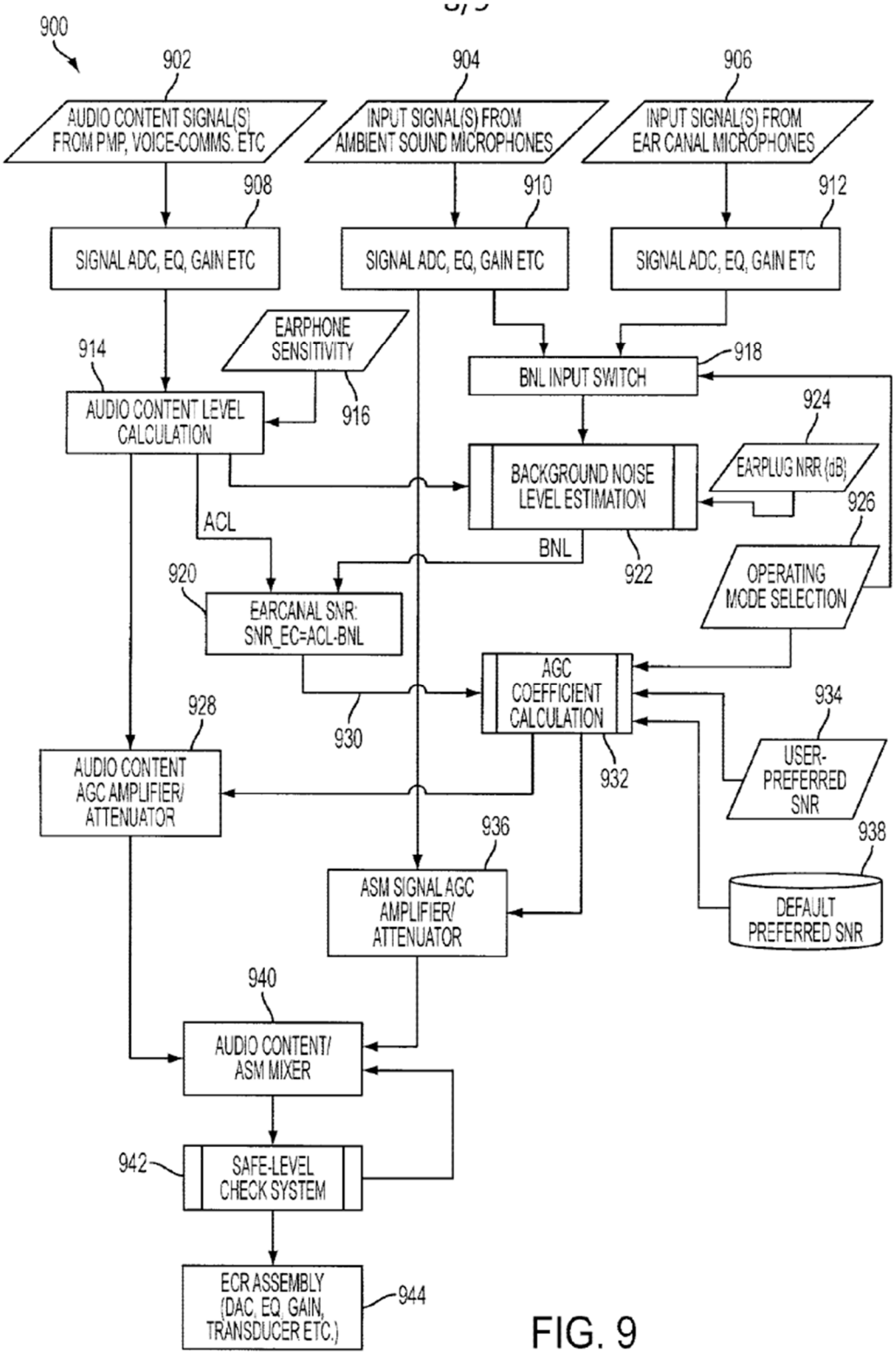


FIG. 9

EX1004 at 97

61. The applicant stated that the written description support in the '143 application for the amendments to claims 6 "can at least be found in the specification [0012], [0036] and in Figure 2." EX1004 at 895.

62. Paragraphs [0012], [0036], and Figure 2 of the '143 application are copied below.

[0012] In a second embodiment, an earpiece can comprise an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal, an audio interface operatively coupled to the processor to receive audio content, and a processor operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change, adjust an audio content level (ACL) of the audio content delivered to the ear canal, and reproduce the acute sound within the ear canal via the ECR responsive to detecting the acute sound and based on the ACL.

EX1004 at 32

[0036] Referring to FIG. 2, a block diagram of the earpiece 100 in accordance with an exemplary embodiment is shown. As illustrated, the earpiece 100 can include a processor 206 operatively coupled to the ASM 111, ECR 125, and ECM 123 via one or more Analog to Digital Converters (ADC) 202 and Digital to Analog Converters (DAC) 203. The processor 206 can monitor the ambient sound captured by the ASM 111 for acute sounds in the environment, such as an abrupt high energy sound corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., "help", "stop", "police", etc.), or specific noise type (e.g., breaking glass, gunshot, etc.). The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated storage memory 208 such as Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device 100. The memory 208 can store program instructions for execution on the processor 206 as well as captured audio processing data.

EX1004 at 36

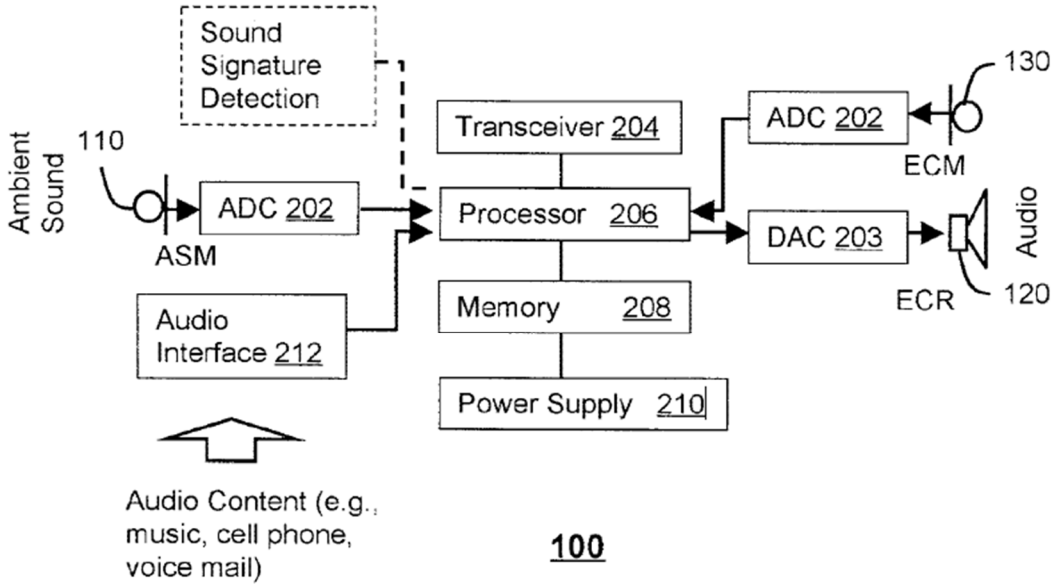


FIG. 2

EX1004 at 90

63. The applicant stated that support for new claims 18-20 "can be found [0013]."

REMARKS

The foregoing amendments and the following remarks are made in response to the Office Action dated 22 Dec. 2022. At the time of the Office Action, the Examiner noted that claims 1-17 were pending. The Examiner rejected claims 1-17 under 35 U.S.C. § 112(a), as discussed in further detail below. Claims 1-17 have been amended, and claims 18-20 are new (support can be found [0013]). In view of the arguments set forth below, claims 1-20 are allowable, and the Examiner is respectfully requested to withdraw the rejection and issue a Notice of Allowance.

EX1004 at 893(annotated)

64. Paragraph 13 of the '143 application is copied below.

[0013] The audio interface can receive the audio content from at least one among a portable music player, a cell phone, and a portable communication device. During operation, the processor can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal. In one arrangement, the processor can mute the audio content and pass the acute sound to the ECR for reproducing the acute sound within the ear canal. In another arrangement, the processor can amplify the acute sound with respect to the audio content level (ACL).

EX1004 at 32

65. On April 20, 2023, the Patent Office mailed an "Ex parte Quayle" office action allowing claims 1-20.



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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/592,143	02/03/2022	Steven Wayne Goldstein	11472-27-5_CONT-17321892	3471

150139 7590 04/20/2023
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

EXAMINER	
SNIEZEK, ANDREW L	

ART UNIT	PAPER NUMBER
2688	

NOTIFICATION DATE	DELIVERY MODE
04/20/2023	ELECTRONIC

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personics_docketing@cardinal-ip.com

EX1004 at 897 (annotated)

Office Action Summary	Application No. 17/592,143	Applicant(s) Goldstein et al.	
	Examiner ANDREW SNIEZEK	Art Unit 2688	AIA (FITF) Status No

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

Status

1)☑ Responsive to communication(s) filed on 3/23/23.

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

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

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

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

Disposition of Claims*

5) ☑ Claim(s) 1-20 is/are pending in the application.

5a) Of the above claim(s) _____ is/are withdrawn from consideration.

6) ☑ Claim(s) 1-20 is/are allowed.

7) ☐ Claim(s) _____ is/are rejected.

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

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

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

EX1004 at 898 (annotated)

66. The Patent Office provided a statement on the reasons for allowing claims 1-20. With respect to claim 1, the Patent Office stated:

> The following is an examiner's statement of reasons for allowance: Page 2 Boersma, considered to be the closest art of record teaches a device a processor (34) that is communicatively coupled to an earpiece (38) with the earpiece including a microphone (50) and a speaker (56); sending an audio content gain value and an ambient sound gain value to the ear piece (on line (88) after gain adjustments made by (84, 90), sending an audio content signal to

the earpiece (by closing switch (70a) to directly sent audio on line (80) to the earpiece; applying an audio content gain value (by use of (84)) to the audio content signal generating a modified audio content signal; applying the ambient sound gain value (by use of (90)) to an ambient sound signal to generate a modified ambient sound signal. This reference however functions differently that the claimed invention and does not teach additionally in combination a display, a processor executing stored instructions on a memory, an earpiece including each of a first microphone, a second microphone, a speaker, a second memory and an ear processor along with as set forth in claim 1 to "mix the modified ambient sound signal with the modified audio content signal to generate a mixed signal". In Boersma, modifying the audio content signal (using (84) occurs after the audio content signal and a modified ambient sound signal have been mixed by element (82). Also, in Boersma, it is not an earpiece processor the receives an audio content gain value and a ambient sound gain value, to form the modified signals and mix the modified signals. This combination of features as set forth are not an obvious variation of the teaching Boersma. The limitations of claims 2-5, 18-19 depend upon those features of claim 1.

EX1004 at 899-900

67. With respect to claim 6, the Patent Office stated:

Also, in addition to that discussed with respect to claim 1, Boersma teaches that the arrangement is used to detect an acute sound (such as sounds produced by police and sirens) as provided in applicants

claim 6. However, Boersma does not alone or obvious combination teach a device controlling ambient sound passthrough that includes the use of in combination first and second microphones, the signals produced by each being modified; generating an ambient sound signal from these modified signal(s), modifying a modified audio content signal if an acute sound is detected to generate a new modified audio content signal; generating a mixed signal from the ambient sound signal and the modified or new modified audio content signal which is then sent to the speaker as set forth in claim 6. The limitations o [*sic*] claims 7-17 and 20 depend upon those features of claim 6.

EX1004 at 900.

68. The Patent Office also included an objection with respect to the term "display" in claim 1:

Specification

The disclosure is objected to because of the following informalities: the specification is objected to as failing to provide proper antecedent basis for the claimed subject matter. See 37 CFR 1.75(d)(1) and MPEP § 608.01(o). Correction of the following is required: The written disclosure must make reference to a "display" as originally set forth in claim 1.

Appropriate correction is required.

EX1004 at 901.

69. On the same day the Patent Office mailed the office action (April 20, 2023), the applicant mailed a "312 Amendment in Response to Exparte

Quayle Action" including various amendments, including an amendment to claim 1.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/592,143	CONFIRMATION NO. 3471
APPLICATION OF	:	Steven Wayne Goldstein	
DATE FILED	:	3 Feb. 2022	
EXAMINER	:	SNIEZEK, ANDREW L	
TC/A.U.	:	2688	
DOCKET	:	CONT-17321892	
TITLE	:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION	

312 Amendment in Response to Exparte Quayle Action

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

Date: April 20, 2023

Sir:

These remarks are filed in response to the Ex parte Quayle Office Action dated 20 Apr. 2023. Please amend the above-identified patent application as follows:

- **Amendments to the Claims** begin on page 2 of this paper; and
- **Remarks/Arguments** begin on page 5 of this paper.

EX1004 at 926 (annotated)

70. The amendments to claim 1 and 6 are shown below.

Listing of Claims:

1.(currently amended) A device that can control ambient passthrough in an earpiece comprising:

~~a display;~~

a first memory that stores instructions; and

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

communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;

sending an audio content gain value and an ambient sound gain value to the earpiece; and

sending an audio content signal to the earpiece, wherein the earpiece processor is configured to:

apply the audio content gain value to the audio content signal to generate a modified audio content signal;

apply the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal; and

mixing the modified ambient sound signal with the modified audio content signal to generate a mixed signal.

EX1004 at 927

6.(currently amended) A device that can control ambient sound passthrough comprising:

a first microphone configured to measure a first acoustic environment to generate a first microphone signal;

a second microphone configured to be positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone is configured to generate a second microphone signal;

a speaker;

a memory that stores instructions; and

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

receiving an audio content signal;

modifying the audio content signal to generate a modified audio content signal;

modifying the first microphone signal to generate a modified first microphone signal;

modifying the second microphone signal to generate a modified second microphone signal;

generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof ~~of both~~;

detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof ~~both~~;

modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal;

generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal; and

sending the mixed signal to the speaker.

EX1004 at 928-929

71. The applicant explained that the "The Examiner allowed claims 1-20 but objected to the term "display" in claim 1. Claim 1 has been amended. In

view of the arguments set forth below, claims 1-20 are allowable, and the Examiner is respectfully requested to withdraw the objection and issue a Notice of Allowance on the currently amended claims." EX1004 at 932. The applicant further stated:

> The Examiner has objected to the specification and drawings as not implicitly showing a "display." The Applicant in the interest of expedited prosecution has amended claim 1 to remove "display" since the display is not referred to directly in any dependent claim. However, Applicant points out that the device can be a phone, tablet, watch or any other communication device, which inherently can include a display, as would be acknowledged by one of ordinary skill in the art. Hence the removal of the term "display" is not an admission that such a term is not inherently discussed within the specification. For at least these reasons, withdrawal of the specification and drawing objection is respectfully requested.

EX1004 at 932.

72. On May 23, 2023, the Patent Office mailed a Notice of Allowance.

NOTICE OF ALLOWANCE AND FEE(S) DUE

150139 7590 05/23/2023

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

EXAMINER	
SNIEZEK, ANDREW L	
ART UNIT	PAPER NUMBER
2688	

DATE MAILED: 05/23/2023

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/592,143	02/03/2022	Steven Wayne Goldstein	11472-27-5_CONT-17321892	3471

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

APPLN. TYPE	ENTITY STATUS	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	SMALL	$480	$0.00	$0.00	$480	08/23/2023

EX1004 at 934 (annotated)

73. In the Notice of Allowance the Patent Office noted that "Claims 1-5, 6-13, 14, 15-17, 18-19 and 20 have been renumbered as claims 1-5, 8-15, 19, 16-18, 6-7 and 20 respectively." EX1004 at 939.

74. On May 25, 2023, the applicant mailed another 312 amendment including various additional amendments. The applicant stated that the amendments "are clarifying amendments and are not intended to add any new matter." EX1004 at 973.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/592,143	CONFIRMATION NO. 3471
APPLICATION OF	:	Steven Wayne Goldstein	
DATE FILED	:	3 Feb. 2022	
EXAMINER	:	SNIEZEK, ANDREW L	
TC/A.U.	:	2688	
DOCKET	:	CONT-17321892	
TITLE	:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION	

312 Amendment

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

Date: May 25, 2023

Sir:

The claim amendments herein are clarifying amendments and are not intended to add any new matter. Please amend the above-identified patent application as follows:

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

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

EX1004 at 973 (annotated)

75. The amendments that the applicant made to claim 1 and claim 6 are shown below.

Listing of Claims:

1.(currently amended) A device that can control ambient passthrough in an earpiece comprising:

a first memory that stores instructions; and

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

communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;

sending an audio content gain value and an ambient sound gain value to the earpiece; and

sending an audio content signal to the earpiece, wherein the earpiece processor ~~is configured to:~~

appl[[y]]ies the audio content gain value to the audio content signal to generate a modified audio content signal[[;]], wherein the earpiece processor

appl[[y]]ies the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal[[;]], and wherein the earpiece processor

mix~~inges~~ the modified ambient sound signal with the modified audio content signal to generate a mixed signal.

EX1004 at 974

6.(currently amended) A device that can control ambient sound passthrough comprising:

a first microphone ~~configured to~~ that measures a first acoustic environment to generate a first microphone signal;

a second microphone ~~configured to be~~ that is positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone ~~is configured to~~ generates a second microphone signal;

a speaker;

a memory that stores instructions; and

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

receiving an audio content signal;

modifying the audio content signal to generate a modified audio content signal;

modifying the first microphone signal to generate a modified first microphone signal;

modifying the second microphone signal to generate a modified second nicrophone signal;

generating an ambient sound signal from at least one of the modified irst microphone signal, the modified second microphone signal or a combination hereof;

detecting an acute sound by analyzing at least one of the modified first nicrophone signal, the modified second microphone signal or a combination thereof;

modifying the modified audio content signal if an acute sound is letected to generate a new modified audio content signal;

generating a mixed signal by mixing the ambient sound signal with the nodified audio content signal or the new modified audio content signal; and

sending the mixed signal to the speaker.

EX1004 at 975-976

76. On June 5, 2023, the Patent Office mailed a response to the applicant's Rule 312 Communication. The Patent Office entered the applicant's amendment.

UNITED STATES PATENT AND TRADEMARK OFFICE

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

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/592,143	02/03/2022	Steven Wayne Goldstein	11472-27-5_CONT-17321892	3471

150139 7590 06/05/2023

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

EXAMINER	
SNIEZEK, ANDREW L	

ART UNIT	PAPER NUMBER
2688	

NOTIFICATION DATE	DELIVERY MODE
06/05/2023	ELECTRONIC

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

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

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

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

EX1004 at 986 (annotated)

Response to Rule 312 Communication	Application No.	Applicant(s)	
	17/592,143	Goldstein et al.	
	Examiner	Art Unit	AIA (FITF) Status
	ANDREW SNIEZEK	2688	No

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

1. ☑ The amendment filed on 25 May 2023 under 37 CFR 1.312 has been considered, and has been:
   - a) ☐ entered.
   - b) ☑ entered as directed to matters of form not affecting the scope of the invention.
   - c) ☐ disapproved because the amendment was filed after the payment of the issue fee.
     Any amendment filed after the date the issue fee is paid must be accompanied by a petition under 37 CFR 1.313(c)(1) and the required fee to withdraw the application from issue.
   - d) ☐ disapproved. See explanation below.
   - e) ☐ entered in part. See explanation below.
   - f) ☐ not entered because the supplemental or corrected Application Data sheet (ADS)
     - ☐ was not accompanied by a petition to accept an unintentionally delayed claim under 37 CFR 1.55 or 27 CFR 1.78;
     - ☐ did not identify the information being changed in accordance with 37 CFR 1.76(c)(2);
     - ☐ was not properly signed in accordance with 37 CFR 1.76(e) (or 37 CFR 1.33(b) for applications filed prior to September 16, 2012).

EX1004 at 987 (annotated)

77. The patent Office mailed an issue notification on July 25, 2023.



UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
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APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
17/592,143	07/25/2023	11710473	11472-27-5_CONT-17321892	3471

150130 7590 07/05/2023

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ISSUE NOTIFICATION

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

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

EX1004 at 998 (annotated)

C. THE CLAIMS

78. The table below includes identifiers for the '473 patent claims and limitations, which are referenced throughout.

Claim 1	
1.pre	A device that can control ambient passthrough in an earpiece comprising:
1.1	a first memory that stores instructions; and
1.2	a processor that executes the instructions to perform operations, the operations comprising:
1.3	communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;

1.4	sending an audio content gain value and an ambient sound gain value to the earpiece; and
1.5	sending an audio content signal to the earpiece,
1.6	wherein the earpiece processor applies the audio content gain value to the audio content signal to generate a modified audio content signal,
1.7	wherein the earpiece processor applies the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal, and
1.8	wherein the earpiece processor mixes the modified ambient sound signal with the modified audio content signal to generate a mixed signal.
Claim 2	
2	The device according to claim 1, wherein the earpiece processor calculates a noise level based upon a first microphone signal from the first microphone, wherein the earpiece processor modifies the mixed signal based upon the noise level.
Claim 3	
3	The device according to claim 1, wherein the earpiece processor generates the ambient sound signal from a first microphone signal from the first microphone.
Claim 4	
4	The device according to claim 1 further including a user interface.
Claim 5	
5	The device according to claim 1, wherein the earpiece processor applies an ambient gain to a first microphone signal from the first microphone to generate the ambient sound signal.
Claim 6	
6	The device according to claim 1, wherein the device is a portable communication device.
Claim 7	

7	The device according to claim 6, where the device portable communication device is at least one of a phone, a watch, a tablet, earpiece, a laptop computer or a combination thereof.
Claim 8	
8.pre	A device that can control ambient sound passthrough comprising:
8.1	a first microphone that measures a first acoustic environment to generate a first microphone signal;
8.2	a second microphone that is positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone generates a second microphone signal;
8.3	a speaker;
8.4	a memory that stores instructions; and
8.5	a processor that executes the instructions to perform operations, the operations comprising:
8.6	receiving an audio content signal;
8.7	modifying the audio content signal to generate a modified audio content signal;
8.8	modifying the first microphone signal to generate a modified first microphone signal;
8.9	modifying the second microphone signal to generate a modified second microphone signal;
8.10	generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;
8.11	detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;
8.12	modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal;
8.13	generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal; and

8.14	sending the mixed signal to the speaker.
Claim 9	
9	The device according to claim 8, wherein the operations further comprise: determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics of at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof.
Claim 10	
10	The device according to claim 9, wherein the operations further comprise: generating a noise signal from at least one of the modified first microphone signal, the modified second microphone signal, or both; generating a noise reduction signal.
Claim 11	
11	The device according to claim 10, wherein the operations further comprise: mixing the noise reduction signal with the mixed signal to generate a new mixed signal, and sending the new mixed signal to the speaker instead of the mixed signal.
Claim 12	
12	The device according to claim 11, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the new mixed signal if the user's voice is detected.
Claim 13	
13	The device according to claim 9, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the mixed signal if the user's voice is detected.
Claim 14	
14	The device according to claim 9, wherein the operations further comprise: detecting whether the acute sound is at least one of a warning sound, siren, voice, specific noise type, or a combination, by analyzing the spectrum of at least one of the modified first microphone signal, the modified second microphone signal, the modified ambient sound signal, or a combination.
Claim 15	

15	The device according to claim 14, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a warning or siren is detected.
Claim 16	
16	The device according to claim 14, where the operations further comprise: sending a notification signal to the speaker if the acute sound is a warning sound or siren.
Claim 17	
17	The device according to claim 16, wherein the warning sound is at least one of a bell, an emergency vehicle, the sound of a security system or a combination thereof.
Claim 18	
18	The device according to claim 14, wherein if an alarm is detected it is at least one of a police siren, an ambulance siren, a car honking or a combination thereof.
Claim 19	
19	The device according to claim 8, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a user's voice is detected.
Claim 20	
20	The device according to claim 8, wherein the acute sound is a voice.

VIII. OVERVIEW OF THE PRIOR ART

A. Background

1. Ambient Pass-Through and Mixing with Audio Content Were Well-known Before January 2007

79. The problem identified by the specification of the '473 patent related to users wearing earbuds listening to music and unable to hear their surroundings (EX1001, 1:66-2:5) was identified well before January 22, 2007

(the filing of the provisional application to which the '473 patent allegedly claims priority). So was the solution of providing "ambient pass-through" for headphones, e.g., equipping a pair of headphones with a microphone and "passing through" the ambient sound picked up by the microphone into the headphone speaker so that the user could hear the ambient sound without having to take off the headphones.

80. For example, U.S. Patent No. 4,455,677 ("*Fox*"), filed in May 1982, explains that "[o]ne of the problems encountered in the use of headphones used as a stereophonic system is that while the listener has the headphones on, he cannot hear extraneous noises such as doorbells, telephones, or other communications. Device for picking up such sounds are not known in the prior art, and yet it is desirable that such an apparatus be available and the reason for the desirability of such a device stems from the need to hear such extraneous raises in order to be aware whenever the telephone rings or the doorbell is sounded." EX1023, 1:16-25. Thus, *Fox* discloses a headphone system that is "structured so that when an ambient sound of sufficient magnitude is detected, an automatically actuated switch allows such sound signal to be first compressed through the voice compressor, then amplified and directed through to the headphones to be heard by the listener. The listener need not actuate any

switches, and can determine his course of action once such a signal is heard." *Id.*, 2:12-19.

81. As another example, U.S. Patent No. 4,827,458 ("*d'Arc*") (EX1024), filed in March 1988, explains that "miniature portable appliances, such as cassette players, compact disc players … have enjoyed great success for several years but they have the major disadvantage of acoustically isolating the user from the surrounding environment." EX1024, 1:18-26. "[S]uch isolation could subject the user to a wide range of accidents or risks." *Id.*, 1:28-30. "[T]he majority of users of portable appliances are, by virtue of their helmets or headphones, completely insensitive to calls, warning signals, and, above all, to any sudden, unexpected change in their surroundings." *Id.*, 1:33-37. "Furthermore, it is unpleasant, when calling someone nearby who is wearing a helmet or headphones, to have to shout in order to warn the person of some danger or simply to attract the person's attention." *Id*., 1:42-45. Accordingly, *d'Arc* discloses "providing an apparatus by which the headphone user is alerted in a reliable manner upon the occurrence of a surge in sound representing an abnormal event or a warning of which the user should be made aware while minimizing false alerts in response to high sound levels that do not represent dangerous situations." 1:58-65.

82. As another example, EP0495653A ("*Smiedt*"), filed in January 1992, discloses that "[p]ortable compact disc players and portable tape cassette players usually include headphones. Additionally, many people prefer to listen to music from a Hi-Fi set through headphones rather than by way of speakers in the room. Modern headphones are so efficient at shutting out ambient sounds that the wearer of the headphones is in effect isolated from the world around him. Consequently the user does not hear someone speak to him, or a telephone ringing or a knock on the door." EX1025, 1:4-13. Thus, *Smiedt* discloses "audio equipment which comprises a sound operated switch which is activated by an ambient sound which is above the response level of the switch, the switch being connected so that the output signal it produces on responding to said sound, alerts the person wearing the headphones that the switch has responded." *Id.*, 1:20-27.

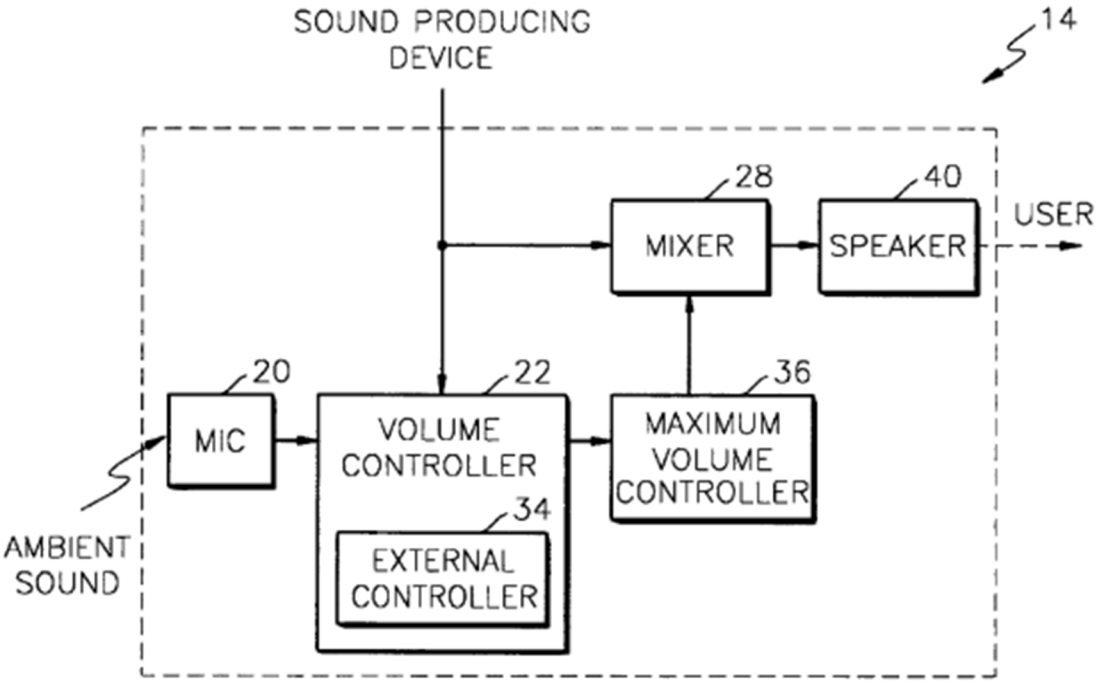
83. As another example, U.S. Patent No. 5,647,011 ("*Garvis*"), filed in March 1995, discloses that "[o]ne of the problems encountered in the wide usage of such headphone stereo systems is that while the listener has the headphones on, he cannot hear extraneous noises, such as sirens or emergency warning devices. When a person has earphones in place over the ears with the sound system playing loud music or the like, extraneous sound signals are completely obliterated. The signals thus become a hazard to the user of the headphones in

the case of emergency or warning signals and are inconvenient when anyone desires to communicate with a person using the headphone stereo system. The headphone wearer, in effect, is tuned out to loud from all extraneous noises and communications." EX1026, 1:16-27. *Garvis* notes that "[i]t has been suggested in the past to pick up extraneous sounds, especially in automobiles and vehicles, and recreate the sounds within the vehicle cab, either over the radio or to interrupt the radio and produce the sounds. This becomes important within vehicles since persons driving with the windows up and a radio playing are unable to hear sirens or emergency vehicles, which may be result in endangering people's lives." *Id.*, 1:29-35. "The problem is aggravated by portable stereo systems worn by people driving cars or in vehicles since, even if the radio is turned down and a warning is made within the vehicle, the wearer of the headphones would not hear the extraneous sounds." *Id.*, 1:36-40.

84. In view of those issues, *Garvis* discloses "a self-contained unit for attaching to the belt of an individual or to his clothing or body and which allows for the headphones to be plugged thereinto with the CD player or cassette player plugged thereinto so that the sound from the sound source to the headphones can be interrupted subject to predetermined sound levels being picked up by a microphone in the housing to switch off the connection between the sound

source and the earphone, which connection is manually reset at the option of the user. Extraneous sounds can also be processed as desired." *Id.*, 2:48-58.

85. As another example, U.S. Patent No. 6,782,106 ("*Kong*"), filed in May 2000, explains that "people frequently listen to sound of small and portable sound producing devices using sound receivers or earphones." EX1027, 1:29-35. "When using these sound receivers and earphones, it frequently happens that users do not hear ambient sound, for example, an alarm whistle which may be a danger signal, a warning voice or a calling sound. In this case, the user him/her self may be in danger or may be placing those around him/her self in danger." *Id.*, 1:43-48 . To solve that problem *Kong* discloses "an apparatus and method for transmitting the sound of a sound producing device to a user together with adjusted external sound." *Id.*, 2:3-6. "The apparatus includes an external sound receiver for receiving external sounds and converting them into an external sound signal, a volume controller for outputting only, the reproduced sound signal and the external sound signal having a volume exceeding a predetermined reference level, and a mixer for mixing the reproduced sound signal with the reproduced and external sound signals output from the volume controller and outputting the result." *Id*., 2:7-17.



Kong, FIG. 3

86. As another example, *Basu* (EX1028), published in May 2001, is titled "Smart Headphones: Enhancing Auditory Awareness Through Robust Speech Detection And Source Localization" and discloses "a method for enhancing auditory awareness by selectively passing speech sounds in the environment to the user." *Basu* at 1. With the "smart headphones," "a user can be listening to music over headphones and hear speech from specified directions mixed in." *Id*., Abstract; *see also* Preliminary Results ("All four reported that the system significantly improved their awareness of the speech around them without distracting them with all of the non-speech sounds, and found they could carry on a conversation with music playing.")

87. As another example, *Mueller* (EX1029), published in April 2002, is titled "Transparent Hearing." The "Transparent Headphones" "is a system in which sound is picked up just before it enters the ears, altered, and then played back in real-time, creating transparent augmented hearing." *Mueller* at 1. "The microphones pick up ambient sound, which is then amplified and sent to a computer, where it can be altered. The modified audio is then played back through the headphones. This allows switching between two modes: one sounds as if you have headphones on (opaque mode), dulling the external, environmental noise, while the other pipes through the audio from the microphones, sounding as if you are not wearing headphones at all (transparent mode)." *Id*. "Positions between these two extremes form a gradient from opacity to transparency and are called semi-transparent modes: each passes a different level of external audio to the ear." *Id*.

88. *Rast*, filed in April 2001, is titled "System And Method For Selective Control Of Acoustic Isolation In Headsets." *Rast* explains that "[a] wide variety of traditional headsets exist for noise abatement or for listening to an audio source, or communication. … Audio headsets are ubiquitous today for the reproduction of audio sources, in particular, music. … Headsets are often further equipped with noise cancelation circuitry which introduces a negative environmental noise feedback signal (opposing phase) into the audio source

being received, so that the ambient noise is further attenuated and thus acoustic isolation of the wearer from the environment increases." EX1030 ¶[0007].

89. *Rast* notes, however, "that being isolated from every sound within the external acoustical environment can often be inconvenient, or even dangerous." *Id.* ¶[0009]. "Therefore, a need exists for the development of methods for controlling acoustic isolation, so that important events are not missed and non-important noises can be properly identified and ignored." *Id*.

90. Thus, *Rast* discloses "environmentally responsive headsets" that are "capable of changing isolation mode in response to a manual trigger, or automatically in response to matching a sound from the external environment with a stored element of sound characterization information which is programmed within the apparatus." *Id*. ¶[0014]. "The mode of the headset may be changed by any of a number of mechanisms, for example, by temporarily routing external sound to earpieces (a 'hearthrough mode'), replaying the detected sound (temporally displaced sound—an 'echo back'), generating an alert signal ('user alert tone'), attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation ('block matched sound'), amplifying the ambient sounds after extracting spurious noises, accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants), combinations of the preceding modes, and so forth. Furthermore, if

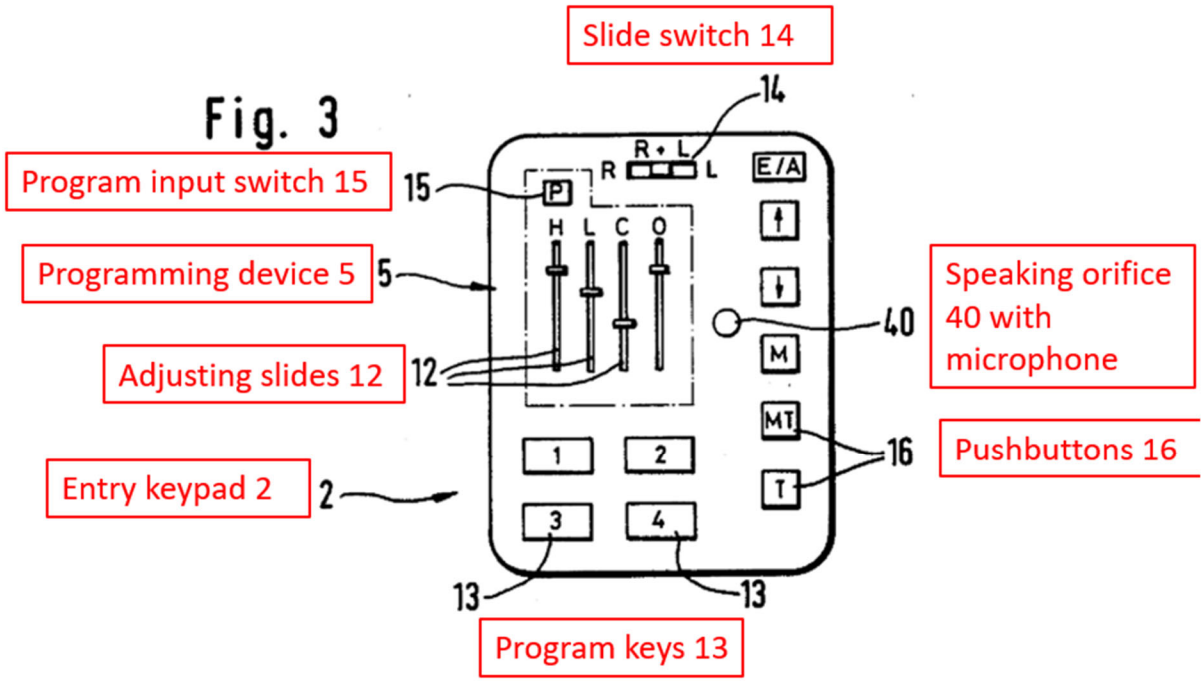
the headset is configured for listening to a programmed sound, such as that being received over a wired connection, or wireless receiver, then the audio amplitude of the programmed audio source may be temporarily attenuated, or disconnected, such that the external sound may be more easily discerned by the wearer." *Id*.

91. Accordingly, *Patel*, *Melanson*, *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* are just exemplary prior art references of many that pre-date January 2007 that are essentially directed to solving the same problem with very similar signal processing solutions, i.e., using a headset augmented with a microphone to pick up sound in the ambient environment, and under certain conditions, pass-through the ambient sound mixed with the user's audio content so that the user is made more aware of their surroundings while wearing the headset. A POSITA would have considered each of these references to be analogous to the '473 patent because they are within the same field of endeavor of the '473 patent and reasonably pertinent to one or more problems addressed by the '473 patent.

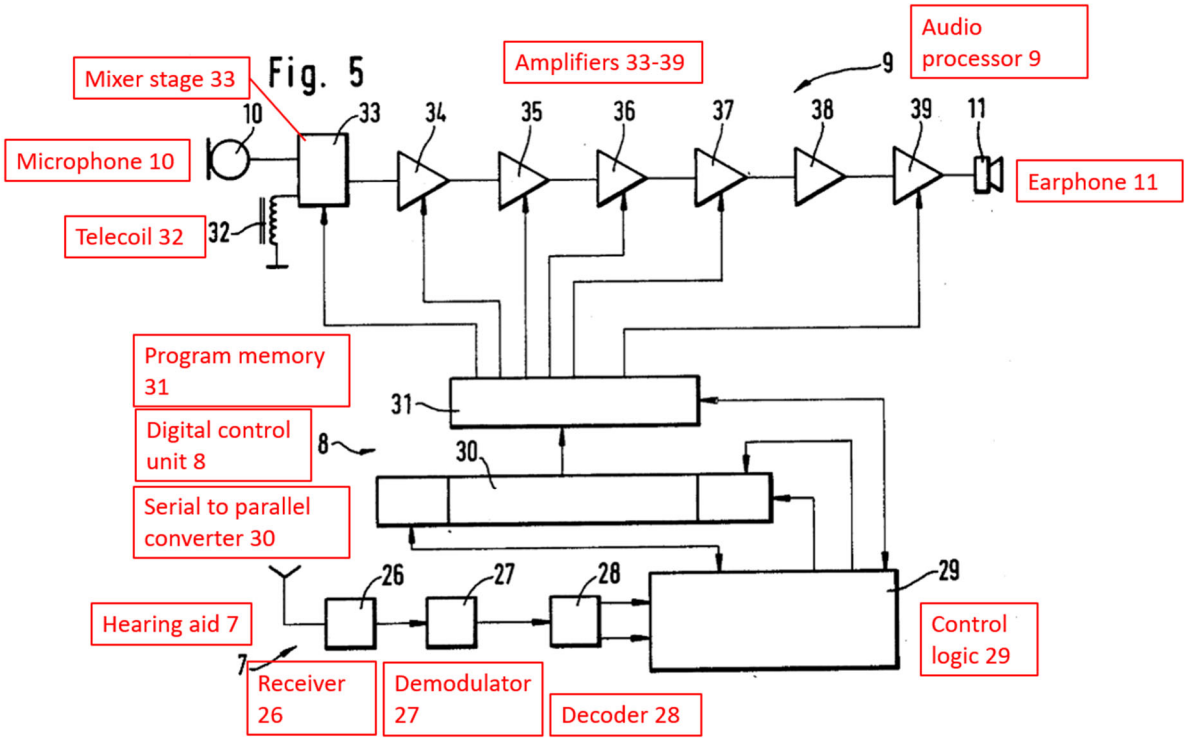
2. Earpiece Remote Controls Were Well-Known Before January 2007

92. Claims 1-7 of the '473 patent generally recite a device that sends an audio content signal, an audio content gain value, and an ambient sound gain value to an earpiece that applies the gain values to an ambient signal and audio signal and then mixes the signals together. It was well-known before January 22,

2007 to use a remote device to control the audio processing parameters of earpieces, particularly in the field of hearing aids. For example, U.S. Patent No. 4,947,432 ("*Topholm* '432"), filed January 22, 1987, discloses "a programmable hearing aid with an amplifier section and a transmission section whose transmission characteristics can be controlled." EX1031, 1:6-8. The programmable hearing aid is assigned "an external control unit with a modulatable transmitter for transmission of specific data, determining the transmission characteristics, by means of wireless transmission to the hearing aid." *Id.*, 2:34-42.

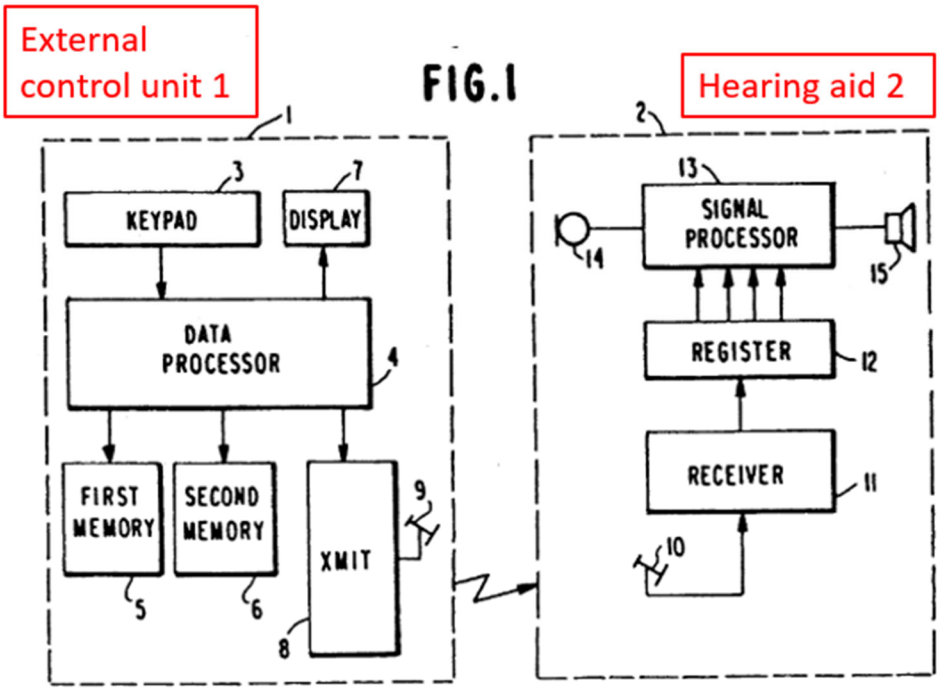


Topholm '432, FIG. 3 (annotated)



Topholm '432, FIG. 5 (annotated)

93. Similarly, U.S. Patent No. 5,202,927 ("*Topholm '927*"), issued April 13, 1993, discloses "a remote-controllable, programmable hearing aid system which consists of an external control unit 1 and an actual hearing aid 2." EX1032, 1:50-54. "The control unit 1 contains a keypad 3 … connected to a data processor 4, i.e. microprocessor, to which a first memory 5 for audiometry data and a second memory 6 for data describing various environmental situations are connected in turn." *Id.*, 1:64-2:4. "[A] transmitter 8 is also connected to the data processor 4, whereby the former is intended for wireless transmission of data/parameters to the actual hearing aid." *Id.*, 2:7-11.



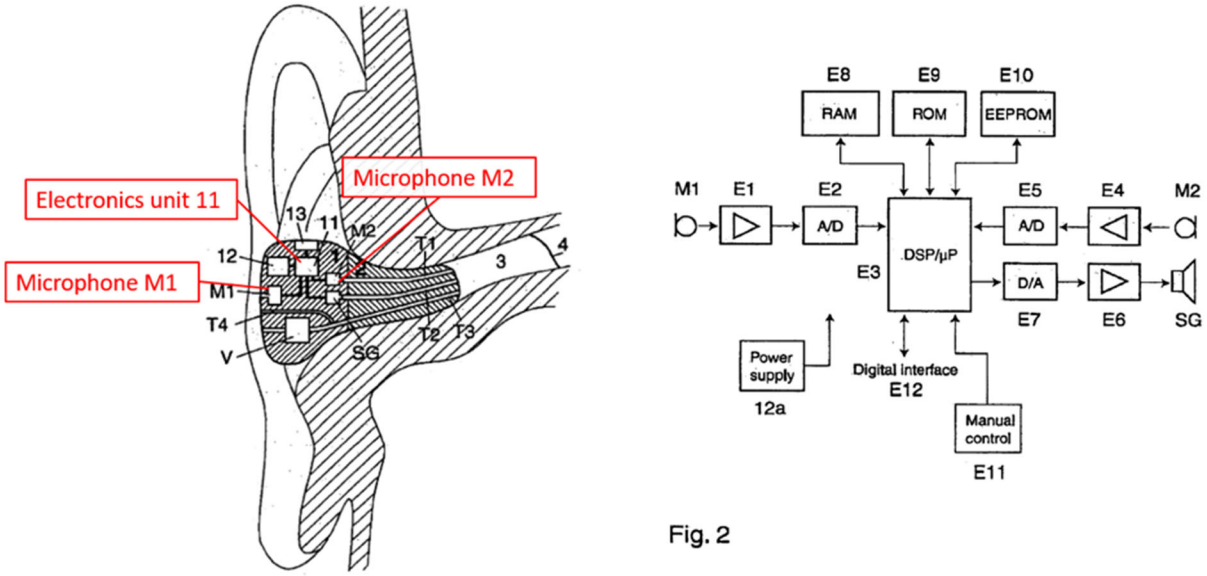
Topholm '927, FIG. 1 (annotated)

94. Thus, while *Patel* and *Topholm*, discussed below, generally disclose controlling the audio processing parameters of an earpiece with a remote device, similar technology and techniques had been known at least since the late 1980s and early 1990s.

3. Earpieces With Multiple Microphones, A Processor, And A Memory Were Well-Known Before January 2007

95. The '473 patent claims recite earpieces or devices that include two microphones, a processor, and a memory storing instructions executed by the processor. Earpieces with such components were well-known long before January 22, 2007. For example, *Kvaløy*, filed March 1, 2002 and titled "Voice Detection And Discrimination Apparatus And Method" discloses such an

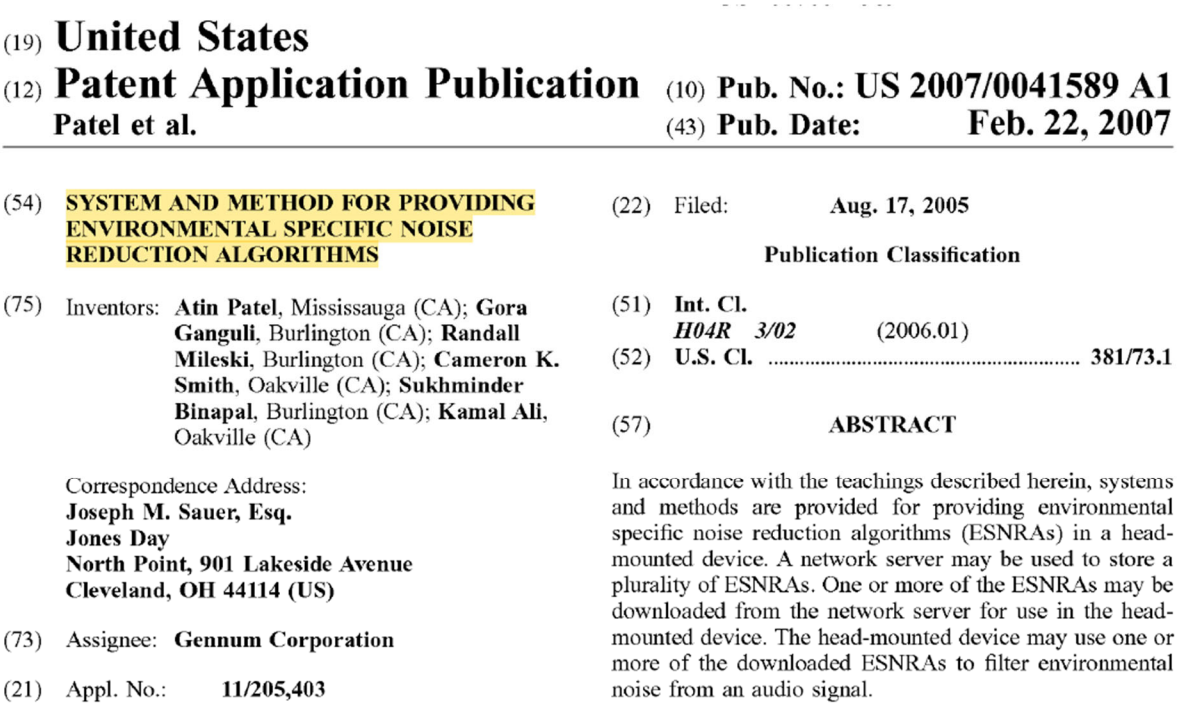
earpiece, as shown in Figures 1 and 2, below. Thus, *Patel* and *Melanson*, discussed below, are merely exemplary of earpiece form factors that were well-known and predictable by January 22, 2007.



Kvaløy, FIGs. 1, 2 (annotated)

B. Patel

96. *Patel* is titled "System And Method For Providing Environmental Specific Noise Reduction Algorithms."



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(54) **SYSTEM AND METHOD FOR PROVIDING ENVIRONMENTAL SPECIFIC NOISE REDUCTION ALGORITHMS**

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(57) **ABSTRACT**

In accordance with the teachings described herein, systems and methods are provided for providing environmental specific noise reduction algorithms (ESNRAs) in a head-mounted device. A network server may be used to store a plurality of ESNRAs. One or more of the ESNRAs may be downloaded from the network server for use in the head-mounted device. The head-mounted device may use one or more of the downloaded ESNRAs to filter environmental noise from an audio signal.

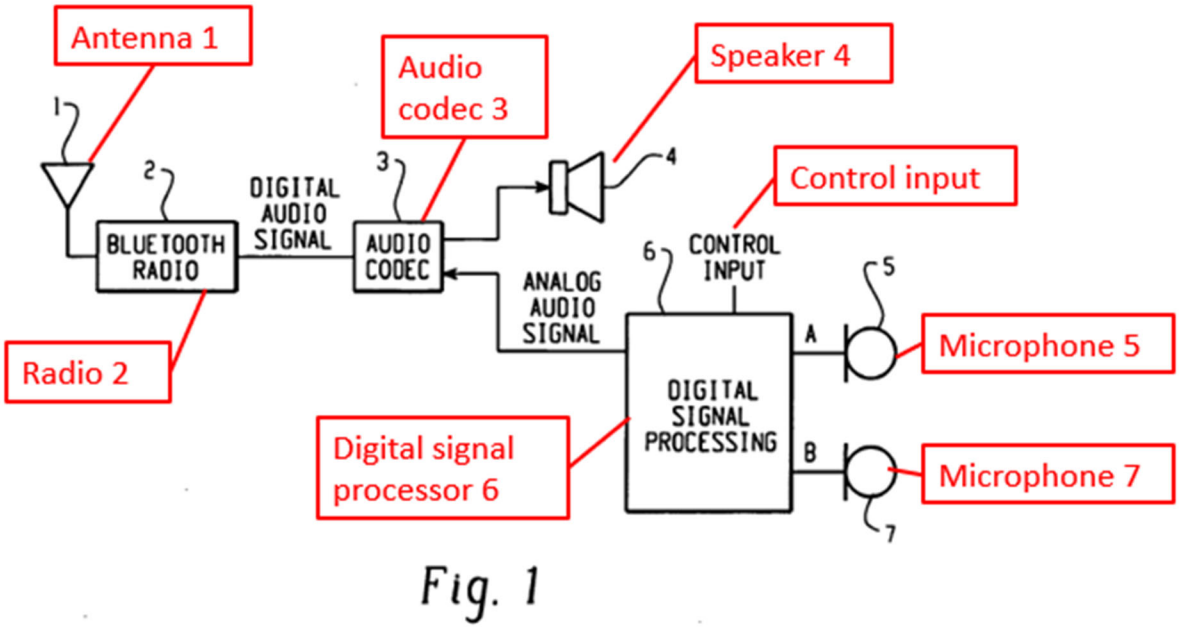
EX1014 at Cover (annotated)

97. *Patel*'s disclosure generally relates to "the field of communication head-mounted devices." *Id.* ¶[0001]. "The head-mounted device is capable of processing incoming noise with environment specific noise reduction algorithms and transmitting a noise-reduced sound wave to the user. In addition, the head-mounted device can be used as a digital hearing aid." *Id.*

98. *Patel* discloses multiple examples of a head-mounted device in Figures 1, 4, 6, 7, 8, 9, 13. *Patel* discloses that "[t]he illustrated wireless head-mounted device may, for example, be used to establish a wireless link (e.g., a Bluetooth link) with an external device, such as a cell phone or PDA, in order to send and receive audio signals. Other types of wireless links could also be

utilized, and the device may be configured to communicate with a variety of different external devices, such as cellular phones, PDAs, radios, MP3 players, CD players, portable game machines, etc." *Id*. ¶[0018].

99. As illustrated below, *Patel* discloses that the wireless head-mounted device example shown in Figure 1 includes "an antenna 1, a radio 2 (e.g., a Bluetooth radio), an audio codec 3, and a speaker 4. In addition, the wireless head-mounted device further includes a digital signal processor 6 and a pair of microphones 5, 7." *Id*. ¶[0018]. "Incoming audio signals may be transmitted from the external device over the wireless link to the antenna 1." *Id*. ¶[0019]. "The audio signals received by the microphones 5, 7 are routed to inputs A and B of the digital signal processor 6, respectively." *Id*. ¶[0020].



Patel, FIG. 1 (annotated)

100. Figure 2 is an example digital signal processor. *Id*. ¶[0021]. "The audio signals from the microphones 5, 7 are digitized by analog to digital converters (A/D) 13, processed through a filter bank 14 to optimize the overall frequency response and combined in a manner that can effectively create a desired directional response." *Id*. "By integrating a signal processor 6 and microphones 5, 7 into the communication head-mounted device, a directional response can be generated that eliminates the need for a mechanical boom extending out from the head-mounted device." *Id*. ¶[0022].

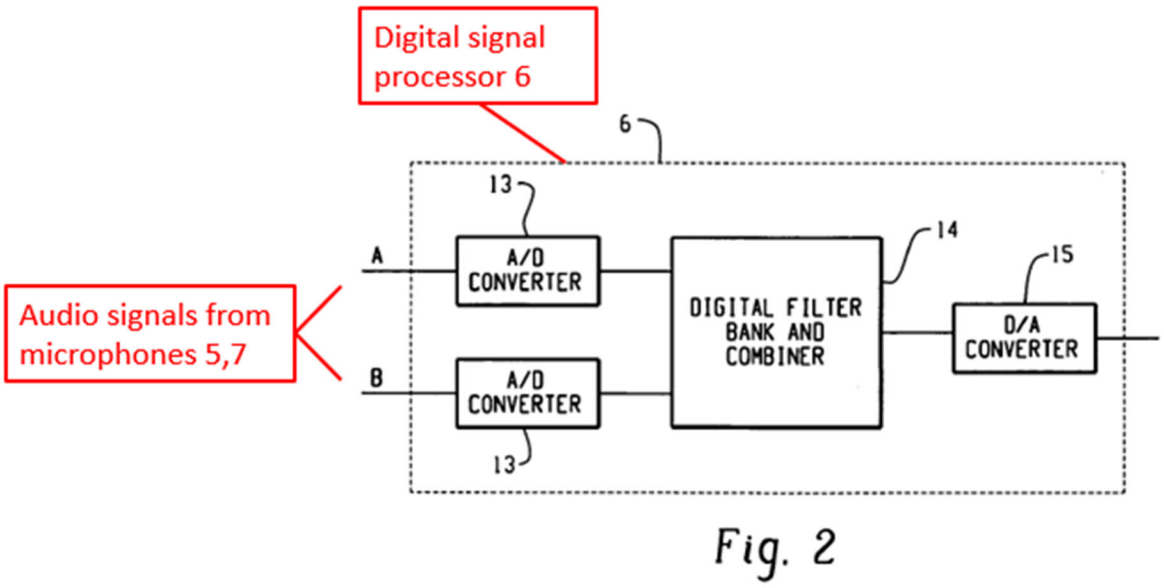


Fig. 2

Patel, FIG. 2

101. "The control input to the digital signal processor 6 may, for example, be used to switch between different head-mounted device modes (e.g., communication mode and hearing instrument mode)." *Id*. ¶[0025]. "The control

input may, for example, be received wirelessly via the radio 2, or may be received through a direct connection to the head-mounted device or via one or more user input devices on the head-mounted device (e.g., a button, a toggle switch, a trimmer, etc.)." *Id*.

102. Figure 4 illustrates another example of a communication head-mounted device in which a pair of signal processors 26, 28 are provided. *Id*. ¶[0026]. *Patel* notes, however, that "[i]t should be understood that although shown as two separate processing blocks in FIG. 4, the digital signal processors 26, 28 may be implemented using a single device." *Id*. ¶[0030].

103. "The analog audio output from the audio CODEC 23 is connected to input A of the signal processor 28, where it is digitized and processed to correct impairments in the overall frequency response." *Id*. "Input B of the signal processor 28 is connected 17 to one 27 of a pair of head-mounted device microphones 25, 27." *Id*.

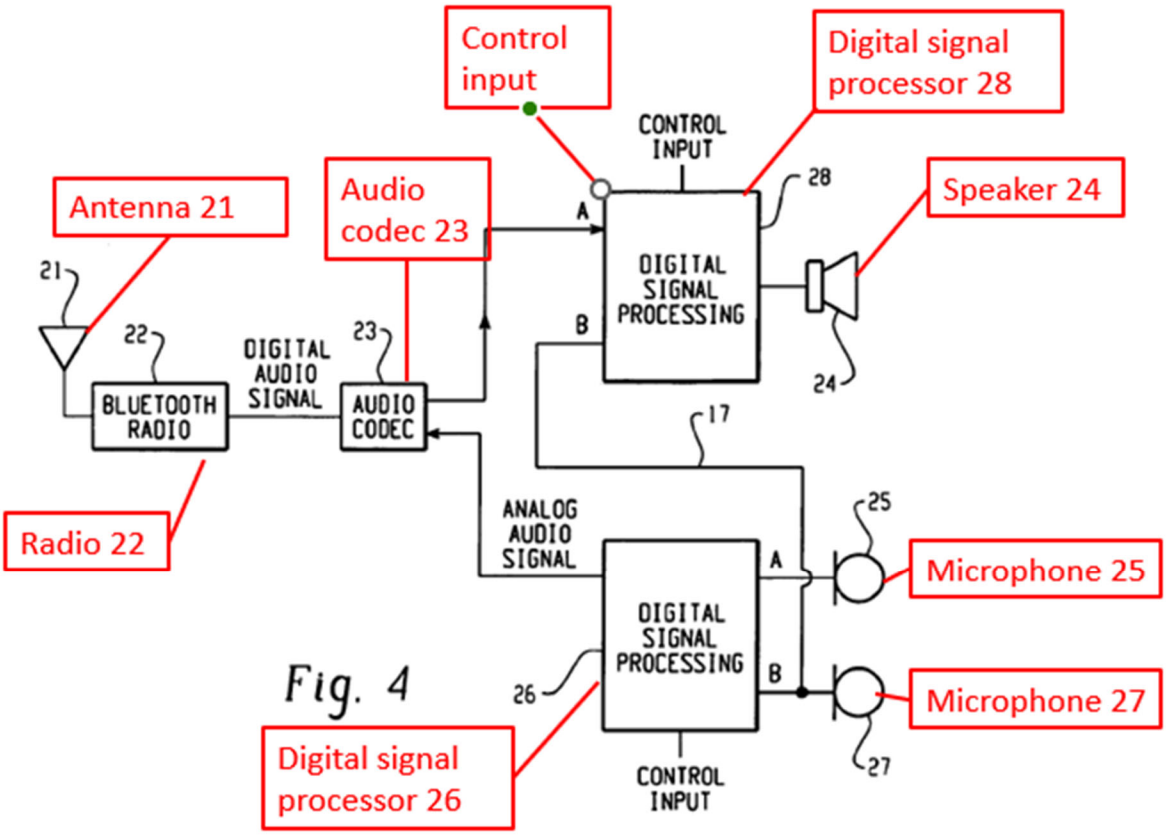


Fig. 4

EX1014, FIG. 4 (annotated)

104. "In one example, the head-mounted device microphone 27 connected to Input B of the signal processor 28 may be an inner-ear microphone." *Id*. ¶[0027]. "The audio signals received from the inner-ear microphone 27 may, for example, be used by the signal processor 28 to reduce the effects of occlusion, particularly when the head-mounted device is operating in a hearing instrument mode." *Id*. "In another example, the occlusion effect may be reduced by providing a sample of environmental sounds to the user's ear. In this example, the microphone 27 connected to Input B of the processor 28 may

be one of a pair of external microphones. Environmental sounds (i.e., audio signals from outside of the ear canal) may be received by the microphone 27 and introduced by the signal processor 28 into the audio signal being transmitted into the ear canal in order to reduce occlusion." *Id*. ¶[0028]. "By electronic (e.g., a control signal sent by a wireless or direct link) or manual means via the control input to the digital signal processor 28, the user may turn down or turn off the environmental sounds, for example when the head-mounted device is in a communication mode (e.g., when a cellular call is initiated or in progress.)." *Id*.

105. Figure 6 is another example of a communication head-mounted device and Figure 5 is another example of a digital signal processor 32 incorporated into the communication head-mounted device of Figure 6. *Id*. ¶[0031].

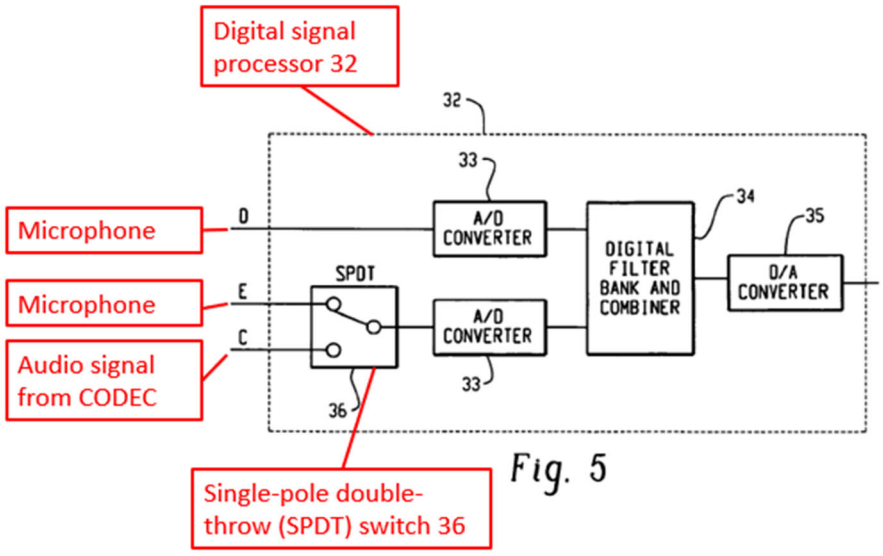


Fig. 5

EX1014, FIG. 5 (annotated)

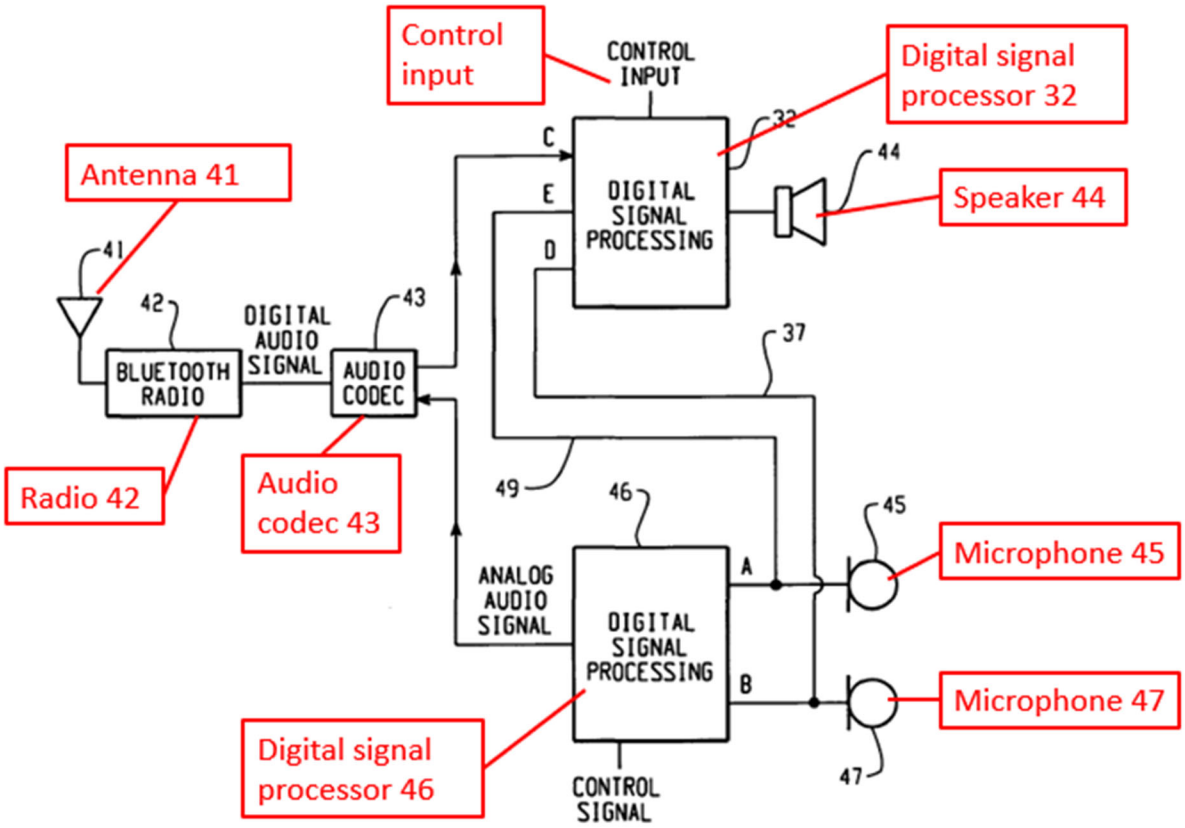


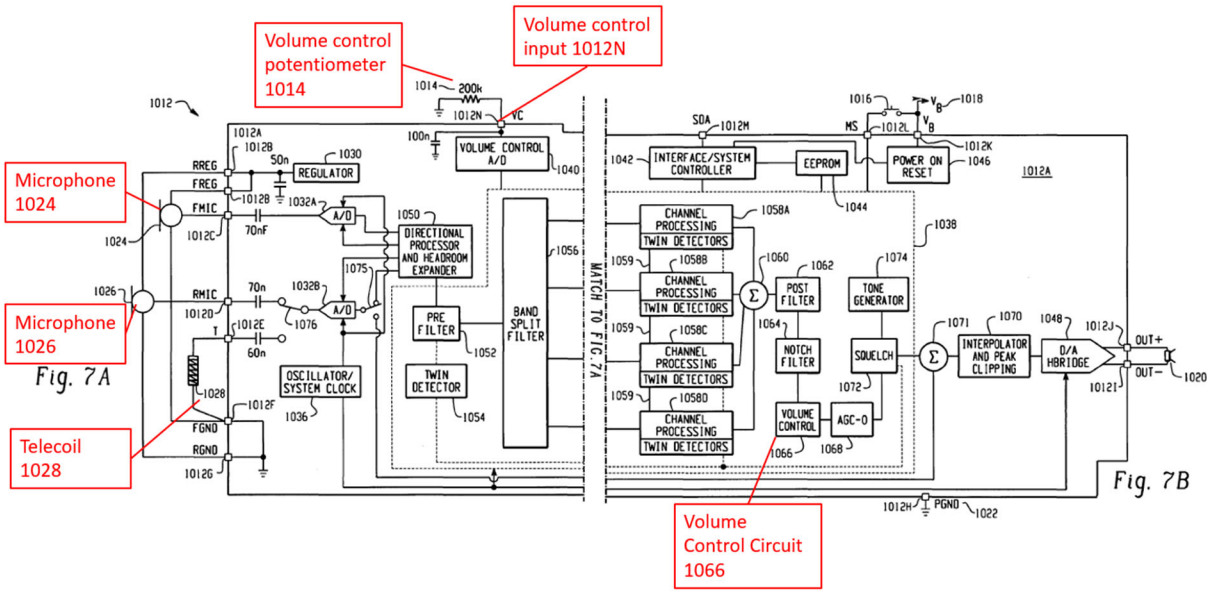
Fig. 6

EX1014, FIG. 6 (annotated)

106. In the example of Figure 6, "a single-pole double-throw (SPDT) switch 36 is added to the signal processing block 32. Inputs C and E to the digital signal processing block 32 are connected to the poles of the switch 36. The audio signal from an audio CODEC 43 is connected to input C and a microphone 45 is connected to input E of the signal processing block 32." *Id.* ¶[0031]. "[T]he switch 36 may be used to toggle the head-mounted device between a communication mode (e.g., a cellular telephone mode) and a hearing

instrument mode. For instance, when the head-mounted device is in communication mode, the switch 36 may connect audio signals (C) received from radio communications circuitry 42 (e.g., incoming cellular signals) to the signal processor 32, and may also connect omni-directional audio signals (D) from one of the microphones 47." *Id.* ¶[0032].

107. Figures 7A and 7B illustrate "a block diagram of an example digital hearing aid system 1012 that may be used in a communication head-mounted device." *Id.* ¶ [0033].



EX1014, FIGs. 7A, 7B (annotated)

108. "Sound signals input to the front and rear microphones 1024, 1026 are coupled to the front and rear A/ D converters 1032A, 1032B, which are preferably Sigma-Delta modulators followed by decimation filters that convert the analog sound inputs from the two microphones into a digital equivalent." *Id.*

¶[0042]. "The front and rear digital sound signals from the two A/ D converters 1032A, 1032B are coupled to the directional processor and headroom expander 1050 of the sound processor 1038." *Id*. ¶[0043].

109. "The pre-conditioned digital sound signal is then coupled to the band-split filter 1056, which preferably includes a bank of filters with variable corner frequencies and pass-band gains. These filters are used to split the single input signal into four distinct frequency bands." *Id*. ¶[0048]. "Channel processing of the four distinct frequency bands from the band-split filter 1056 is accomplished by a plurality of channel processing/twin detector blocks 1058A-1058D." *Id*. ¶[0049].

110. "After channel processing is complete, the four channel signals are summed by summer 1060 to form a composite signal." *Id*. ¶[0053]. "This composite signal is then coupled to the post-filter 1062, which may apply a post-processing filter function as discussed above. Following post-processing, the composite signal is then applied to a notch-filter 1064, that attenuates a narrow band of frequencies that is adjustable in the frequency range where hearing aids tend to oscillate." *Id*.

111. "Following the notch filter 1064, the composite signal is then coupled to a volume control circuit 1066. The volume control circuit 1066 receives a digital value from the volume control A/D 1040, which indicates the

desired volume level set by the user via potentiometer 1014, and uses this stored digital value to set the gain of an included amplifier circuit." *Id*. ¶[0054].

112. Figure 8 shows "an example of a communication head-mounted device that is configured to listen to a high fidelity external stereo audio source such as a CD player or MP3 player." *Id*. ¶[0059].

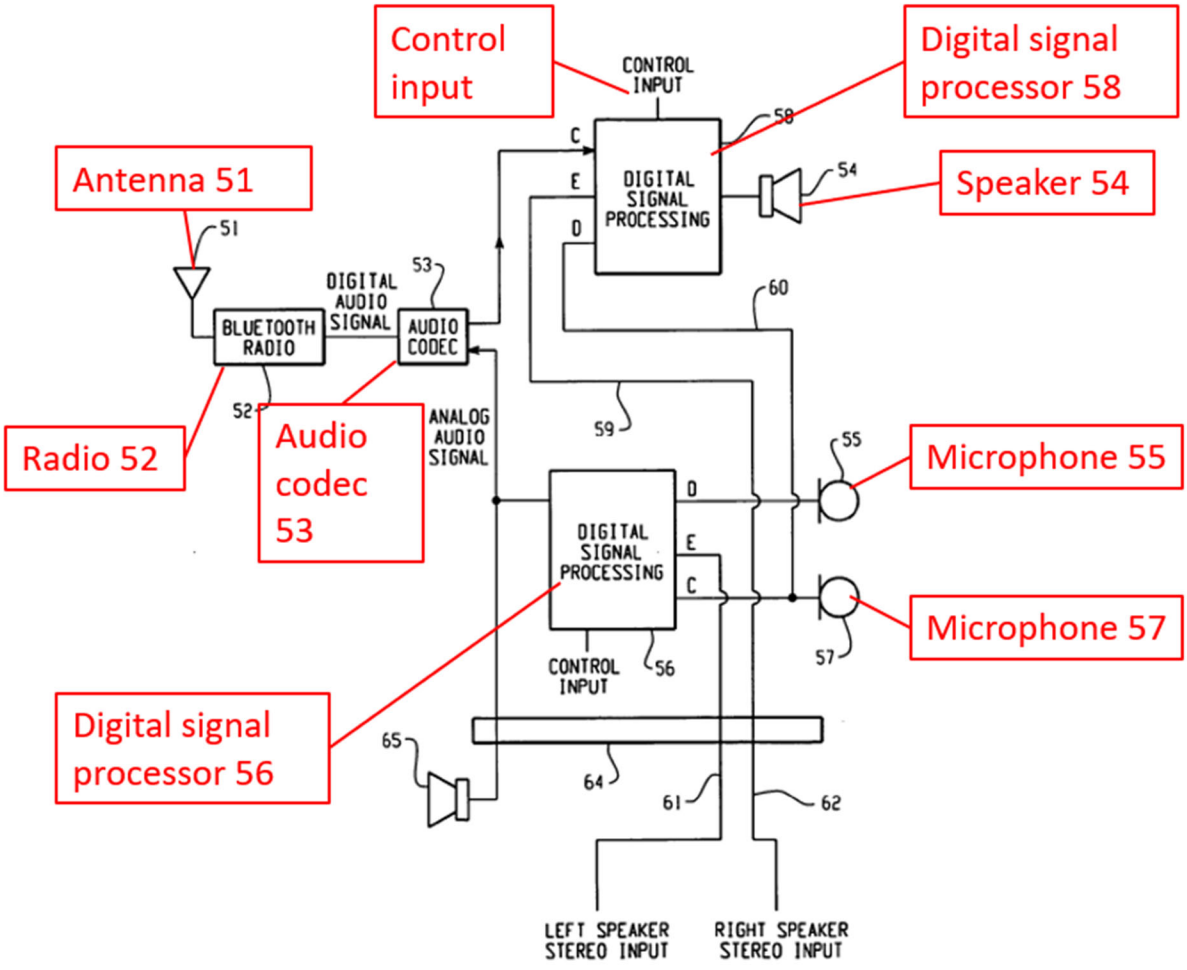


Fig. 8

EX1014, FIG. 8 (annotated)

113. In Figure 8, "the left and right side audio feeds 61, 62 from an external source are connected to input E on each digital signal processing block 56, 58, respectively, where the audio feeds 61, 62 are processed to provide an optimum audio response." *Id*. ¶[0059]. "When the user wishes to listen to the external stereo audio source, switches in both digital signal processing blocks 56, 58 may be set in position E to receive the stereo audio feed. When a call arrives, the switches in both digital signal processing blocks 56, 58 may be switched to position C, via the control input, in order to turn off the stereo feed and allows the user to answer the call." *Id*.

114. Figure 9 shows another example of a communication head-mounted device. *Id*. ¶[0060]. In this example, the device has "connections 86 and 87 from a radio communications circuitry 72 to a programming port of the digital signal processing blocks 76, 78." *Id*. If the head-mounted device is "configured in a stereo mode with left and right audio feeds 81, 82, then the digital signal processing blocks 76, 78, as a result of individually adjustable filters (amplitude and bandwidth) within the processors' filter banks, can be made to function as an audio equalizer." *Id*. "That is, the audio characteristics of the left and right audio feeds 81, 82 may be altered by the digital signal processing blocks 76, 78 using pre-programmed equalizer settings, such as amplitude and bandwidth settings." *Id*. "Using these settings, the digital signal processing blocks 76, 78 may divide

a given signal bandwidth into a number of bins, wherein each bin may be of equal or different bandwidths." *Id*. "In addition, each bin may be capable of individual amplitude adjustment." *Id*. "An application running on a computer, which emulates a graphical equalizer, can be displayed on a computer screen and adjusted in real time under user control." *Id*. "The equalizer settings may be transferred over the wireless link to the head-mounted device, where the amplitude and bandwidth settings for each filter within the filter bank of the signal processors 76, 78 are programmed via the programming ports of digital signal processing blocks 76, 78." *Id*. "It should be understood that other devices may also be used to program the head-mounted device equalizer settings, such as an MP3 player or other mobile device in wired or wireless communication with the head-mounted device." *Id*.

115. Figure 13 "shows an example head-mounted device 250 that contains a processor 260, a memory for storing [Environment Specific Noise Reduction Algorithms] 270, a user input device 280, and a speaker 290." *Id*. ¶[0066]. "[T]he processor 260 may be able to automatically recognize which [Environment Specific Noise Reduction Algorithm] would be most effective for a particular environment. The processor 260 is operable to filter environmental sounds using the selected ESNRA to create a filtered audio signal, which may be transmitted to the user 295 via the speaker 290." *Id*.; *see also id*. ¶[0064] ("In

another example, the [Environment Specific Noise Reduction Algorithms] 205 may be automatically selected from memory by a device processor based on an analysis of the environmental noise present in the audio signal(s).").

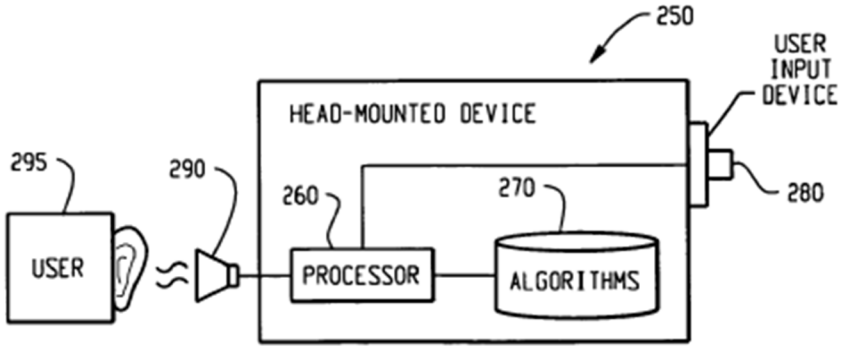


Fig. 13

EX1014, FIG. 13

116. "With respect to incoming signals, the [Environment Specific Noise Reduction Algorithms] may be applied to (a) reduce environmental noise present on the other end of the call or (b) to filter noise from the user's environment. Thus, the user can benefit by having the environmental noise present in the other party's location reduced, or by filtering environmental noise present in the user's location. In another example, the [Environment Specific Noise Reduction Algorithms] could be used to reduce noise from both the user's location and the other party's location at the same time." *Id*. ¶[0067]. "In another example, the head-mounted device may operate in a noise reduction only mode. In the noise reduction only mode the [Environment Specific Noise Reduction Algorithm] function is applied to environmental signals in the user's location. This mode, for

example, would aid users in hearing face-to-face communications in noisy environments. It could also be used, for example, to listen to television or music in a noisy environment." *Id*. ¶[0068].

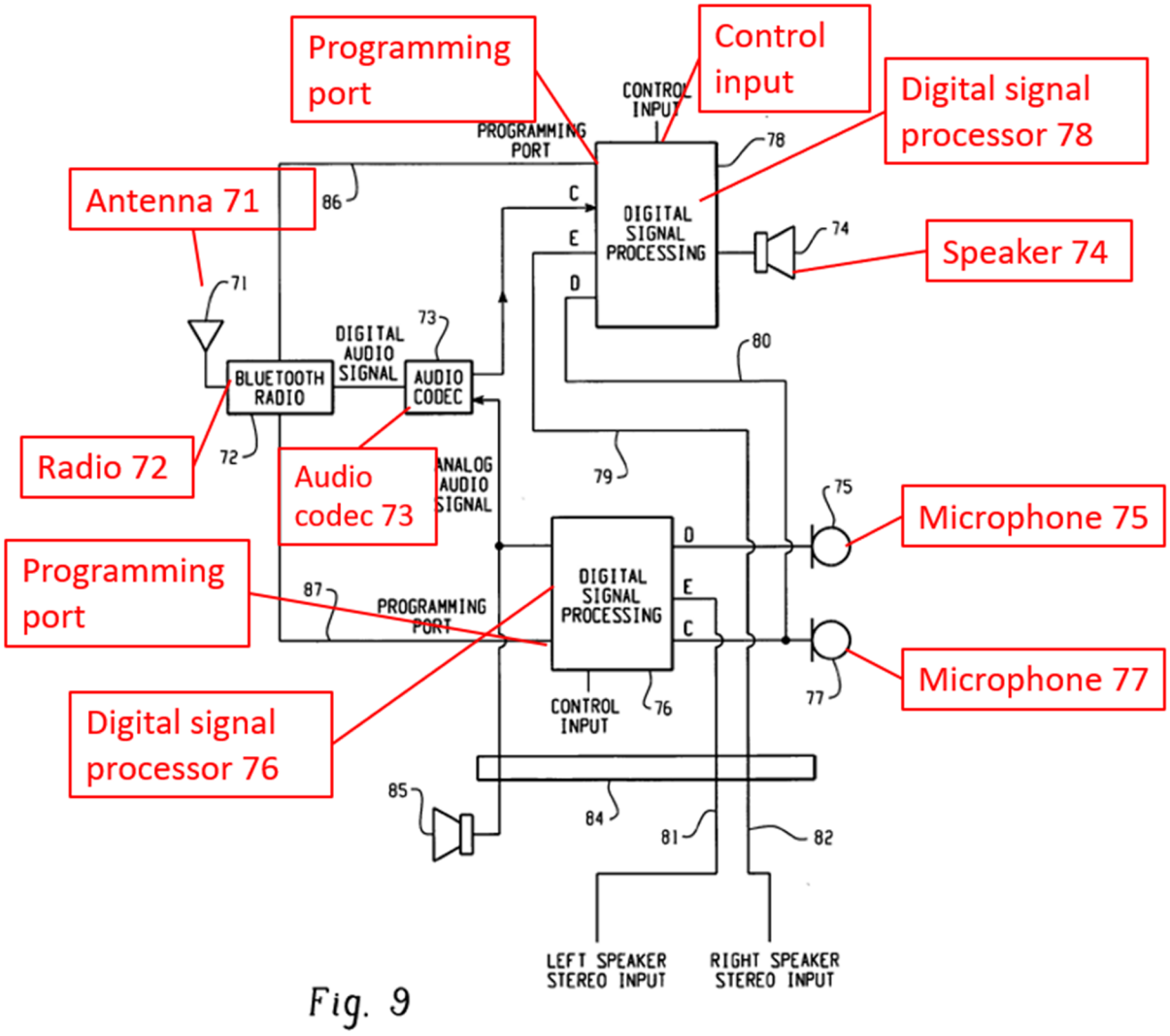


Fig. 9

EX1014, FIG. 9

C. Topholm

117. *Tophpolm* is titled "Remote control system for a hearing aid."

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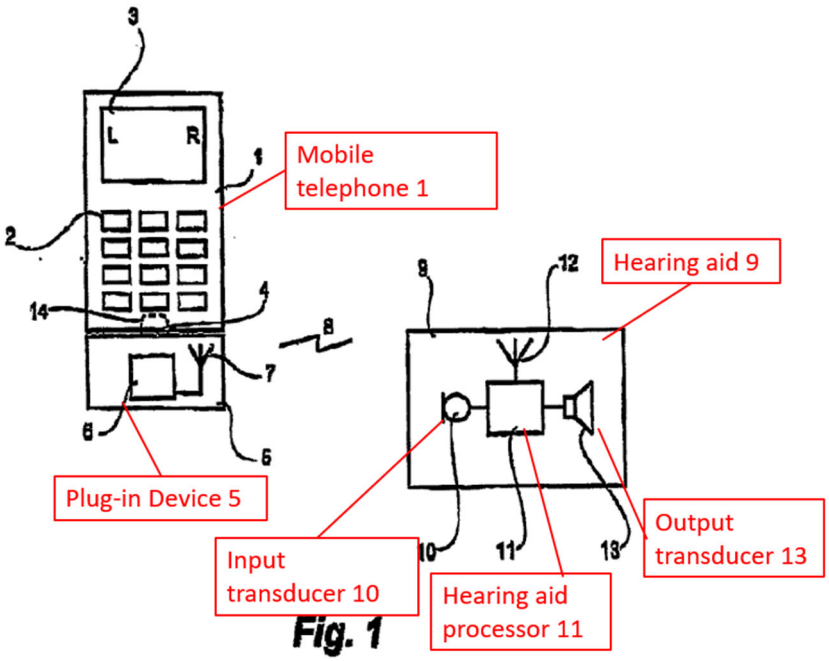
(54) Title: REMOTE CONTROL SYSTEM FOR A HEARING AID

EX1015 at Cover (annotated)

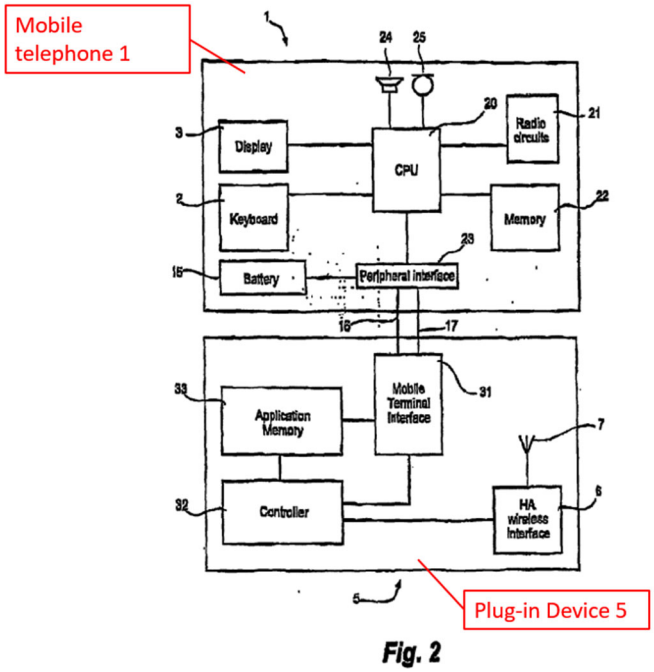
118. *Tophpolm* relates to "programmable hearing aids with wireless remote control capabilities and to devices for wireless remote control of hearing aids." *Id.*, 1:3-5. *Topholm* explains that wireless remote controls for user control of hearing aids are known. *Id.*, 1:12. "They are usually dedicated, hand-held devices for controlling simple functions, e.g. regulating the output volume from the hearing aids or changing among different programs stored in the hearing aids." *Id.*, 1:12-15. "Upgrading the remote controls with new functionality may

be difficult or impossible because of this dedication, and a remote control device may also easily get lost or mislaid." *Id*., 1:15-17.

119. Due to the above issues, *Topholm* discloses a mobile telephone to double as a user interface of a wireless remote control of a hearing aid." *Id*., 2:7-8. *Topholm* discloses that the mobile telephone connects to a plug-in device. *Id*., 2:16-20. Figure 1 "shows a plug-in device 5 connected to a mobile telephone 1 through a plug 4 and a socket 14 for the purpose of remotely controlling a hearing aid 9." *Id*., 3:13-17. Figure 2 "shows a more detailed view of a mobile telephone 1 with the plug-in device 5." *Id*., 4:5-6. The mobile telephone includes a "CPU 20 connected to a keypad 2, a display 3, radio circuits 21 for mobile telephone communications, a memory 22, and a peripheral interface 23 having means for connecting various peripheral devices to the mobile telephone 1." *Id*., 4:6-9.



EX1015, FIG. 1 (annotated)



EX1015, FIG. 2 (annotated)

120. "While the plug-in device is plugged in, the keypad 2 and the display 3 of the mobile telephone 1 are used to remotely operate a hearing aid (ref. fig. 1)." *Id*., 5:4-5. "[T]he keypad 2 is used to enter user commands such as requests for program changes and volume adjustments in the hearing aid, the changes being reflected in the display 3 and in the behavior of the hearing aid." *Id*., 5:5-10. "Digital audio stored in the memory 22 of the mobile telephone may also be transmitted to the hearing aid (not shown) via the plug-in device 5." *Id*., 6:5-9.

121. Figure 3 shows "an algorithm showing the basic flow of communication between a mobile telephone 1 and a plug-in device 5 as shown in figs. 1 and 2." *Id*., 6:14-15. "When the device 5 is plugged into the mobile telephone 1, the application is loaded from the application memory 33 to the mobile telephone memory 22." *Id*., 6:25-26. "When a key is pressed, the routine determines the associated command and executes it in step 104." *Id*., 7:11-12. "The command associated to the particular key press is prepared by the controller 32 of the plug-in device 5 and transmitted to the hearing aid using the wireless interface 6 and the antenna 7." *Id*., 7:13-15.

D. Williamson

122. *Williamson* is titled "Adaptive, programmable signal processing and filtering for hearing aids."

United States Patent [19]

Williamson et al.

[11] **Patent Number: 5,027,410**

[45] **Date of Patent: Jun. 25, 1991**

[54] **ADAPTIVE, PROGRAMMABLE SIGNAL PROCESSING AND FILTERING FOR HEARING AIDS**

[75] Inventors: **Malcolm J. Williamson; Kenneth L. Cummins; Kurt E. Hecox,** all of Madison, Wis.

[73] Assignee: **Wisconsin Alumni Research Foundation,** Madison, Wis.

[21] Appl. No.: **269,937**

[22] Filed: **Nov. 10, 1988**

[51] **Int. Cl.**[5] **H04R 25/00**

[52] **U.S. Cl.** **381/68.4**; 364/724.01

[58] **Field of Search** 381/68.4, 68, 68.2, 381/71, 73.1, 94, 104, 106, 107, 99; 333/166, 167, 168, 173, 14; 328/167; 364/724.01, 724.08, 724.09

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Primary Examiner—Forester W. Isen

Attorney, Agent, or Firm—Foley & Lardner

[57] **ABSTRACT**

A hearing aid system utilizes digital signal processing to correct for the hearing deficit of a particular user and to maximize the intelligibility of the desired audio signal relative to noise. An analog signal from a microphone is converted to digital data which is operated on by a digital signal processor, with the output of the digital signal processor being converted back to an analog signal which is amplified and provided to the user. The digital signal processor includes a time varying spectral filter having filter coefficients which can be varied on a quasi-real time basis to spectrally shape the signal to match the hearing deficit of the user and to accommodate ambient signal and noise levels. The coefficients of the spectral filter are determined by estimating the energy in several frequency bands within the frequency range of the input signal, and using those energy estimates to calculate desired gains for the frequency bands and corresponding spectral filter coefficients. The spectral energy analysis may be carried out using pairs of high pass and low pass digital filters in cascade relation, with the output of each low pass filter being provided to the next pair of high pass and low pass filters. The rate at which output data is provided from the filters in each pair may be reduced from the sample rate of input data by one half for succeeding pairs of filters in the cascade to thereby reduce the computation time required.

EX1016, Cover

123. As background, *Williamson* explains that "[c]onventional hearing aids which provide pure linear amplification inevitably amplify the ambient noise as well as the desired signal, such as speech or music, and thus do not

improve the signal to noise ratio." *Williamson*, 1:35-43. "Moreover, even if an aid is relatively well matched to an individual's hearing deficit under certain conditions, such as a low noise environment where speech is the desired signal, the aid may perform poorly in other environments such as one in which there is high ambient noise level or relatively high signal intensity level." *Id*., 1:48-54.

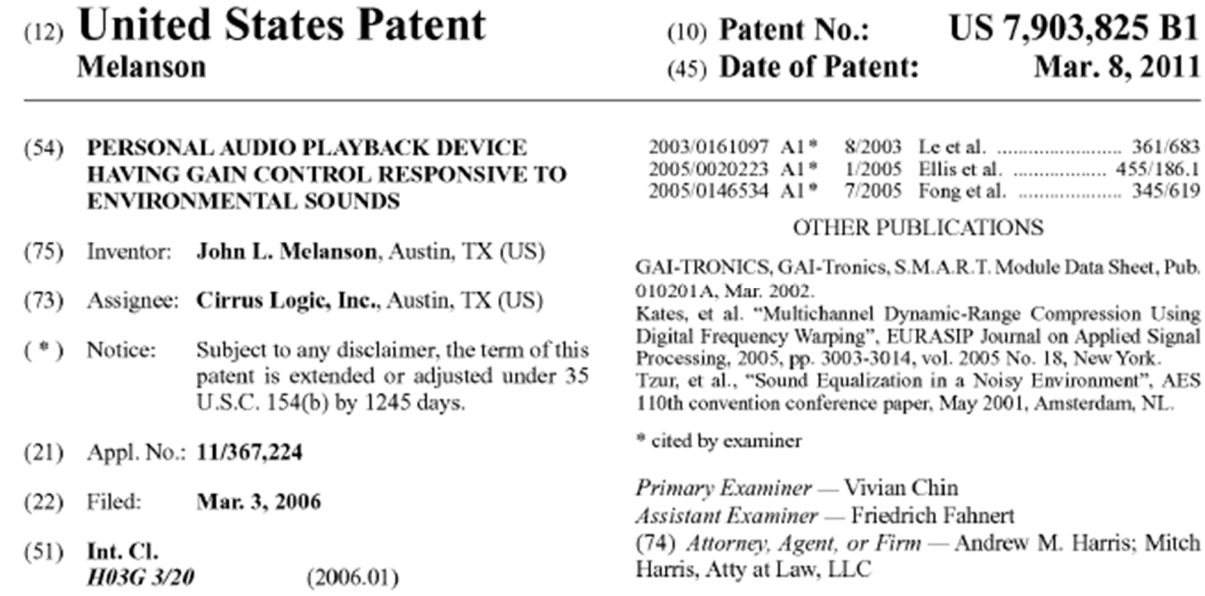
124. Thus, *Williamson* discloses "a hearing aid system which is both programmable to fit the hearing deficit of a particular user and adaptive to the sound environment to maximize the intelligibility and quality of the audio signal provided to the user." *Id*., 1:57-62. "Background noise levels are reduced in either a fixed or an adaptive manner to enhance the signal to noise ratio of the desired signal, such as speech." *Id*., 1:62-65. "The signal processing hearing aid includes a microphone preferably located near or at the ear of the wearer, associated analog filtering and amplifying circuits, an analog to digital converter for converting the analog signal to digital data, a digital signal processor which operates on the digital data, a digital to analog converter for converting the processed data back to analog signal form, and analog filters and amplifiers which drive a receiver or speaker in an ear piece worn by the user." *Id*., 2:15-24.

125. "An estimate of the level of background noise is preferably made from the energy envelope of the input signal in various frequency ranges." *Id*.,

2:67-3:1. This estimate of the noise is used to adjust a spectral filter. *See id.*, 2:51-3:16, 3:36-41.

E. Melanson

126. *Melanson* is titled "Personal Audio Playback Device Having Gain Control Responsive To Environmental Sounds."



(12) **United States Patent**
Melanson

(10) **Patent No.: US 7,903,825 B1**
(45) **Date of Patent: Mar. 8, 2011**

(54) **PERSONAL AUDIO PLAYBACK DEVICE HAVING GAIN CONTROL RESPONSIVE TO ENVIRONMENTAL SOUNDS**

(75) Inventor: **John L. Melanson**, Austin, TX (US)

(73) Assignee: **Cirrus Logic, Inc.**, Austin, TX (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 1245 days.

(21) Appl. No.: **11/367,224**

(22) Filed: **Mar. 3, 2006**

(51) **Int. Cl.**
H03G 3/20 (2006.01)

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Assistant Examiner — Friedrich Fahnert
(74) *Attorney, Agent, or Firm* — Andrew M. Harris; Mitch Harris, Atty at Law, LLC

EX1017 at Cover

127. As background, *Melanson* explains that "to provide the best listening experience, the headphones used with present-day personal audio devices have improved to the point that outside environmental sounds are attenuated quite severely and the transducers themselves have improved to provide very high acoustic program levels from typically low power levels available from such devices." *Id.*, 1:22-27. "Although the increased loudness and environmental attenuation is preferable for uninterrupted listening," there are

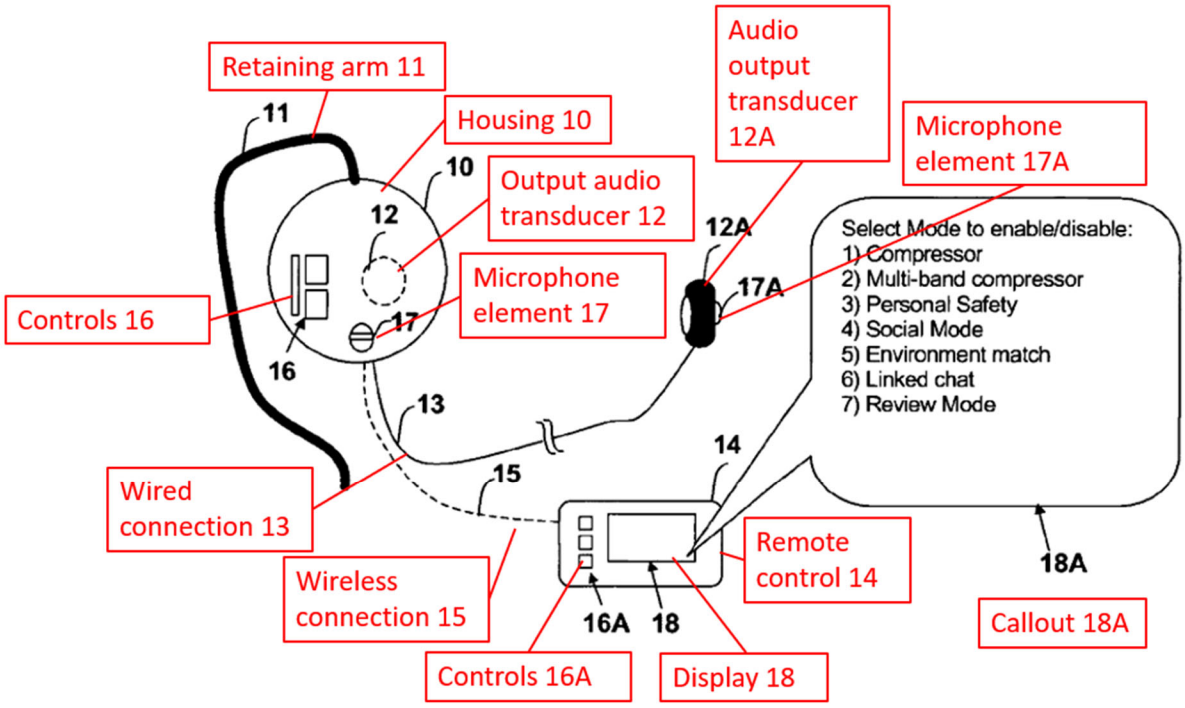
drawbacks. *Id.*, 1:29-31. "For example, it has become increasingly difficult to get the attention of a personal audio device user in order to converse with them, and conversation between persons using personal audio devices tends to be mutually exclusive of such use. … As another example, a pedestrian listening to audio via such a device may not notice a vehicle horn or siren that is alerting them to a hazard." *Id.*, 1:31-41.

128. "Recently, the technique of noise cancellation has been applied to consumer headphones. … However, such a device does not solve the above-described problems of providing for conversations, and if an environmental noise indicative of a hazard is not sufficiently loud to as to defeat the noise-canceling mechanism, then the noise cancellation will also not produce a desirable result." *Id.*, 1:58-2:2.

129. "Therefore, it would be desirable to provide a personal consumer audio device that provides a quality individual program listening experience in the presence of environmental noise, while providing for communications with others and awareness of environmental sounds indicative of a hazard." *Id.*, 2:3-8.

130. *Melanson* Figure 1 discloses one example of a personal consumer audio device. *Id.*, 2:65-67. "A retaining arm 11 is attached to a device housing 10 for retaining device housing 10 to a wearer's ear." *Id.*, 3:36-37. "An output audio transducer 12 is included on housing 10 to provide audio program

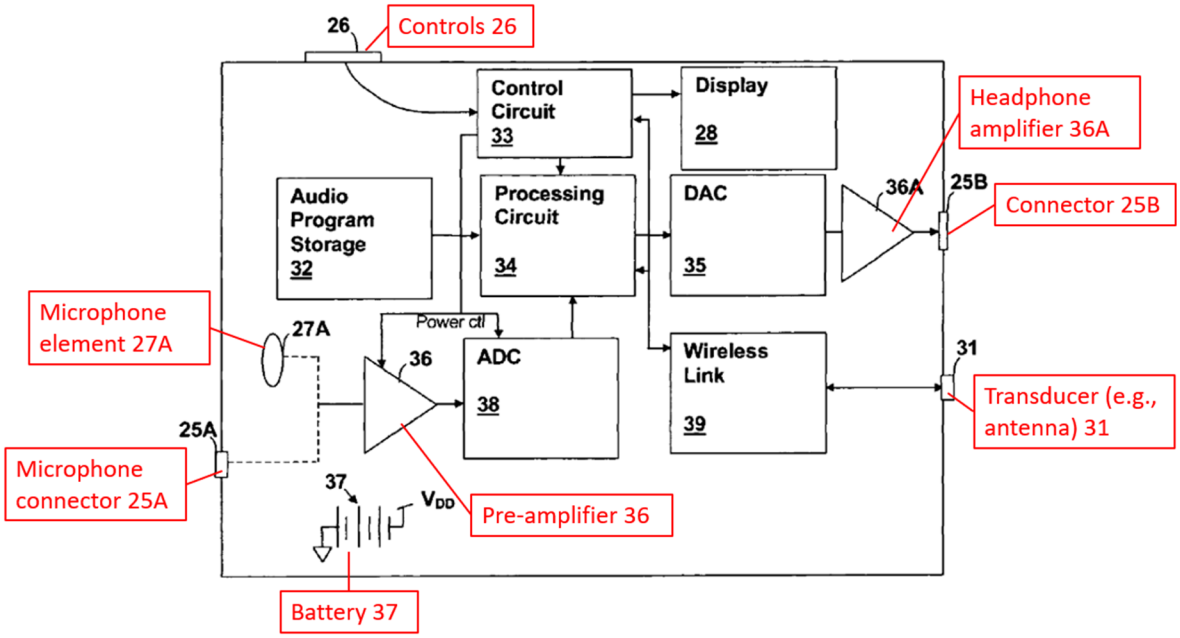
information to the wearer from program media stored within housing 10." *Id.*, 3:38-40. "A microphone element 17 is disposed within housing 10 and acoustically coupled through the surface of housing 10 to detect environmental sounds." *Id.*, 3:40-43. "Another output audio transducer 12A is connected to the device via a wired connection 13 extending external to housing 10 and another microphone element 17A may be included in the packaging of output audio transducer 12A and connected via wired connection 13 as well." *Id.*, 3:47-52.



EX1017, FIG. 1 (annotated)

131. *Melanson* Figure 3 shows "a block diagram of electrical circuits within the above-described consumer personal audio playback devices." *Id.*, 4:31-33. "Processing circuit 34 performs gain control and rendering of the

program information retrieved from storage 32 and provides the result to a digital-to-analog converter (DAC) 35 that converts the processed program information to analog form that is provided to a headphone amplifier 36A connected to headphones via a connector 25B or via a direct connection as, shown above with respect to the device of FIG. 1." *Id.*, 4:41-48. "Microphone element 27A and/or microphone connector 25A provides one or more electrical signals corresponding to environmental sounds that are amplified by preamplifier 36 and converted via an analog-to-digital converter (ADC) 38 to digital information provided to processing circuit 34." *Id.*, 4:49-53. "Processing circuit 34 is generally a digital signal processor with data and program memory, as well, and may be the same processor as control circuit 33." *Id.*, 5:21-24.



Melanson, FIG. 3 (annotated)

132. *Melanson* discloses that "[t]he processing circuit may have a selectable operating mode tailored to control the gain of the program material according to a variety of algorithms, depending on the expected environment." *Id.*, 2:31-34. "Most of the … modes are … not mutually exclusive and therefore can be engaged at one time. While compression mode will generally be single or multi-band, the other modes can be implemented simultaneously in a single device and each may be selectively enabled without interfering with the others." *Id.*, 8:32-39. For example:

- **Noise-cancellation mode.** "The device may … include subtractive noise-cancellation as a selectable operating mode and, if so, the microphones used to provide input to the noise-cancellation process may be used to provide input to the processing circuit of the present invention." *Id.*, 3:25-29.
- **Compressor mode.** "As mentioned above, there are multiple operating modes that the above-described consumer personal audio devices can enter, and the particular algorithm used to control the gain of the audio program material will vary depending on the selected mode. In the most basic 'compressor' mode, selected to overcome environmental sounds in general, the gain of low levels of program audio is adjusted upward in conformity with the detected loudness of the external sounds. Processing

circuit 34 acts as an electronic volume control that increases the gain of the program audio as ambient noise increases. However, the gain of higher levels of program audio are not increased linearly, but are adjusted according to a 'soft knee' compression curve that causes less incremental increase in program material for larger increases in ambient noise as higher levels of audio are reached." *Id.*, 5:41-55.

- **Multi-Band Compressor Mode.** "A multi-band compressor mode, which may be selectable or may be the only expander mode implemented within the personal device, provides more hearing protection while overcoming environmental noise by analyzing the microphone signal provided by ADC 38, to detect the loudness of the environmental noise in multiple frequency bands. The detected levels of loudness are used to control compression of the program material gain in multiple frequency bands by splitting the program material into corresponding frequency bands and independently controlling the gain of the signal provided in each band to DAC 35. The loudness of the program material in each frequency band can also be detected, and a combination of the loudness of the program material and the detected environmental noise level in each band can be used to control the gain of the signal supplied to DAC 35 in each frequency band." *Id.*, 5:65-6:14.

- **Personal Safety Mode.** "Another useful mode that is implemented by a signal processing algorithm according to program instructions stored within a memory of processing circuit 34 or a dedicated circuit is a personal safety mode. In personal safety mode, sudden changes in volume, or particular frequency patterns such as vehicle horns or loud voices, are detected and processing circuit 34 either partially or completely attenuates the program material gain, so that a user of the personal audio playback device will hear the outside sounds. The microphone input can be provided at the headphone output and/or mixed with the program audio, so that the user is made more aware of the external environment while the sound is present, and optionally for some time thereafter according to a timer and timing value implemented by processing circuit 34." *Id.*, 7:10-24.
- **Social Mode.** "Similar to the personal safety mode, in social mode, also implemented by a signal processing algorithm according to program instructions stored within a memory of processing circuit 34 or a dedicated circuit, speech of another person is detected and the program material is attenuated, so that a user of the personal audio playback device can converse with another person. The microphone input can be provided at the headphone output and/or mixed with the program audio, so that the

user is able to hear the other person quite well without requiring removal of headphone elements. A timer can be used to restore the program material after a period without speech by either the user or the other person is detected." *Id.*, 7:25-37.

- **Linked Chat Mode**. "Similar to conversation mode, in chat mode, also implemented by a signal processing algorithm according to program instructions stored within a memory of processing circuit 34 or a dedicated circuit, speech of the user (wearer) is detected and transmitted to another device via wireless link 39. The program material is paused or attenuated for both devices until a timer has expired and the other device can transmit audio information directly from the other device's microphone circuits to the device of FIG. 3, so that the other device-user's speech is sent to headphone amplifier 36A and connected headphone elements." *Id.*, 7:50-60.

133. *Melanson* explains how multiple microphones may be used in the modes: "[i]n each of the above modes where the character of external environmental sounds is being determined, the use of multiple microphones, such as microphone elements 17 and 17A, provides for the possibility of determining the direction of the environmental sound source. In such an implementation, it is possible to determine whether a person speaking or other

sound is coming from the space directly in front of the user/wearer and selection of activation of such modes as social mode or linked chat mode made in conformity with the determination. Also, the mode can be initiated in this manner, and then subsequently maintained, so that two persons can walk astride while the mode is continued." *Id.*, 8:10-22.

134. Figure 4, copied below, shows the operation of the playback device. *Id.*, 8:40-42. "The acoustic environment is monitored either periodically or continuously as described above (step 50)." *Id.*, 8:42-43. "If an event responsive mode is selected (decision 57), then the device will respond to an ambient audio or wireless event by attenuating, pausing or discontinuing playback of the program material and replacing it in the audio output with event-related material (step 58). For example, in linked chat mode the program material is replaced with the wireless received speech, in social and personal safety modes, the program material is replaced with the sounds received by the device microphone(s)." *Id.*, 8:53-60. "Until playback is complete, or the device is disabled (decision 60), the ambient audio environment is continually monitored and the above-described steps repeated." *Id.*, 8:62-64.

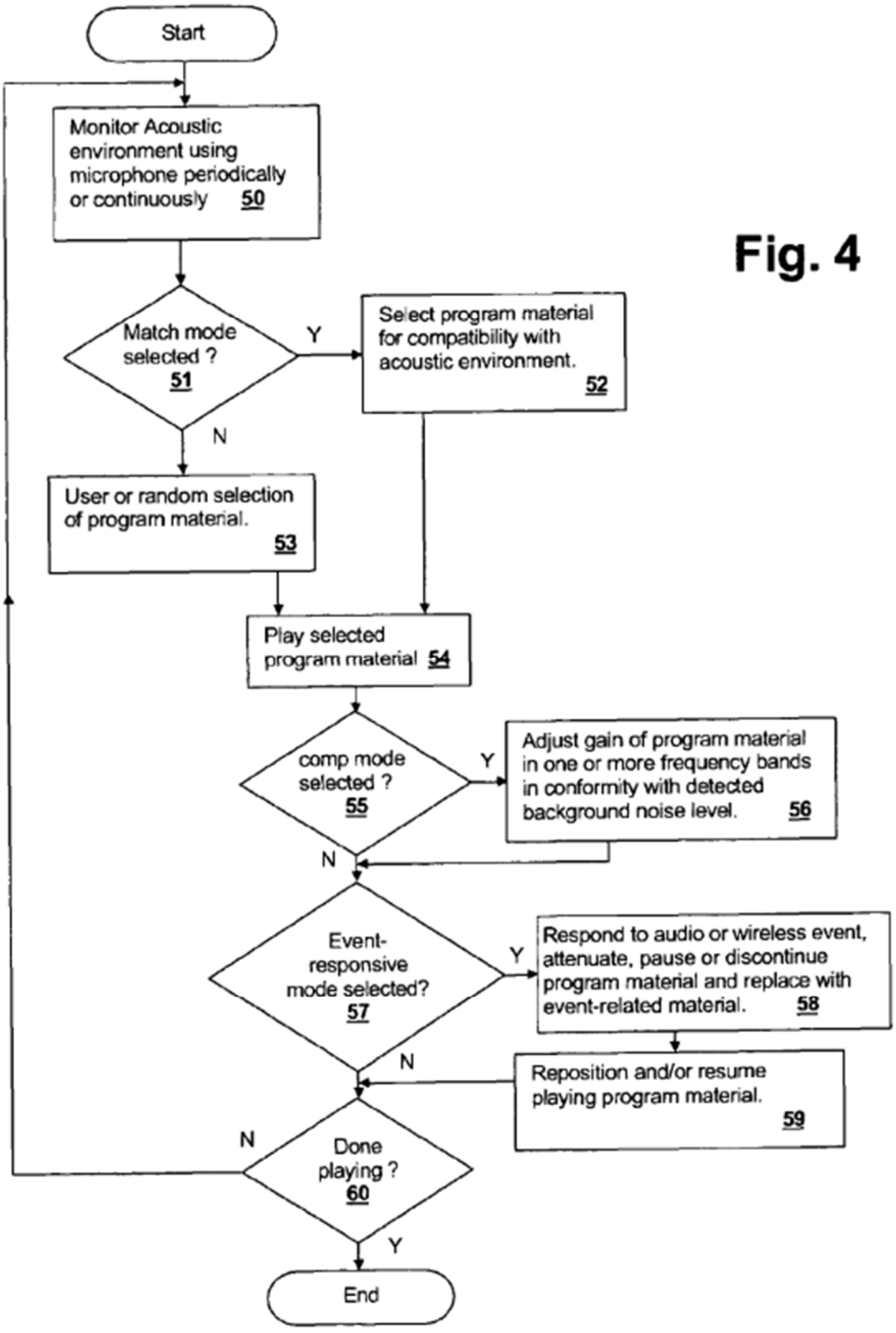


Fig. 4

Melanson, FIG. 4

F. Rosenberg

135. *Rosenberg* is titled "Ambient sound responsive media player." (Cover). *Rosenberg* was filed on April 7, 2007 and claims priority to several provisional applications including provisional application number 60/841,990 ("'990PROV") filed on August 31, 2006. Petitioners' counsel has informed me that a published nonprovisional application is entitled to claim priority to the filing date of its provisional application (as opposed to the filing date of the published non-provisional application) if (1) the provisional application discloses the portions of the published non-provisional application relied upon as prior art, and (2) at least one claim of the published non-provisional claim is supported by the provisional application, i.e., the provisional application must disclose the subject matter of the claim and enable a person of ordinary skill in the art to make and use the claim.

136. Thus, I understand that *Rosenberg* is entitled to claim priority to the August 31, 2006 filing date of its provisional application if the provisional application discloses the portions of *Rosenberg* that I rely upon and if at least one claim of *Rosenberg* is supported by the provisional application, i.e., the provisional application discloses the subject matter of one claim of *Rosenberg* and enables a person of ordinary skill in the art to make and use the claim.

(19) **United States**

(12) **Patent Application Publication** (10) **Pub. No.: US 2007/0189544 A1**

Rosenberg (43) **Pub. Date: Aug. 16, 2007**

(54) **AMBIENT SOUND RESPONSIVE MEDIA PLAYER**

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(73) Assignee: **Outland Research, LLC**, Pismo Beach, CA (US)

(21) Appl. No.: **11/697,706**

(22) Filed: **Apr. 7, 2007**

Related U.S. Application Data

(63) Continuation-in-part of application No. 11/267,079, filed on Nov. 3, 2005.
Continuation-in-part of application No. 11/223,368, filed on Sep. 9, 2005.
Continuation-in-part of application No. 11/610,615, filed on Dec. 14, 2006.

(60) Provisional application No. 60/665,291, filed on Mar. 26, 2005. Provisional application No. 60/648,197, filed on Jan. 27, 2005. Provisional application No. 60/644,417, filed on Jan. 15, 2005. Provisional application No. 60/793,214, filed on Apr. 19, 2006. Provisional application No. 60/841,990, filed on Aug. 31, 2006.

Publication Classification

(51) **Int. Cl.**
H03G 3/20 (2006.01)
H03G 3/00 (2006.01)

(52) **U.S. Cl.** **381/57**; 381/104

(57) **ABSTRACT**

Some embodiments of the present invention provide a method of adjusting an output of a media player comprising capturing an ambient audio signal; processing the ambient audio signal to determine whether one or more characteristic forms are present within the ambient audio signal; and reducing an output of a media player from a first volume to a second volume if the one or more characteristic forms are present within the ambient audio signal. The characteristic forms may be, for example, a name or personal identifier of a user of the media player, the voice of a user of the media player, or an alarm or siren.

Rosenberg at Cover

137. In my opinion, *Rosenberg* is entitled to the August 31, 2006 filing date of its provisional application because both of the above requirements are met. The table below shows claim 1 of *Rosenberg* and the corresponding exemplary disclosures of *Rosenberg*'s provisional application that disclose and enable each limitation of claim 1.

Rosenberg Claim 1	'990PROV
A method of adjusting an output of a media player comprising:	¶[0012] ("Referring to Figure 1, a generalized ***block diagram of a media player 100*** is depicted. The media player 100 includes a communications infrastructure 90 used to transfer data, memory addresses where data

capturing an ambient audio signal;	items are to be found and control signals among the various components and subsystems of the media player 100.") ¶[0028] ("Referring to Figure 2, shown is an example flow chart of an example program of an example embodiment of the present invention. … The program flow shown is generally performed, at leaset [*sic*] in part, by at least a portion of at least one *ambient sound responsive volume adjustment program* operatively loaded into main memory 10. In the particular embodiment shown herein, the entire program flow shown is performed by the at least one ambient sound responsive volume adjustment program operatively loaded into main memory 10. At the time in which the program flow begins, the media player has already selected and begun to play a media file through a separate process (not shown).") (emphasis in original) ¶[0029] ("In the particular flow shown in Figure 2, the first step in the continuous loop is step 201 ***wherein ambient audio signals are captured through microphone 95A. This ambient audio signals are generally captured as analog signals from the microphone element and then are digitized through an analog to digital conversion process***.")
processing the ambient audio signal to determine whether one or more characteristic forms are present within the ambient audio signal; and	¶¶[0030]-[0032] ("***The process then proceeds to step 202 wherein additional signal processing is performed on the captured ambient signal. This signal processing may include sound recognition processing, speech recognition processing, and/or vocal identity recognition processing steps and/or sub-steps.*** Because sound recognition, speech recognition, and/or vocal identity recognition processes are known to the prior art the specifics of such processes will not be described in detail herein. For example, US Patent 4,054,749 and US Patent 6,298,323 each of which are hereby incorporated by reference, both disclose methods and apparatus for voice identity recognition wherein a particular user's voice may be identified as being present within an audio

	signal within certain accuracy limits. Similarly, US Patent 6,804,643 which is hereby incorporated by reference, discloses a speech recognition system in which particular verbal utterances may be identified from within an audio signal, the particular verbal utterances including particular words, phrases, names, and other verbal constructs. Similarly, other pieces of art disclose methods and systems by which particular non-verbal sounds may be identified within an ambient sound signal. One example of such sound recognition methods is disclosed in HABITAT TELEMONITORING SYSTEM BASED ON THE SOUND SURVEILLANCE by Castelli, Vacher, Istrate, Besacier, and Serignat which is hereby incorporated by reference. Another example of such sound recognition methods is disclosed in a 1999 doctoral dissertation from MIT by Keith Dana Martin entitled SoundSource Recognition: A Theory and Computational Model which is hereby incorporated by reference. Another example of such sound recognition methods is disclosed by Michael Casey in the IEEE 1RANSACTIONS ON CIRCUITS AND SYSTEMS FOR VIDEO TECHNOLOGY, VOL. 11, NO. 6, JUNE 2001 in a paper entitled, MPEG-7 Sound-Recognition Tools which is hereby incorporated by reference. In such papers it is explained that recent advances in pattern recognition methodologies make the automatic identification of characteristic environmental sounds, animal sounds, non-verbal human utterances, and other non-verbal environmental sounds possible. Using such techniques, for example, alarm sound and/or siren sounds may be identified from within an ambient audio signal. Thus by using the prior art methods of speech recognition, voice identity recognition, and environmental sound identification, the ambient sound signal captured by microphone 95A and stored in local memory, m***ay be processed such that (A) the utterance of the media player user's name may be identified if substantially present within the captured ambient***

	audio signal, (B) the unique voice of the media player user may be identified if substantially present within the captured ambient audio signal, and/or (C) the sound of an alarm and/or siren and/or other similar emergency related alert sound may be identified if substantially present within the captured ambient audio signal. To perform such identifications, processing is performed in step 202. Note - in general this step is performed upon a certain time-sample's worth of ambient audio signal during each loop of the program flow. Also in general the time-samples generally proceed as overlapping time windows with each loop of the program flow. ***The process then proceeds to step 203 wherein a set of conditional routines are performed based upon whether or not a characteristic form (i.e. a signal conforming to A, B, or C above) is identified as present within the ambient signal."***)
reducing an output of a media player from a first volume to a second volume if the one or more characteristic forms are present within the ambient audio signal.	¶[0033] ("In the 'yes' branch, the process proceeds along arrow 204to step 206. At step 206, the routines of the present invention perform an Intelligent Automatic Volume Reduction routine in which the currently playing media audio signal is automatically reduced in volume so that the user can better hear the ambient sounds around him or her.") ¶[0035] ("If a characteristic form was identified within the ambient signal in step 203 and the playing volume of the media content was reduced at step 206, the process then proceeds to step 208 wherein a time delay may be optionally performed.")

138. In addition, the table below shows the portions of *Rosenberg* I rely upon in my analysis below in Grounds 2A-2C and the corresponding disclosures in *Rosenberg*'s provisional application. *Rosenberg* is therefore entitled to the filing date of its provisional application.

Rosenberg	'990PROV
[0013], [0054]-[0058], claims 19, 21	[0039]-[0043]
[0026]	[0011]
[0039]	[0024]
[0044]-[0047]	[0029]-[0032]
[0061]	[0046]

139. As background, *Rosenberg* explains that "most users of portable media players generally wear headphones to play music directly into their ears," so "users experience the beneficial effect of separating themselves from the noises of daily life, providing a serene audio environment of personally played music." *Rosenberg* ¶[0010]; *see also* EX1019 ¶[0004]. "Unfortunately, users often miss important sound events within the real world when listening to music through headphones of a portable media player. For example, another person might be talking to the media player user but because of the music playing through their headphones, the user is unable to hear the fact that they have been verbally addressed. Similarly, a siren or alarm may sound in the environment of a headphone-wearing media player user, but they may not hear the warning

sound effectively, thus creating a dangerous situation for the user. Finally, a headphone-wearing media player user may try to talk to someone else within their immediate environment, but because they cannot hear their own voice, they may find themselves talking substantially too loud for the current situation. This may create an embarrassing situation for the user." *Rosenberg* ¶[0010]; EX1019 ¶[0004].

140. In view of the above deficiencies, *Rosenberg* discloses "systematic, methodic and computer program product embodiments which provides an ambient sound responsive portable media player that enables a media player to intelligently adjust and/or vary the playing volume of a musical media file to a user based at least in part upon detected sounds from the ambient environment of the user." *Rosenberg* ¶[0021]; EX1019 ¶[0005].

141. With reference to Figure 1, *Rosenberg* discloses a generalized block diagram of a media player 100. *Rosenberg* ¶[0027]; EX1019 ¶[0012]. "A central processor 5 is provided to interpret and execute logical instructions stored in the main memory 10. The main memory 10 is the primary general purpose storage area for instructions and data to be processed by the central processor 5." *Rosenberg* ¶[0028]; EX1019 ¶[0013].

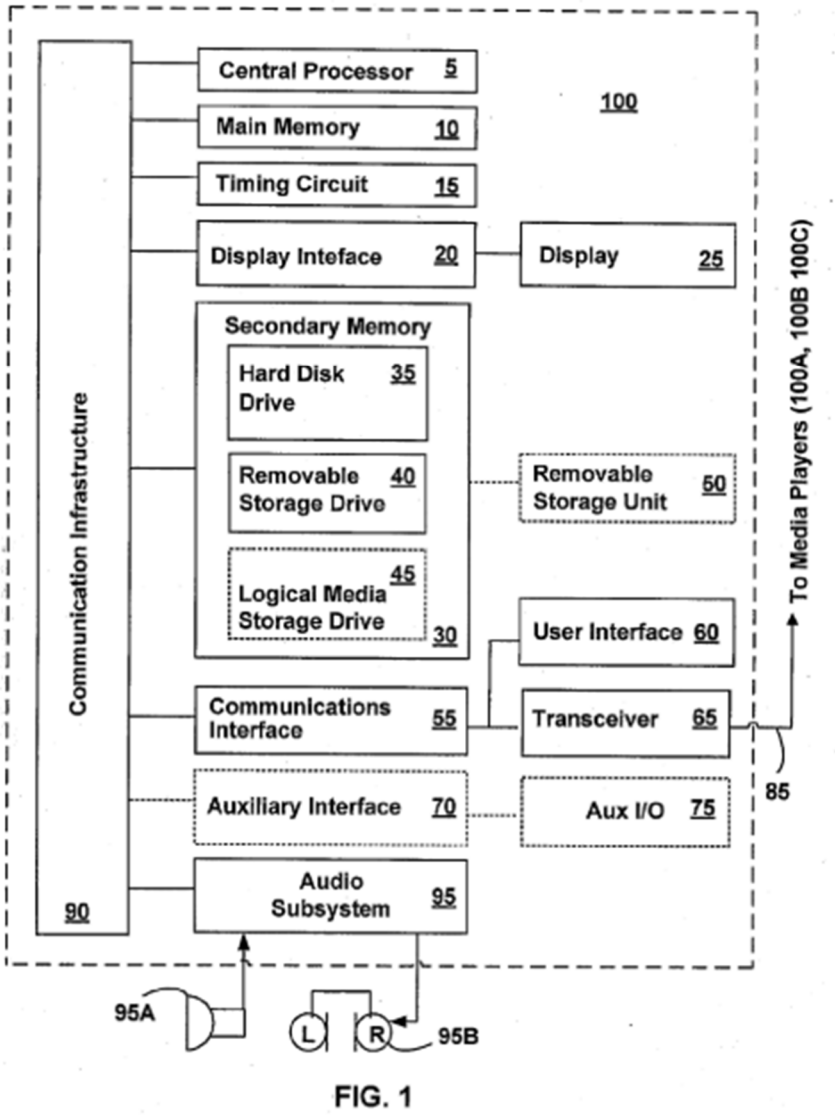


FIG. 1

Rosenberg, FIG. 1; EX1019, FIG. 1

142. "[A]n audio subsystem 95 is provided and electrically coupled to the communications infrastructure 90. The audio subsystem is configured for the playback and recording of digital media, for example, multi or multimedia encoded in any of the exemplary formats MP3, AVI, WAV, MPG, QT, WMA, AIFF, AU, RAM, RA, MOV, MIDI, etc." *Rosenberg* ¶[0038]; EX1019 ¶[0023].

"The audio subsystem includes a microphone 95A which is used for the detection of sound signals from the user's local ambient environment. The microphone 95A may be incorporated within the casing of the portable media player or may be remotely located elsewhere upon the body of the user and is connected to the media player by a wired or wireless link. Ambient sound signals from microphone 95A are generally captured as analog audio signals and converted to digital form by an analog to digital converter or other similar component and/or process. A digital signal is thereby provided to the processor of the media player, the digital signal representing the ambient audio content captured by microphone 95A. In some embodiments the microphone 95A is local to the headphones or other head-worn component of the user. In some embodiments the microphone is interfaced to the media player by a Bluetooth communication link. In some embodiments the microphone comprises a plurality of microphone elements." *Rosenberg* ¶[0039]; EX1019 ¶[0024].

143. "Referring to FIG. 2, shown is a flow chart of a process of an ambient sound responsive media player unit in accordance with some embodiments of the present invention." *Rosenberg* ¶[0043]; EX1019 ¶[0028]. "In the particular flow shown in FIG. 2, the first step in the continuous loop is step 201 wherein ambient audio signals are captured through microphone 95A. This ambient audio signals are generally captured as analog signals from the

microphone element and then are digitized through an analog to digital conversion process. In addition, noise reduction, filtering, and/or other commonly known signal processing steps may be performed upon the ambient signal." *Rosenberg* ¶[0044]; EX1019 ¶[0029].

144. "The ambient audio signals, once converted to a final digital form, are generally stored in a temporary local memory of the portable media player. It should be noted that this ambient audio signal capture step 201 may be performed by a separate process that runs in parallel with the program flow of FIG. 2. This separate process may, for example, store digitized ambient audio signal into a shared memory space that is accessible by the steps of this program flow." *Rosenberg* ¶[0044]; EX1019 ¶[0029].

145. "The process then proceeds to step 202 wherein additional signal processing is performed on the captured ambient signal. This signal processing may include sound recognition processing, speech recognition processing, and/or vocal identity recognition processing steps and/or sub-steps." *Rosenberg* ¶[0045]; EX1019 ¶[0030]. "Thus by using the prior art methods of speech recognition, voice identity recognition, and environmental sound identification, the ambient sound signal captured by microphone 95A and stored in local memory, may be processed such that (A) the utterance of the media player user's name may be identified if substantially present within the captured ambient

audio signal, (B) the unique voice of the media player user may be identified if substantially present within the captured ambient audio signal, and/or (C) the sound of an alarm and/or siren and/or other similar emergency related alert sound may be identified if substantially present within the captured ambient audio signal. To perform such identifications, processing is performed in step 202." *Rosenberg* ¶[0046]; EX1019 ¶[0031].

146. "The process then proceeds to step 203 wherein a set of conditional routines are performed based upon whether or not a characteristic form (e.g. a signal conforming to A, B, or C above) is identified as present within the ambient signal." *Rosenberg* ¶[0047]; EX1019 ¶[0032]. "In the 'yes' branch, the process proceeds along arrow 204 to step 206. At step 206, the routines of the present invention perform an Intelligent Automatic Volume Reduction routine in which the currently playing media audio signal is automatically reduced in volume so that the user can better hear the ambient sounds around him or her." *Rosenberg* ¶[0048]; EX1019 ¶[0033].

147. "In some embodiments of the present invention, the media player is operative to mix musical audio content derived from a stored media file with ambient audio content captured from a microphone local to the user." *Rosenberg* ¶[0054]; EX1019 ¶[0039]. "[A] single audio signal is presented to the user through the headphones or other similar sound display hardware, the signal

audio signal including an audio combination of a musical media file accessed from a memory of the media player and an ambient audio signal derived from the signal captured by Microphone 95A." *Rosenberg* ¶[0054]; EX1019 ¶[0039]. "It should be noted that the ambient audio signal content may be filtered or otherwise processed to extract extraneous noise and/or sound content that is outside certain magnitude and/or frequency limits or thresholds." *Rosenberg* ¶[0054]; EX1019 ¶[0039].

148. "[S]ome embodiments of the present invention include a further inventive method in which the relative volume balance of the mixed signal (i.e. the relative volume of the musical media content and the ambient microphone content) are selectively adjusted in response to detected ambient audio events." *Rosenberg* ¶[0055]; EX1019 ¶[0040]. "More specifically, the relative volume of the microphone content is automatically increased with respect to the musical media content, for a period of time, in response to detected characteristic ambient audio events within the ambient audio signal stream. The detected characteristic ambient audio events may include, but are not limited to: (A) the detection of the media player's name being uttered within the ambient audio signal, (B) the detection of the media player's own voice within the ambient audio signal, and/or (C) detection of an alarm or siren sound present within the ambient audio signal." *Rosenberg* ¶[0055]; EX1019 ¶[0040].

149. "It should also be noted that in some embodiments the nominal relative volume levels of the two signals may be set such that the volume of the ambient microphone content is substantially zero at times when an ambient audio event has not been detected. In this way the user only hears the musical content until and unless an ambient audio event is detected." *Rosenberg* ¶[0057]; EX1019 ¶[0042]. "In response to such a detected ambient audio event (for example an event such as A, B, or C above), the automatic routines of the present invention adjust the relative volumes of the two signals such that the ambient environment microphone signal is no longer zero, instead being substantial with respect to the musical media content. In this way the pure musical content is played to the user until an ambient audio event is detected, then in response to the detected event a mixed audio signal is presented with both musical content and ambient audio content such that ambient audio content is clearly audible at a substantial relative volume." *Rosenberg* ¶[0057]; EX1019 ¶[0042].

150. "Note—in some embodiments the mixed volume level is such that the musical audio content is gradually decreased down to substantially zero while the ambient audio musical content is gradually increased up to the prior music volume level. Such a cross-fade enables the music to fade out while the ambient audio content fades in. This lasts for a period of time. After the period

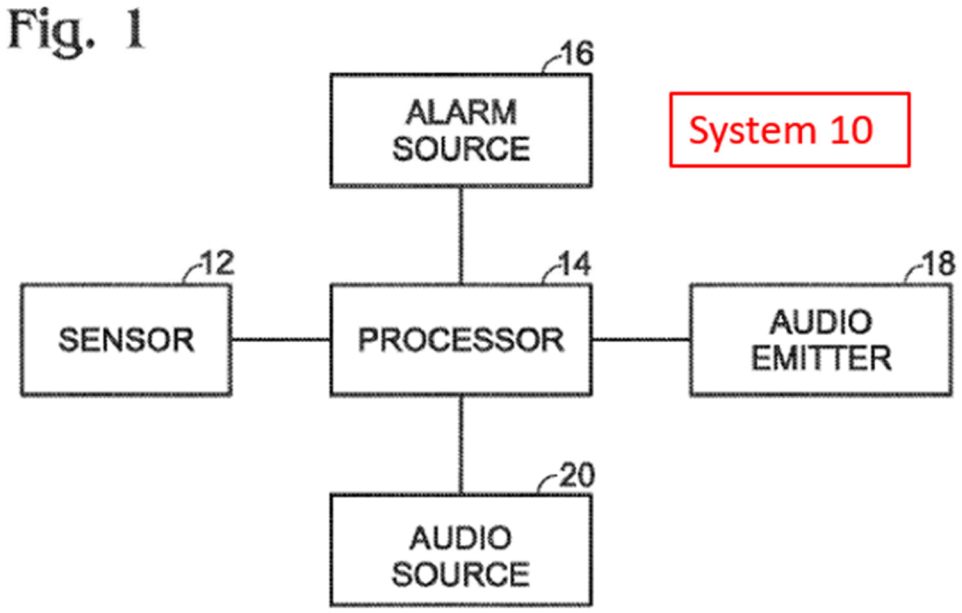
of time, the process reverses, the ambient audio content fading out to zero volume and the musical content fading back to its pre-event nominal volume." *Rosenberg* ¶[0058]; EX1019 ¶[0043].

G. Samadani

151. *Samadani* is titled "Safety Device." It discloses "Methods and systems for detecting a warning condition and alerting a headphone wearer to the warning condition." *Samadani*, Abstract. As background, *Samadani* discloses that "[a] number of safety devices have been described that detect a signal that is emitted from a potentially hazardous object (such as a construction truck) and alert a device user when the user comes within a certain proximity of the object." *Id.* ¶[0002]. "[T]hese devices sound the alarm whether or not the hazardous object is approaching the device user." *Id*. "Other safety devices have been described that detect when a certain sounds, such as a siren. However, these safety devices do not determine whether or not the source of the sound is approaching the device user." *Id.* ¶[0003].

152. *Samadani* therefore discloses a system 10 "incorporated into a headset or earphones that can be worn by a user." *Id.* ¶[0010]. "System 10 includes a sensor 12. Sensor 12 is configured to monitor the external environment surrounding the device user. According to one embodiment, sensor

12 may be or include one or more microphones configured to monitor external audio signals produced by the external environment." *Id.* ¶[0011].



Samadani, FIG. 1 (annotated)

153. "Processor 14 is configured to receive and process signals from sensor 12 in order to filter out background and other uninteresting signals and enhance signals of interest." *Id.* ¶[0013]. "It will be understood that the specific signals that will be identified as signals of interest by the processor will differ depending on the specific desired function of the system 10. For example, a system that is incorporated into an MP3 player that is intended to be worn by a jogger could enhance signals created by cars and other vehicles as signals of interest. Alternatively, or additionally, signals that match human speech may or

may not be considered signals of interest. Various exemplary methods for enhancing signals of interest are discussed below. System 10 may incorporate any suitable method for enhancing signals of interest, including, but not necessarily limited to, those methods described herein." *Id.* "According to one embodiment, processor 14 may be configured to examine the power signature of the audio signals monitored by sensor 12. If an object is producing a constant energy signature (i.e. volume), the energy signature of the sound produced by the object typically increases as the object approaches the sensor. Accordingly, processor 14 may be configured to identify those audio signals that demonstrate an increase in energy." *Id.* ¶[0024].

154. "Upon detection of a signal of interest, processor 14 may be configured to further process the signal to determine if the signal of interest fits within a given set of parameters in order to determine if a warning condition exists. For example, in the case where system 10 is configured to identify and alert the user to objects that are approaching the user, it will be understood a given signal may be created by the type of object the system was designed to identify but that the signal does not indicate that the object is approaching the user." *Id.* ¶[0014]; *see also id.* ¶[0031] ("As stated above, once a signal of interest is identified, processor 14 could apply one or more filters or other processes to determine if the identified signal of interest indicates a warning

condition. The stationarity and non-stationarity of the audio power could also be used to determine if a signal indicates a warning condition or is a false alarm.")

155. "Once the processor has determined that a warning condition exists, processor 14 is configured to provide an alert to the user." *Id*. ¶[0016]. "Processor 14 may further include or be in electronic communication with a mixer (not shown). The mixer may be any device configured to receive multiple audio signals, such as those generated by alarm source 16 and audio source 20 and mix and/or alter the audio signals when a warning condition is detected." *Id.* ¶[0018].

H. Dijkstra

156. *Dijkstra* is titled "Dynamic Hearing Assistance System And Method Therefore." As background, *Dijkstra* explains that "hearing assistance systems such as assistive listening systems (ALS) have become widely used for alleviating difficulties that people with a hearing impairment are faced with in daily life." *Dijkstra* ¶[0002]. "Frequency modulated (FM) hearing systems are one of the systems used today. These systems use the FM transmission techniques to transmit wirelessly signals from the source to the listeners." *Id.* ¶[0003]. "Most FM systems in use today provide two or three different operating modes. The choices are to get the sound from: (1) the hearing instrument microphone alone, (2) the FM microphone alone, or (3) a

combination of FM and hearing instrument microphones together." *Id.* ¶¶[0004]-[0007]. "Most of the time, the FM system is used in the FM plus hearing instrument combination (often called FM+M or FM+ENV)." *Id.* ¶[0008]. "This operating mode allows a main person speaking to have a consistent signal to the listener's ear while the integrated hearing instrument microphone also stays on so that environmental sounds can be heard." *Id.*

157. "The so-called 'FM advantage' measures the relative loudness of signals when both the FM signal and the hearing instrument microphone are active at the same time." *Id.* "Heretofore, depending on the type of hearing instrument, the output of the FM receiver is adjusted in such a way that the FM advantage is either fixed or programmable by a professional. However, the FM advantage should be determined according to the particular listening situation (quiet environment, loud background noise, lecture in the school, conference etc.). Consequently, any fixed FM advantage is only a compromise, and cannot offer an optimal result in all listening situations. The existing hearing assistance methods do not provide a solution to this problem." *Id.* ¶[0013].

158. In view of the above drawbacks in the prior art, *Dijkstra* discloses a system and method for "dynamic loudness adaptation of audio signals." *Id.* ¶[0015]. "FIG. 3 shows a diagram of a segment of a particular embodiment of the hearing assistance system according to the invention." *Id.* ¶[0036]. "[T]he

hearing instrument 15 comprises a microphone 36 for capturing environmental sounds, such as own voice and/or speech from the fellow students in school classes." *Id.* ¶[0037]. "The hearing instrument 15 further comprises a loudspeaker 38 that reproduces audio signals to the listener's ear 39." *Id.* "the audio signals proceeding from the microphone 36 are reproduced together with the audio signals proceeding from the remote speaker and received over the communication channel 27 by means of the receiving unit 24." *Id*. "The hearing instrument can contain various modules and units dedicated to signal processing and in particular a processing module 33 that processes signals received from the receiving unit 24, a classification unit 34 for determining the classification index based on the amplitude and/or frequency and/or temporal characteristics of the audio signals, and a central unit 35 for adaptation of the predefined amplitude and/or frequency ratio of the first audio signals relative to the second audio signals, based on the classification index." *Id*.

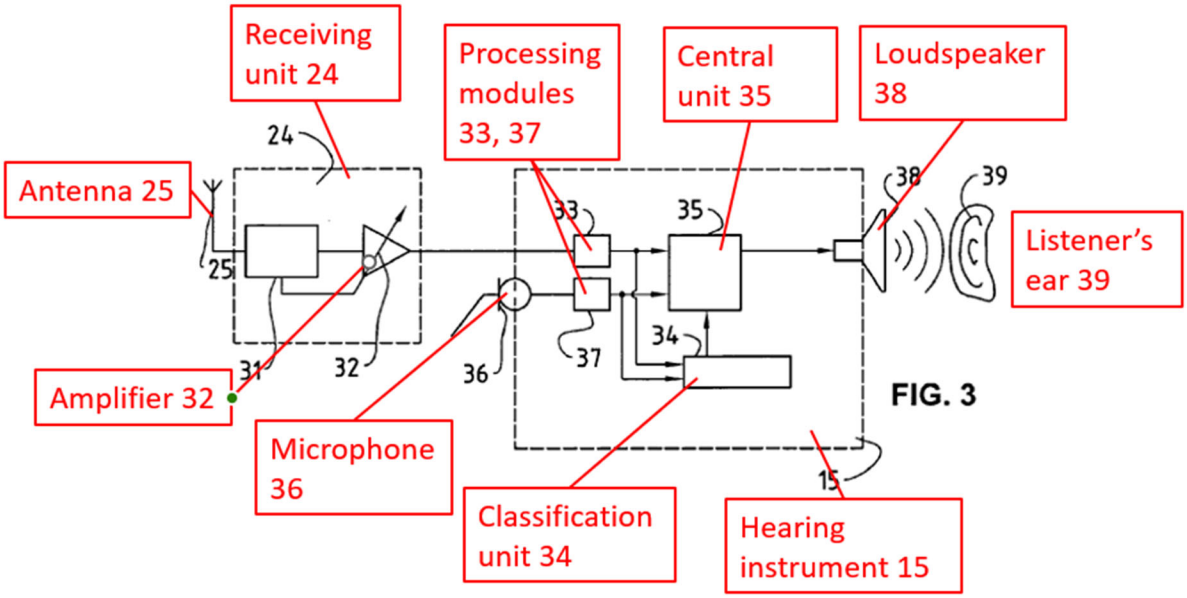


FIG. 3

159. "Classification unit 34 performs the determination of the classification index based on the amplitude and/or frequency and/or temporal characteristics of the audio signals received through the radio communication channel 27 from the remote speaker 11 and/or the audio signals corresponding to the environmental sounds received from the microphone 36." *Id.* ¶[0039]. "The determination of the classification index can be based on temporal and/or spectral analysis of the signals, as well as on any other analysis method, including auditory scene analysis as performed by many hearing instruments. The system may be implemented in such a way that the classification index only takes a limited number of predefined values, corresponding for instance to the most common and most typical auditory situations: quiet environment without

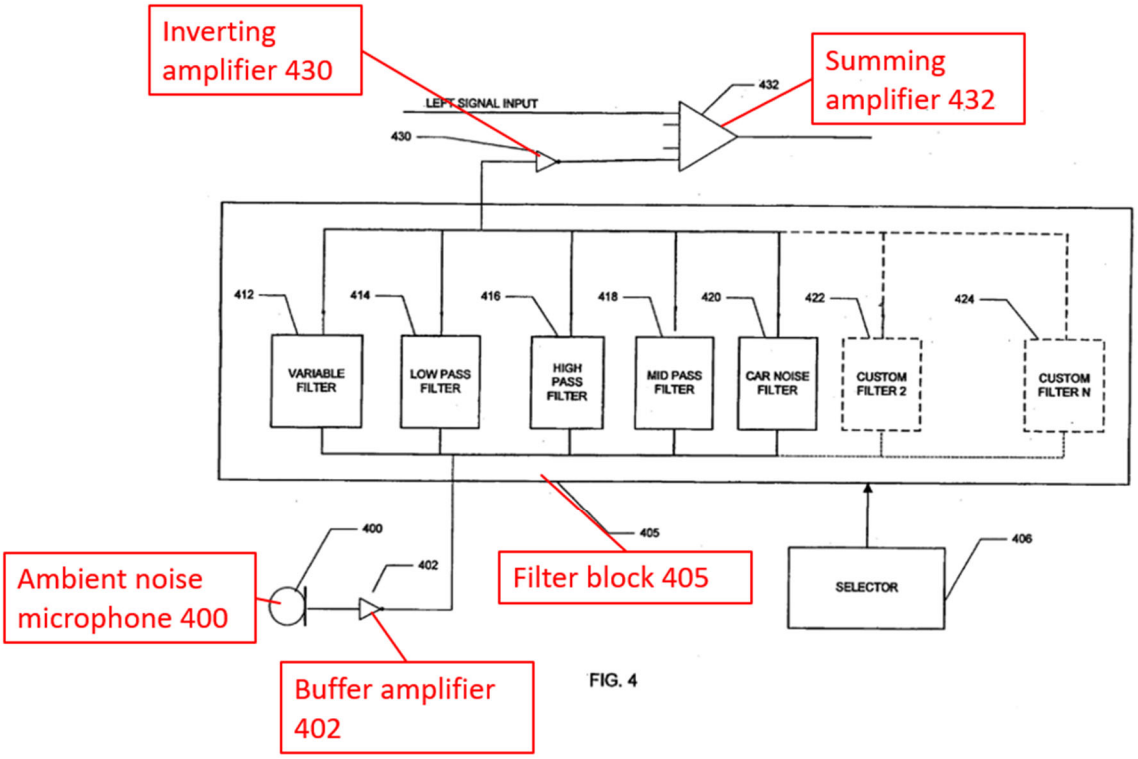
background noise, loud background noise, one single speaker, own voice, a multitude of speakers etc." *Id*.

160. "Once determined, the value of the classification index is used by the central unit 35 to adapt correspondingly the ratio of amplitudes and/or frequencies of the audio signals proceeding from the remote speaker 11 and audio signals corresponding to the environmental sound, adjusting in this way the FM advantage in the hearing instrument 15." *Id*. "The adapted audio signals are then transmitted to the loudspeaker 38 or any other adequate reproduction means and output to the listener's 12 ear 39." *Id*.

I. Engle

161. *Engle* is titled "Method and Apparatus for Controlling a Headphone." EX1022, Cover. As background, *Engle* notes that a "problem of a conventional noise reduction headphone is that fixed noise reduction filters are used, which limits the amount of noise reduction available, and does not allow a user to vary the noise canceling characteristics based on the external noise." *Id.*, 2:11-14. "For example, a user may wish that certain types of noise, such as voices or emergency sirens, not be canceled. The types of noise that should be canceled, and those that should not be canceled, may vary depending on the environment where the headphones are worn." *Id.*, 2:14-18.

162. With reference to Figure 4, *Engle* discloses a headphone with "active noise reduction including active noise reduction filters applicable to one or more predefined frequency bands selectable by a user, and filters with frequency bands and/or gains selectable by a user." *Id.*, 16:5-9



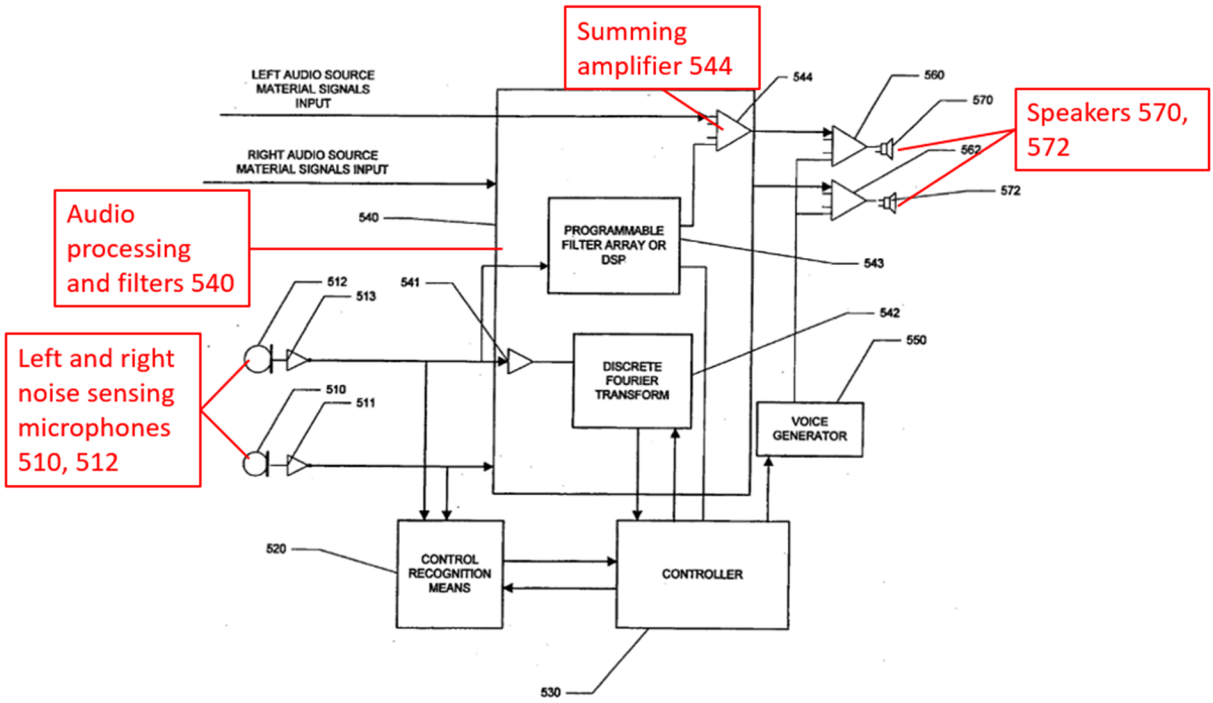
EX1022, FIG. 4 (annotated)

163. "[A]mbient noise from a microphone 400 is buffered and amplified by a buffer amplifier 402. The output from the buffer amplifier is fed to one of the filters in filter block 405, as selected by the user employing selector 406." *Id.*, 16:10-13. "The output signal from a selected filter is inverted and amplified by an inverting amplifier 430. The output of the inverted noise signal is added to

an input audio signal by a summing amplifier 432 and the sum signal is then fed to a speaker (not shown) in the headphone, so that the output signal from the speaker can cancel or reduce the ambient noise present at an output of the headphone." *Id.*, 16:16-22. "Switchable frequency bands, and high pass, low pass, and mid pass filters, as shown in Figure 4, could be independently switched in. Two or more such bands may be activated at the same time. Variable filter 412 may include variable frequency, variable gain, and variable bandwidth, each of which parameters may be user-defined. These features permit a user great flexibility to cancel out a specific band of noise that is a problem." *Id.*, 17:1-7. "The custom filters can be specially tuned filters for other environments, such as the cabin of a jet airplane, similarly configured passenger cabins of propeller planes, helicopter environment and associated noise, noises of motors found in a home, office or other building, such as air conditioner noise (e.g., window unit), vacuum cleaner, and industrial environments, such as warehouses and factory environments." *Id.*, 17:17-21.

164. "Figure 5 illustrates another embodiment of the invention, where a digital signal processing (DSP) system is added to characterize the received noise, for example using spectral and amplitude information obtained through a spectral analysis, and program a programmable filter array or a digital signal processor according to the characterization. In this way, noise cancellation or

reduction can be dynamically optimized during transient conditions, such as passing vehicles, trees, trains, or aircraft." *Id.*, 18:3-9.



EX1022, FIG. 5 (annotated)

165. "Ambient noise is received at left and right noise sensing microphones 510, 512 … The received ambient noise signals are provided to audio processing and filters 540, and are converted into digital signal … The digitized noise signal is discrete Fourier transformed at block 542 to obtain the spectra and amplitudes of the ambient noise." *Id.*, 18:12-19.

166. "The spectrum and amplitude information is sent to the controller 530, which stores information in a memory on those bands in the ambient noise that should not be cancelled. For example, bands corresponding to human speech

and emergency sirens should not be cancelled. Based on the received spectrum and amplitude information and the information in the memory, the controller 530 determines one or more bands of ambient noise to be cancelled or reduced, and instructs the programmable filter array or DSP system 543 to construct filters to provide active noise cancellation for those bands that should be cancelled or reduced." *Id.*, 18:19-19:6.

167. "The DSP system 543 performing the construction of the filter may be a programmable filter array or a digital signal processor. DSP system 543 may also invert the filtered passed signal and output the inverted signal to a summing amplifier 544. Another input of summing amplifier 544 is coupled to left audio source materials input. Summing amplifier 544 sums the two received signals and the sum signal is digital to analog converted and input to one of buffer amplifiers 560, 562. Buffer amplifiers 560, 562, also receive signals from voice generator 550, which is similar to voice generator 150 of Fig. 1. Buffer amplifiers 560, 562 sum the received signals and provide an amplified output signal used to drive speakers 570, 572." *Id.*, 19:6-16.

IX. INTERPRETATION OF THE '473 PATENT CLAIMS

168. I have given all the claim terms their plan and ordinary meaning, as would have been understood by a person having ordinary skill in the art, at the

time of the alleged invention, having taken into consideration the language of the claims, the specification, and the prosecution history of record.

X. SPECIFIC GROUNDS FOR CHALLENGE

A. GROUND 1A: *PATEL* IN COMBINATION WITH *TOPHOLM* RENDERS OBVIOUS CLAIMS 1 and 3-7

169. At the outset, I note that *Patel* describes various embodiments of a head-mounted device and its components. A POSITA would have understood that (1) *Patel*'s embodiments build on each other and have components in common, and (2) that *Patel* does not repeat the description of the structure and functionality of all common components for each embodiment. For analytical clarity, my opinion focuses on *Patel*'s embodiment in Figure 9. My understanding of how a POSITA would have understood the structure and function of the embodiment in Figure 9 is informed by the description of that Figure, but also the description of components and functionality of the embodiment in Figure 9 that *Patel* describes in connection with *Patel*'s other embodiments.

1. Claim 1

a. *[1.pre]: A device that can control ambient passthrough in an earpiece comprising:*

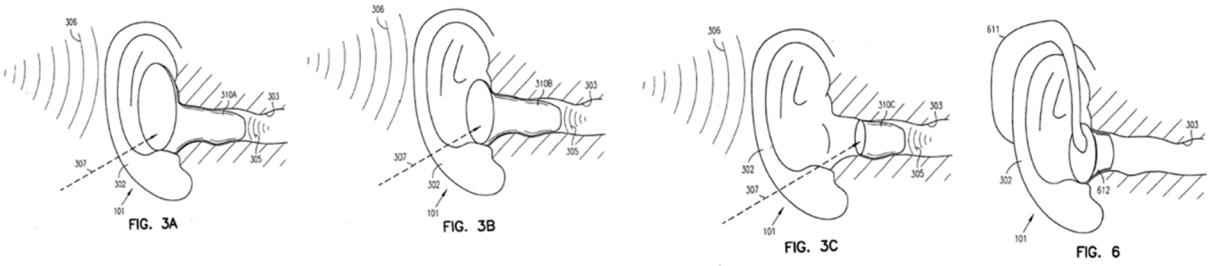
170. To the extent the preamble is limiting, *Patel* discloses or suggests this feature.

171. *Patel* discloses a "head-mounted device" that establishes a wireless link with an "external device." *Patel* ¶¶[0001], [0018] ("The illustrated wireless head-mounted device may, for example, be used to establish a wireless link (e.g., a Bluetooth link) with an external device, such as a cell phone or PDA, in order to send and receive audio signals."), [0060] (explaining that a computer or other devices such as an MP3 player or other mobile device in communication with the head-mounted device can be used to program the head-mounted device).

172. A POSITA would have understood that the description of the head-mounted device in paragraphs [00018]-[0020] regarding Figure 1 apply to the embodiment in Figure 9 because, for example, both Figures 1 and 2 have a Bluetooth radio ("radio 2" in Figure 1 and "radio 72" in Figure 9). Thus, a POSITA would have understood that the head-mounted device illustrated in Figure 9 communicates with an external device.

173. *Patel* discloses that the head-mounted device "could rest behind a single ear … or it could rest within the ear canal like a hearing instrument." *Id*. ¶[0062]. This is consistent with how the '473 patent describes an "earpiece." *'473pat*., 4:64-66 ("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."). Therefore, a POSITA would have understood *Patel* to disclose an "earpiece."

174. *Patel* explains that the head-mounted device "could rest behind a single ear … or it could rest within the ear canal like a hearing instrument" in paragraph 62 in the context of describing Figures 10 and 11, but a POSITA would have understood that *Patel*'s description in this regard applies to the "head-mounted device" described in *Patel*'s other disclosed embodiments. This is consistent, for example, with *Patel*'s other disclosures that the head-mounted device may be used as a "wireless headset" and "hearing aid," as a POSITA would have understood that such devices could be implemented in a device that rests behind an ear or within the ear canal. *Patel* ¶[0001]. This is corroborated, for example, by EP1519625A2 ("*Victorian*"), which illustrates various known form-factors for an "ear-level communication system" including "wireless headsets," such as "in-the-ear," "completely-in-the-canal," and "behind-the-ear" devices. EX1033 (*Victorian*) ¶¶[0003], [0013], FIGs. 3A-3C, 6.[2]



Victorian, FIGs. 3A, 3B, 3C, 6

[2] To be clear, I am relying on *Victorian* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

175. *Patel* further discloses that the head-mounted device includes a microphone 27 (*id.* ¶[0026]) and that "[e]nvironmental sounds (i.e., audio signals from outside of the ear canal) may be received by the microphone 27 and introduced by the signal processor 28 into the audio signal being transmitted into the ear canal." *Id.* ¶[0028].

176. Although microphone 27 is described in Figure 4, a POSITA would have understood that the description of microphone 27 applies to microphone 77 in Figure 9 because of the similar numerals and illustrated configuration in Figures 4 and 9.

177. A POSITA would have understood that "environmental sounds" are a type of "ambient sound." A POSITA would have understood the signal processor introducing environmental sounds into the audio signal provided to the ear canal to comprise the signal processor passing the environmental sounds through speaker 24 into the user's ear canal, which is consistent with how the '473 patent describes "ambient passthrough." *'473pat.*, 7:28-31 ("If at step 310, an acute sound is detected within the ambient sound, the earpiece 100 activates 'sound pass-through' to reproduce the ambient sound in the ear canal by way of the ECR 125."), 8:51-53 ("The pass-through mode permits ambient sound detected at the ASM 111 to be transmitted directly to the ear canal.").

178. *Patel* further explains that, "[b]y electronic (e.g., a control signal sent by a wireless or direct link) or manual means via the control input to the digital signal processor 28, the user may turn down or turn off the environmental sounds, for example when the head-mounted device is in a communication mode (e.g., when a cellular call is initiated or in progress.)." *Patel* ¶[0028].

179. A POSITA would have understood that *Patel*'s description of microphone 27 and the provision of environmental sounds in paragraphs [0026]-[0030] regarding Figure 4 apply to microphone 77 and the embodiment in Figure 9. In addition to the like numerals, a POSITA would have recognized that the structural similarities between microphones, including that both microphone 27 and microphone 77 include a connection to the digital signal processing block in the receiving path, shown in the annotated image below.

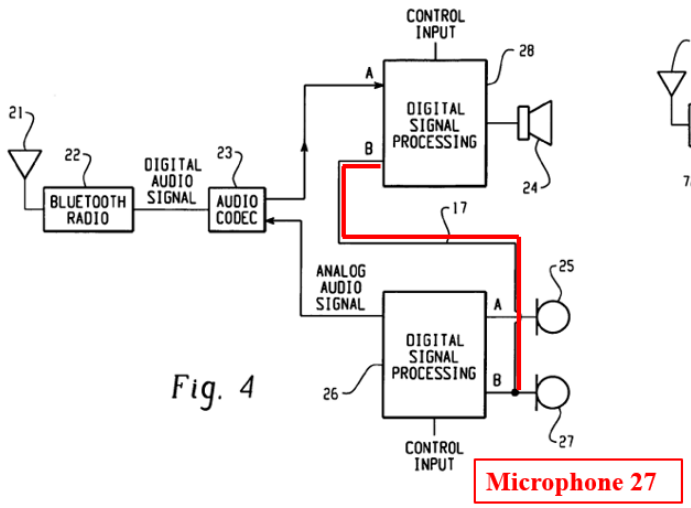


Fig. 4

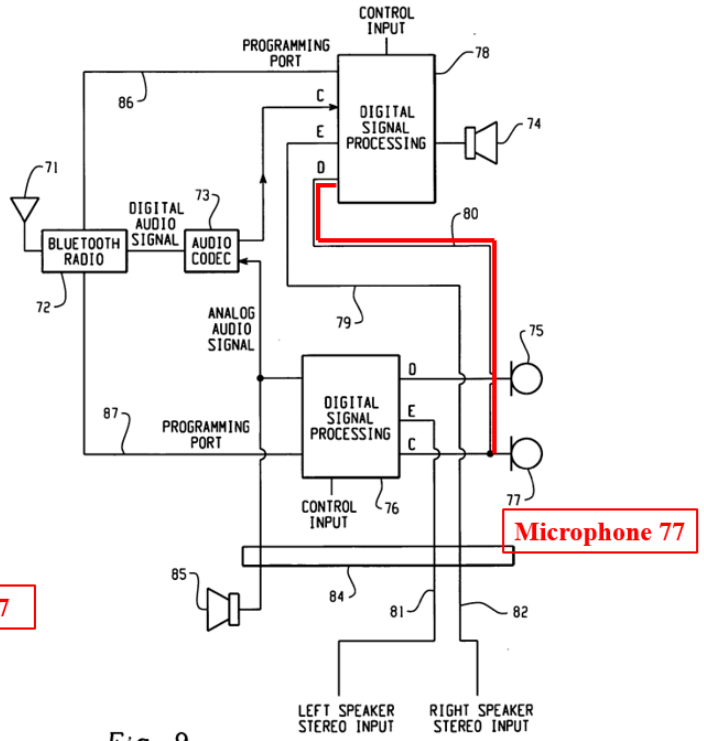


Fig. 9

Patel, FIGs. 4, 9 (annotated)

180. *Patel* discloses that the "control input" may be received "via the radio 2." *Id*. ¶[0025]. Because *Patel* discloses that the radio 2 is used for the wireless link between head-mounted device and the external device (*id*. ¶¶[0018]-[0019]), a POSITA would have understood a "control signal" sent by a "wireless … link" to comprise a control signal sent by the external device. *See, e.g.*, *id.* ¶¶[0002], [0018]-[0019]; *see also id.* ¶[0032] ("In one example, the signal processor 32 may receive a control signal from an external device (e.g., a cellular telephone) via the radio communications circuitry 42 ….").

181. A POSITA would have understood that the description of digital signal processor in paragraph [0025] applies to both digital signal processing blocks in Figure 9 because, for example, both digital signal processing blocks include a control input and a connection to radio 2.

182. Accordingly, a POSITA would have understood *Patel* to disclose or suggest an external device ("a device") wirelessly linked to a head-mounted device to control the volume of environmental sounds introduced into the audio signal provided to the user ("that can control ambient passthrough") wherein the head-mounted device rests behind a single ear or within the ear canal ("in an earpiece").

b. *[1.1]: a first memory that stores instructions; and [1.2] a processor that executes the instructions to perform operations, the operations comprising*

183. *Patel* in combination with *Topholm* discloses or suggests this feature.

184. As discussed in §X.A.1.a, *Patel* discloses or suggests an external device for controlling ambient sound passthrough in an earpiece. *Patel* discloses that "cellular phones, PDAs, radios, MP3 players, CD players, portable game machines, etc." may be used as the external device. *Patel* ¶[0018]; *see also id*. ¶¶[0059]-[0060] ("***An application running on a computer, which emulates a graphical equalizer, can be displayed on a computer screen*** and adjusted in real

time under user control. The equalizer settings may be transferred over the wireless link to the head-mounted device, where the amplitude and bandwidth settings for each filter within the filter bank of the signal processors 76, 78 are programmed via the programming ports of digital signal processing blocks 76, 78. It should be understood that other devices may also be used to program the head-mounted device equalizer settings, ***such as an MP3 player or other mobile device in wired or wireless communication with the head-mounted device***.”). This description of the external devices in paragraph 18 applies to Figure 9. *See* §X.A.1.a.

185. To the extent a POSITA required additional implementation details related to the structure of such external devices, a POSITA would have readily identified *Topholm*. *Topholm* discloses a remote control system for a hearing aid comprising a mobile telephone and a plug-in device. *Topholm*, 2:10-20.

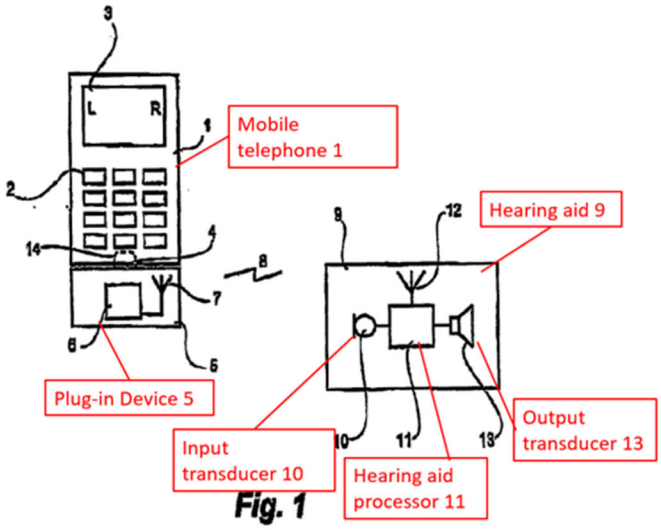


Fig. 1

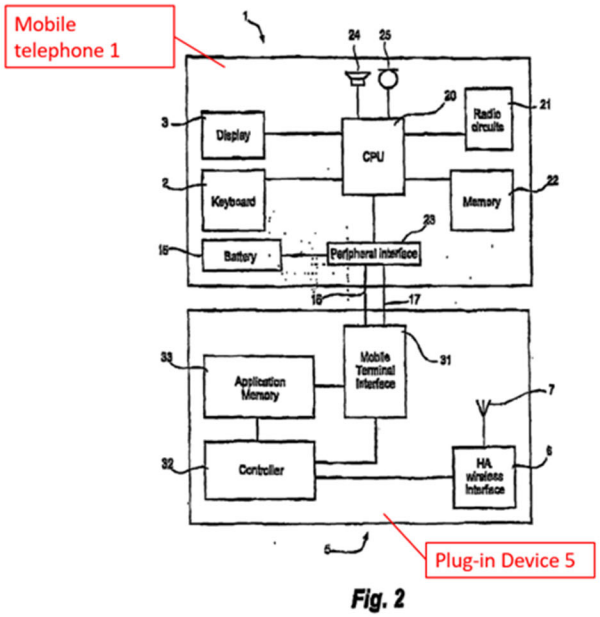


Fig. 2

Topholm, FIGs. 1, 2 (annotated)

186. The mobile telephone comprises a CPU 20 connected to a memory 22. *Topholm*, 4:6-7. The remote control functions of the mobile telephone are "controlled by the CPU 20 executing applications stored in the memory 22." *Id.*, 4:22. "The application software provides the functionality of a remote control for a hearing aid to the mobile telephone 1, e.g. displaying a remote control menu and enabling remote control input by the keys of the telephone keypad." *Id.*, 4:30-5:2. *Topholm* explains that the software "enables a mobile telephone to double as a user interface of a wireless remote control of a hearing aid." *Id.*, 2:6-8. *Topholm* also discloses that a keypad on the mobile telephone may be used to make volume adjustments in the hearing aid (*id.*, 5:4-10) and that digital audio

stored in memory 22 on the mobile telephone may be transmitted to the hearing aid (*id.*, 6:5-9).

187. A POSITA would have understood memory 22 comprises memory that stores instructions because "applications" and "software" are a series of instructions. EX1034 (*Computer dictionary*).[3] Thus, a POSITA would have understood *Topholm* to disclose or suggest a remote control system including a mobile telephone with instructions stored in memory. A POSITA would have understood that CPU 20 is a processor because a POSITA would have understood that "CPU" is a common initialism for "central processing unit," i.e., a processor. EX1035 at 39.[4] Thus, a POSITA would have understood *Topholm* to disclose that the mobile telephone includes a processor for executing the instruction stored in memory to perform the remote control operations, including making volume adjustments and transmitting audio.

188. A POSITA would have been motivated to combine *Patel* with *Topholm*'s teachings related to a mobile device with a memory and processor and would have had a reasonable expectation of success in doing so. *Patel*

[3] To be clear, I am relying on *Computer Dictionary* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

[4] To be clear, I am relying on *Tanenbaum* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

discloses using an external device to control a head-mounted devices and *Patel* describes an example remote control system including a device to control a hearing aid (a type of head-mounted device), including to control the volume settings of the hearing aid. At a minimum, a POSITA would have considered *Patel*'s disclosure to amount to an express suggestion to implement the teachings of *Topholm* to control the ambient passthrough in *Patel*'s head-mounted device. A POSITA also would have considered the combination to amount to the application of a known technique (*Topholm*'s teaching of a wireless remote control system including software stored in a memory and a processor that executes the software stored in the memory to send audio and control signals to a hearing aid) to a known device (*Patel*'s external device) ready for improvement (*Patel*'s external device is already capable of transmitting audio and control signals to a head-mounted device) to yield predictable results (*Patel*'s head-mounted device in communication with an external device including a memory and processor).

189. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest an external device with software and applications stored in memory ("a first memory that stores instructions") and a CPU ("a processor") that executes the applications and software to perform operations ("that executes the instructions to perform operations").

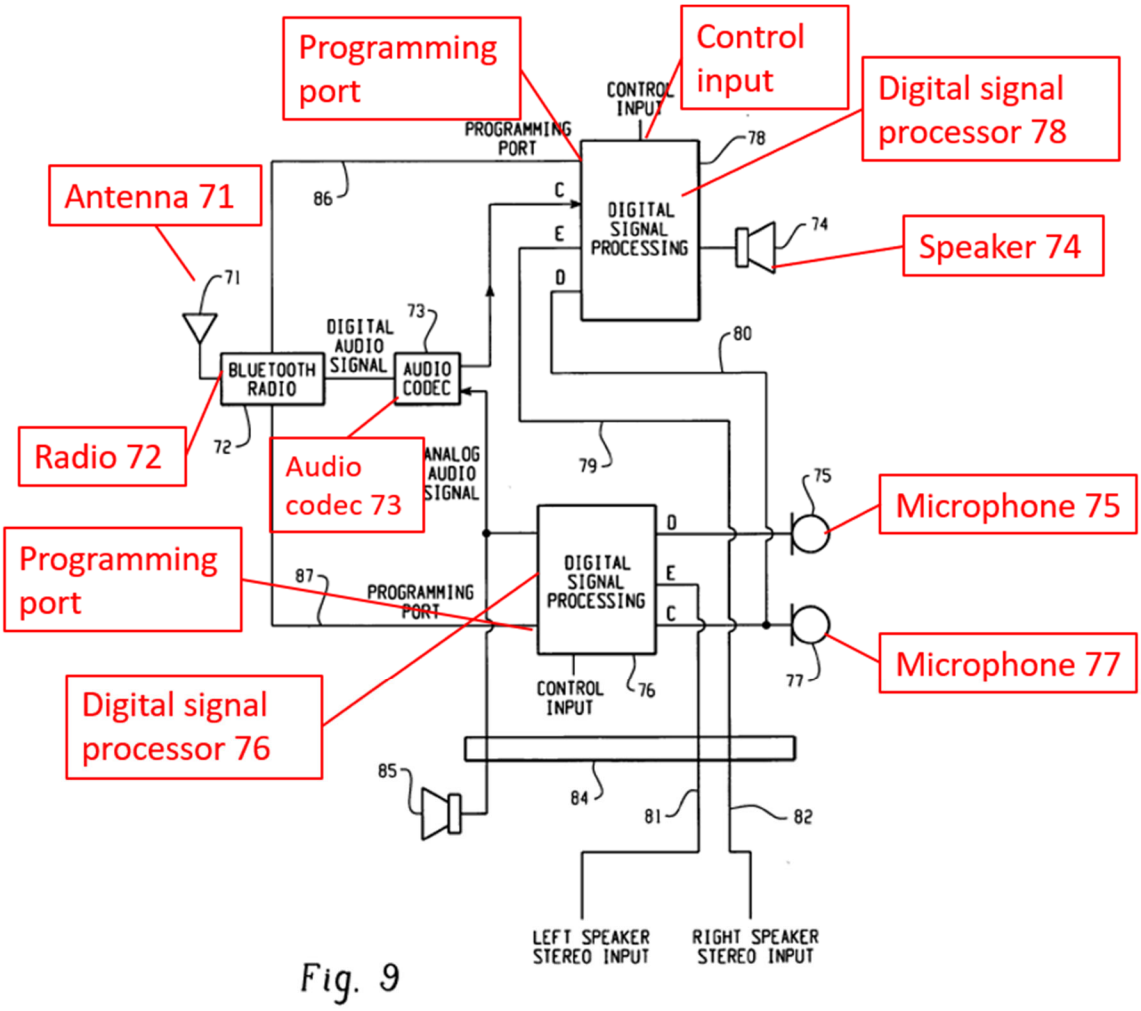
c. *[1.3]: communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;*

190. *Patel* in combination with *Topholm* discloses or suggests this feature.

191. *Patel* discloses that the external device wirelessly links with the head-mounted device for exchanging audio and control signals. *Patel* ¶¶[0001], [0018]-[0021]. *Patel* explains that, "[i]ncoming audio signals may be transmitted from the external device over the wireless link to the antenna 1" and outgoing audio signals are "transmitted to the external device by the antenna 1." *Id.* ¶¶[0019], [0021]. *Patel* further discloses connecting left and right audio feeds from an external device. *Id*. ¶¶[0059], [0060], FIGs. 8, 9. As explained in §X.A.1.a, a POSITA would have understood that the description in paragraphs [0001] and [0018]-[0021] applies to the head-mounted device of Figure 9. Forming a wireless link and exchanging signals with the head-mounted device is consistent with how a POSITA would have understood "communicatively connecting" to the head-mounted device. Thus, a POSITA would have understood *Patel* to disclose or suggest an external device communicatively connecting to the head-mounted device.

192. As detailed in §X.A.1.a, a POSITA would have understood *Patel* to disclose implementing its head-mounted device as an "earpiece."

193. *Patel* further discloses that the head-mounted device includes two microphones 75 and 77. *Patel* ¶[0060], FIG. 9; *see also id.* ¶¶[0018] ("In addition, the wireless head-mounted device further includes … a pair of microphones 5, 7."), [0020], [0026], [0032], [0034], FIGs. 1, 2, 4-7, 8.



Patel, FIG. 9 (annotated)

194. *Patel* further discloses that the head-mounted device includes a speaker. *See Patel* ¶[0060], FIG. 9; *see also id.* ¶¶[0018] ("The wireless head-

mounted device includes … a speaker 4."), [0024], [0033], [0059], FIGs. 1, 4, 6-9, 11, 12, 13.

195. *Patel* further discloses that the head-mounted device includes a processor with a memory. In particular, *Patel* discloses that the head-mounted device includes a programmable digital signal processor. *Patel* ¶[0060], FIG. 9; *see also id*. ¶¶[0018] ("the wireless head-mounted device further includes a digital signal processor 6"), [0022] ("***By integrating a signal processor 6 and microphones 5, 7 into the communication head-mounted device***, a directional response can be generated that eliminates the need for a mechanical boom extending out from the head-mounted device. … Moreover, because the signal processor 6 is programmable, it can generate a number of different directionality responses and thus can be tailored for a particular user or a particular environment."), [0026], FIGs. 1, 2, 4, 5, 6-9. A POSITA would have understood that a programmable digital signal processor (DSP) includes a memory. For example, *Oshana* mentions this basic fact in the Introduction section: "A DSP is really just a special form of a microprocessor. It has all of the same basic characteristics and components; a CPU, memory, instruction set, buses, etc." EX1036 (*Oshana* p. xi).[5] *Ohsana* further discloses, in the DSP Architectures

[5] To be clear, I am relying on *Oshana* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

chapter, "This simultaneous use of two data buses is a fundamental characteristic of a DSP system. If the coefficients were constants, the user might want to store them in ROM. This is a common place to hold the program, and it allows the user to perform the MAC [multiply and accumulate] operations using coefficients stored in program memory and accessed via the program bus." EX1036 (*Oshana* p. 124).

196. To note, although *Patel*'s Figure 9 shows two digital signal processing blocks, *Patel* discloses, in reference to Figure 4, that the two "digital signal processing blocks" may be implemented with "a single device." *Id*. ¶[0030] ("It should be understood that although shown as two separate processing blocks in FIG. 4, the digital signal processors 26, 28 may be implemented using a single device."). A POSITA would have understood this disclosure to mean that the functions of the digital signal processing blocks 26 and 28 may be implemented with a single digital signal processor. This is confirmed, for example, by *Patel*'s claims, which recite a single processor performing functions for both the receive path processing block and the transmit path processing block. *Patel* claims 22, 28, and 30 are copied below, and together they recite a processor that both receives an audio signal from an external device (as with digital signal processing block 28 in the receive path in

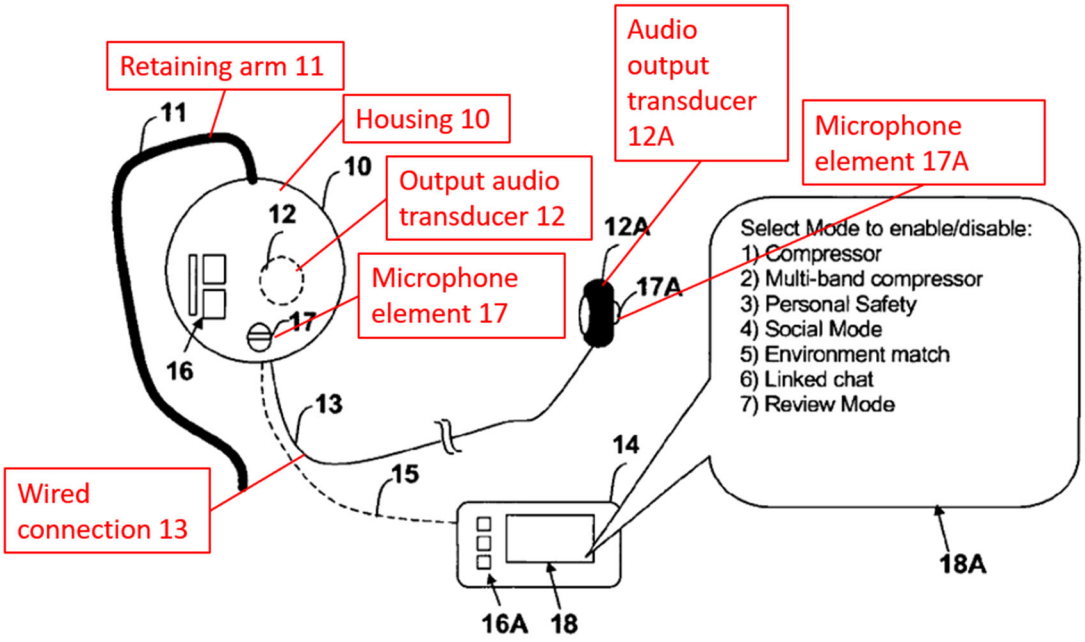
Figure 4) and receives a signal from the microphone and transmits it to an external device (as with digital signal processing block 26 in Figure 4).

> **22**. A head-mounted device for providing environment specific noise reduction comprising:
> a memory device for storing a plurality of environment-specific noise reduction algorithms (ESNRAs);
> a user input device operable to select one or more of the ESNRAs;
> ***a processor*** operable to filter environmental sounds using the selected ESNRA to create a filtered audio signal.
>
> 28. The head-mounted device of claim 22,
> wherein the head-mounted device is operable in a communications mode;
> wherein in the communications mode ***the processor is operable to receive an audio signal from an external device***, is operable to filter the audio signal by using the selected ESNRA, and is operable to transmit a filtered audio signal from the external device to a user via the speaker.
>
> 30. The head-mounted device of claim 28, wherein in the communications mode ***the processor is further operable to receive an audio signal from the microphone***, to filter the audio signal by using a user-selected ESNRA, and to transmit a filtered audio signal to an external device.

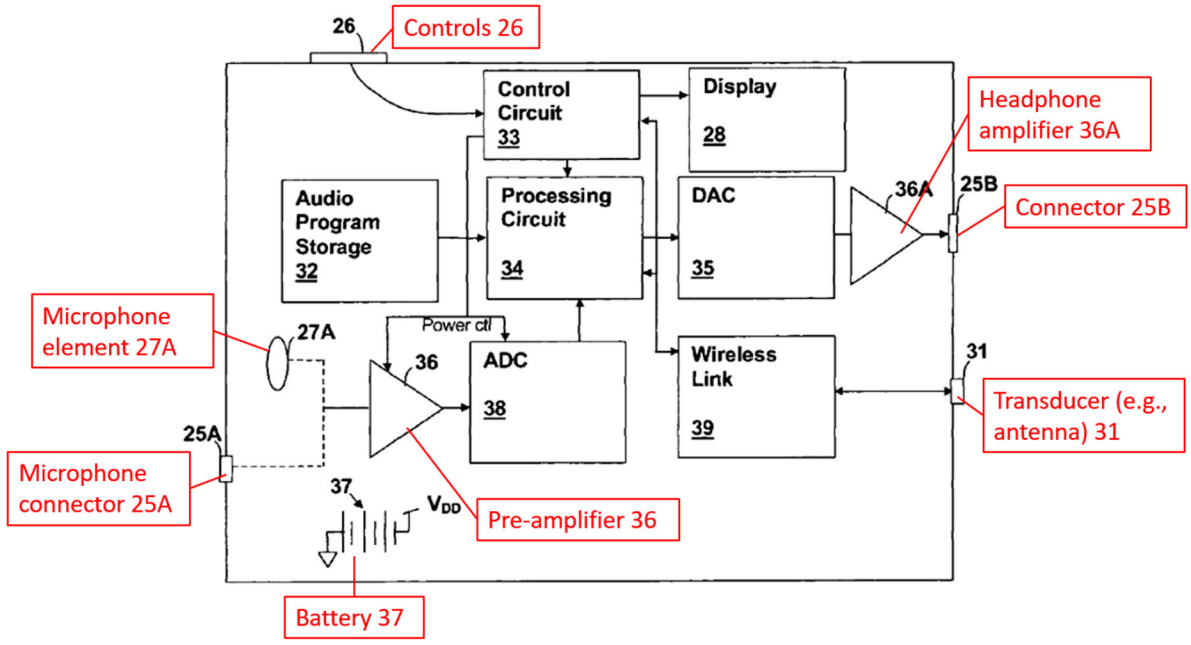
197. A POSITA would have understood that *Patel*'s disclosure in paragraph 30 that the two digital signal processing blocks can be implemented

by a single processor applies to the digital signal processing blocks 76 and 78 in Figure 9. In addition to the use of like numerals, a POSITA would have recognized the similar structure and functionality ascribed to the processing blocks in Figure 9 compared to Figure 4.

198. To be sure, even though Figure 9 is a stereo arrangement (i.e., a speaker in each ear), a POSITA would have understood that the single-processor disclosure applies to Figure 9 and that Figure 9 is compatible with that implementation. For example, a POSITA would have been aware of form factors where the processing functions of digital signal processing blocks 76 and 78 could be performed in one earpiece with a connection to a second earpiece that houses a speaker but no separate/additional processor. For example, *Melanson*, as discussed further below in §X.C.1, discloses such a device. *See Melanson* FIGs. 1, 3. The housing 10 includes a processor, memory, and speaker and is connected to a second earpiece that includes an output audio transducer 12A but no processor.



Melanson, FIG. 1 (annotated)



Melanson, FIG. 3 (annotated)

199. Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest t that an external device with a processor

executes instructions to form a wireless link with a head-mounted device ("communicatively connecting to an earpiece") wherein the head-mounted device has two microphones ("wherein the earpiece includes a first microphone, a second microphone"), a speaker ("a speaker"), and a programmable digital signal processor ("a second memory and an earpiece processor").

d. *[1.5]: sending an audio content signal to the earpiece,*

200. *Patel* in combination with *Topholm* discloses or suggests this feature.

201. *Patel* discloses that the head-mounted device may be "configured to listen to a high fidelity external stereo audio source such as a CD player or MP3 player" via "left and right side audio feeds." *Id.* ¶¶[0059], [0060], FIGs. 8, 9. A "high fidelity external stereo audio source" is consistent with how the '473 patent describes this feature. *'473pat.*, 12:28-32 ("The method 900 also receives as input an Audio Content signal 902, e.g. a music audio signal from a portable Media Player or mobile-phone."); *see also id.*, 3:8-10, 6:63-7:27. A POSITA would have understood that the description in paragraph [0059] related to Figure 8 applies to the head-mounted device in Figure 9 because of the similar numerals and structural components in common (including the left and right stereo inputs).

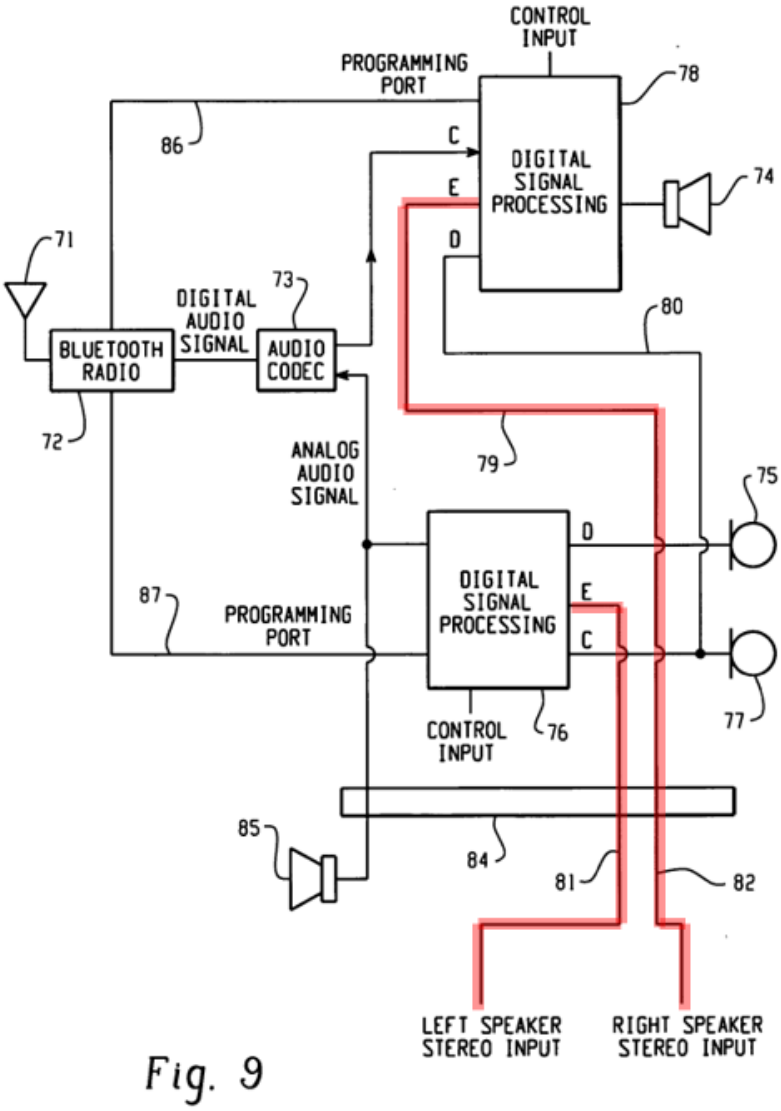


Fig. 9

Patel, FIG. 9 (annotated)

202. Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest a processor located in an external device that is configured to execute instructions stored in memory to send audio signals from the external device to an earpiece ("sending an audio content signal to the earpiece").

e. *[1.4]: sending an audio content gain value and an ambient sound gain value to the earpiece; and*

203. As explained in §X.A.1.d, the *Patel-Topholm* combination discloses or suggests an external device sending an audio content signal to the earpiece. The combination further discloses or suggests the same external device sending an audio content gain value and an ambient sound gain value to the earpiece.

(i) [1.4.1]: sending an audio content gain value … to the earpiece

204. *Patel* in combination with *Topholm* discloses or suggests this feature.

205. First, *Patel* discloses that the external device sends an audio content gain value to the earpiece. In particular, *Patel* discloses that the head-mounted device "can be made to function as an audio equalizer" due to "individually adjustable filters (amplitude and bandwidth) within the processors' filter banks." *Patel* ¶[0060]. In this manner, the "the audio characteristics of the left and right audio feeds 81, 82 may be altered" by "using pre-programmed equalizer settings, such as amplitude and bandwidth settings." *Patel* ¶[0060]. More particularly, the digital signal processing blocks 76, 78 "divide a given signal bandwidth into a number of bins" and "each bin may be of equal or different bandwidths," and "each bin may be capable of individual amplitude adjustment." *Id*.

206. Furthermore, with an application running on an external device, "[t]he equalizer settings may be transferred over the wireless link to the head-mounted device, where the amplitude and bandwidth settings for each filter within the filter bank of the signal processors 76, 78 are programmed via the programming ports of digital signal processing blocks 76, 78." *Id*.

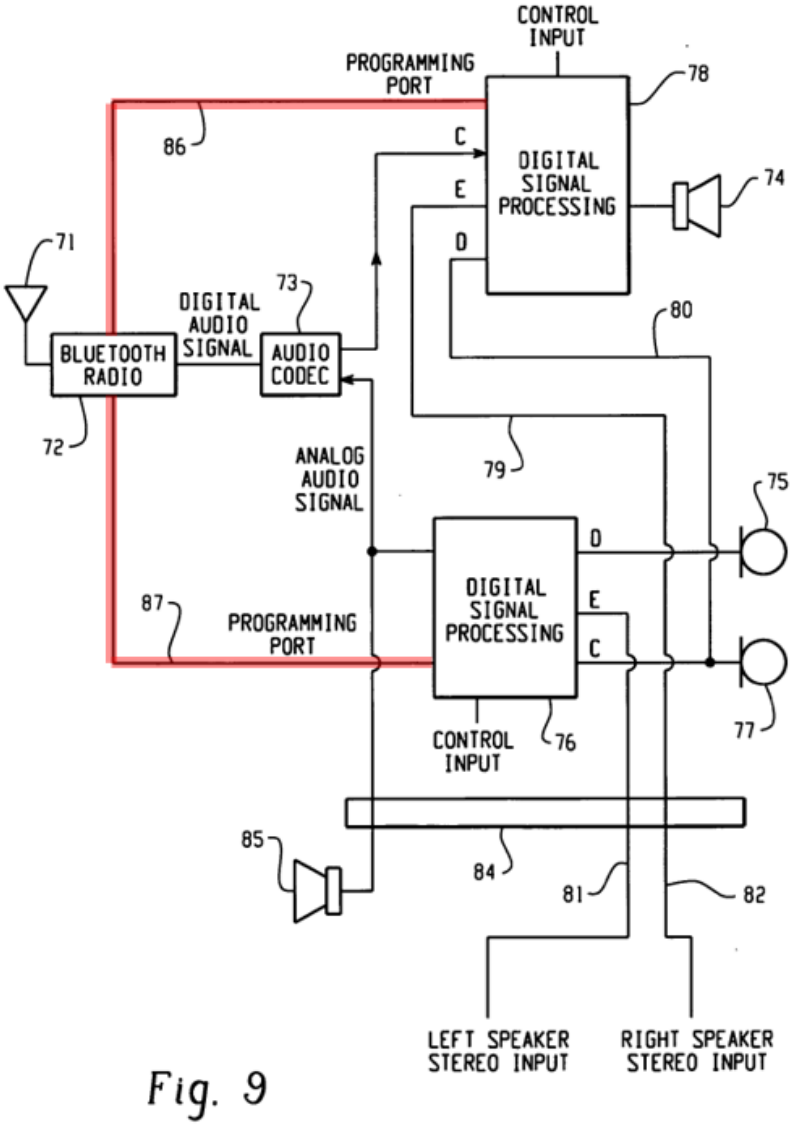


Fig. 9

Patel, FIG. 9 (annotated)

207. A POSITA would have understood that that the amplitude setting of a filter is a gain value. This is corroborated, for example, by *Maag* at 1:8-2:19 (explaining that well known graphic equalizers allow the user to select which bands of the audio signal are boosted (i.e., amplified) and which are cut (i.e., attenuated)).[6] A POSITA would have understood that changing (boosting or cutting) the level (amplitude) in one or more bands of a graphic equalizer results in an overall gain change in the audio signal. Thus, transferring the amplitude and bandwidth settings for each filter within the filter bank, from an external device to the head-mounted device, comprises transmitting a gain value from the external device to the head-mounted device (more particularly, a gain value for each frequency band in the equalizer). And because the filter settings are applied to the audio content signal, a POSITA would have understood that the transmitted gain value was applied to the audio content signal (i.e., an audio content gain value).

208. A POSITA would have understood *Patel* to disclose or suggest the same external device that transmits the high fidelity audio signal ("audio content signal") also transmits the equalizer settings. First, *Patel* explains that both the high fidelity audio signal and the equalizer settings may be received from an

[6] To be clear, I am relying on *Maag* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

MP3 player. *Patel* also consistently identifies an MP3 player among the exemplary external devices with which the head-mounted device may communicate. *Patel* ¶[0018], [0059], [0060]. Thus, at a minimum, *Patel* suggests a system and method involving the transmission of a high fidelity audio signal and equalizer settings from, e.g., an MP3 player, to the head-mounted device.

209. Even if *Patel* does not itself disclose or suggest that the same external device that transmits the high fidelity audio signal also transmits the equalizer settings, *Topholm* discloses an external mobile device that transmits both control signals (e.g., for volume adjustments) (*Topholm*, 5:4-10) and digital audio (*Topholm*, 6:5-6) to a hearing aid. A POSITA would have considered it obvious to implement *Topholm*'s teaching in this regard in *Patel* because this would have been more convenient to the user than having to carry two separate external devices—one to transmit music and the other to transmit the equalizer settings for the music. Moreover, *Patel* discloses using Bluetooth to connect to the remote device, and a POSITA would have understood that Bluetooth allows transmission of audio and remote control over the Bluetooth link. The combination would have amounted to applying a known technique (transmitting control signals and audio data) to a known device (the *Patel* head-mounted device system) ready for improvement (*Patel*'s head-mounted device is already

configured to receive control signal and audio data from a remote device) to yield predictable results (a head-mounted device that can receive controls signals and audio data from the same remote device).

210. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest the external device with a processor executing instructions to transfer an amplitude setting for each filter within the filter bank to the head-mounted device ("sending an audio content gain value … to the earpiece.").

(ii) [1.4.2]: sending … an ambient sound gain value to the earpiece

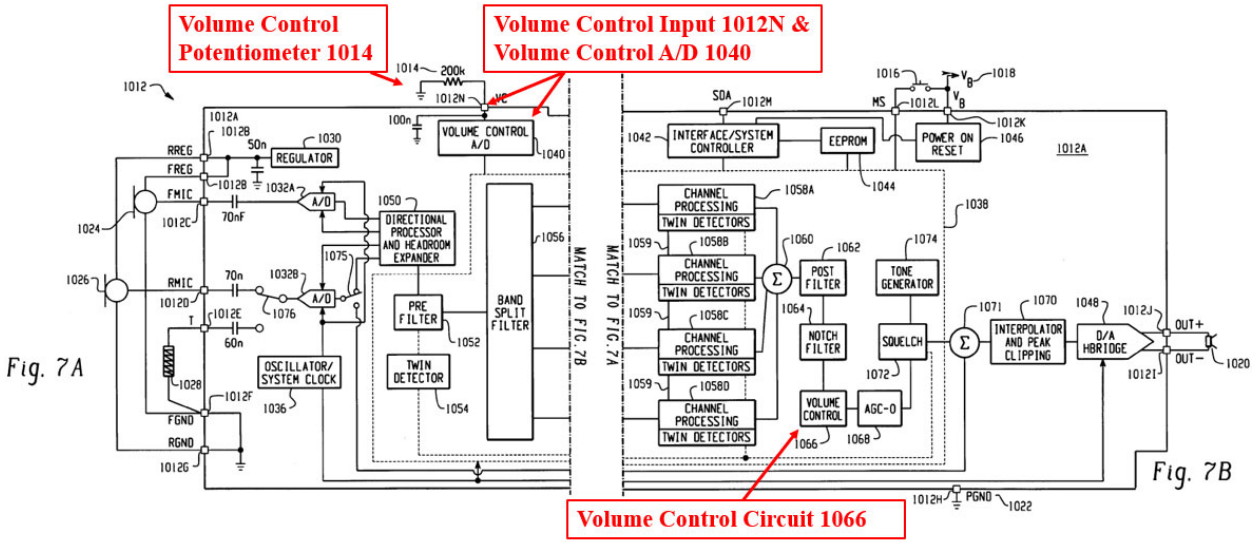
211. *Patel* in combination with *Topholm* discloses or suggests this feature.

212. *Patel* discloses that environmental sounds may be introduced by the digital signal processor into the audio signal transmitted into the ear canal and that with a control signal sent from an external device, the user may turn down or off the environmental sounds. *Patel* ¶[0028] ("Environmental sounds (i.e., audio signals from outside of the ear canal) may be received by the microphone 27 and introduced by the signal processor 28 into the audio signal being transmitted into the ear canal in order to reduce occlusion. By electronic (e.g., a control signal sent by a wireless or direct link) or manual means via the control input to the digital signal processor 28, the user may turn down or turn off the

environmental sounds, for example when the head-mounted device is in a communication mode (e.g., when a cellular call is initiated or in progress.)"). A POSITA would have understood that the description of that feature in paragraph [0028] applies to the head-mounted device in Figure 9 because the related structure, e.g., the microphone input to the DSP processing block in the receive path, is present in both Figures 4 and 9.

213. A POSITA would have understood *Patel*'s reference to "turn[ing] down" the environmental sounds to refer to lowering the volume of the environmental sounds because *Patel* explains that the user may turn down the environmental sounds "when the head-mounted device is in a communication mode," which indicates that the purpose of "turn[ing] down" the environmental sounds is to better hear the audio signals from an external device.

214. A POSITA would have understood *Patel* to suggest that the control signal to turn down the volume of the environmental sounds includes a gain value. For example, *Patel* discloses a detailed circuit diagram for an exemplary head-mounted device in Figures 7A-7B. *Patel* ¶[0033]. In relevant respect, the circuit includes a volume control potentiometer 1014 that is coupled to a volume control input 1012N and is used to set the desired volume level provided to volume control A/D 1040. *Patel* ¶[0036].



Patel, FIGs. 7A, 7B (annotated)

215. A POSITA would have understood that the potentiometer is a device that allows the user to adjust the volume of the audio output. In hearing aids, this was typically in the form of a rotatable wheel or dial that the user could adjust up or down to a particular setting. *See AudioCyclopedia* at 560.[7] *Patel* further explains that a volume control circuit 1066 "receives a digital value from the volume control A/D 1040, which indicates the desired volume level set by the user via potentiometer 1014, and uses this stored digital value to set the gain of an included amplifier circuit." *Id.* ¶[0054]. Thus, *Patel* suggests that adjusting the volume of the microphone output involves receiving a "digital value" that is

[7] To be clear, I am relying on *AudioCyclopedia* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

used to "set the gain of an included amplifier circuit." This is consistent with how a POSITA would have understood a "gain value."

216. In view of *Patel*'s disclosure with respect to the exemplary head-mounted device in Figure 7, a POSITA would have considered it obvious to implement a similar technique when using an external device to transmit a control signal to turn down environmental sounds in the embodiment of Figure 9. Indeed, a POSITA would have understood that a control signal to turn down the environmental sounds would include either a value to increment or decrement a gain value applied to the audio signal (e.g., a value of +0.1 or -0.1), or a gain value to replace the currently stored gain value (e.g., a value of 0.8). This is corroborated, for example, by the *Dorogusker* reference.[8]

217. *Dorogusker* "relates to media devices and, more particularly, to portable media devices, such as portable media players, that receive accessory devices." *Dorogusker* ¶[0003]. *Dorogusker* is directed to providing "improved interaction between media devices and accessory devices." *Id*. ¶[0008]. "Media devices" include a handheld media player, and "accessory devices" include headphones. *Id*. ¶[0028]. *Dorogusker* discloses a "flow diagram of an audio playback process" in Figure 5A including a detailed description of how volume

[8] To be clear, I am relying on *Dorogusker* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

adjustments may be made. *See id.* ¶¶[0049]-[0055]. The method includes a step of receiving a volume adjustment, and *Dorogusker* explains that "[w]hen the volume setting adjustment 504 is performed on the portable media device, the stored volume status of the auxiliary speaker system is updated 506. Typically, this requires that the portable media device inform the auxiliary device of the adjustment to the volume setting. In so informing the auxiliary device, ***the portable media device could provide an adjustment relative to a current setting (e.g., increase by two levels), or could provide a new volume setting (e.g., volume level 8)***." *Id*. ¶[0051].

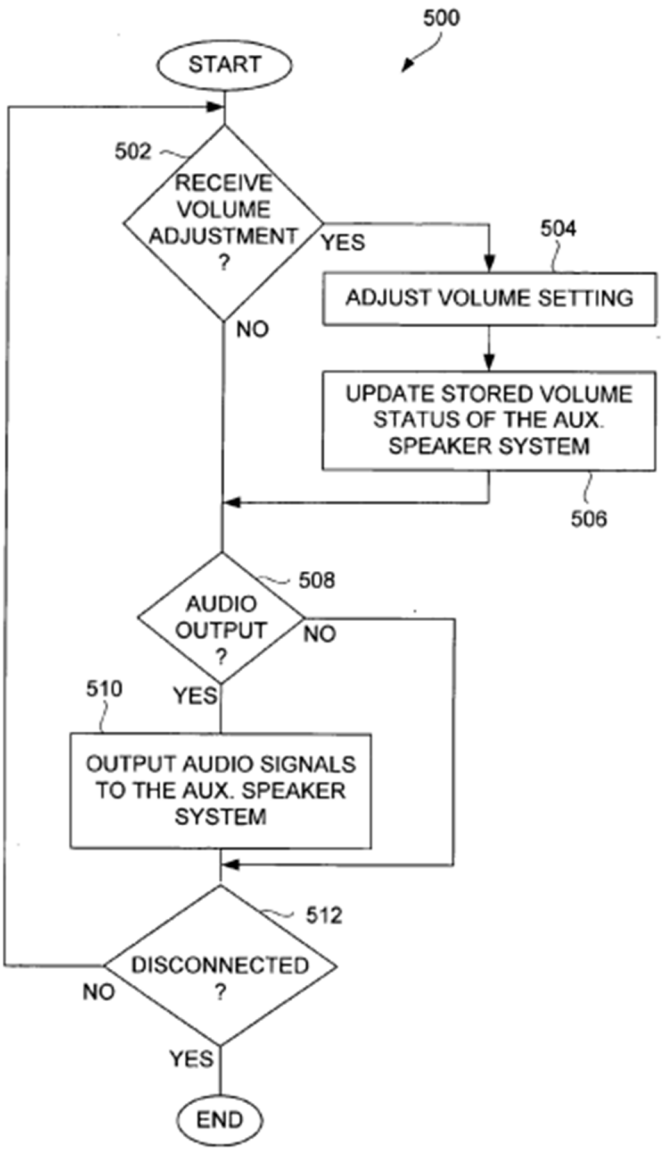


FIG. 5A

Dorogusker, FIG. 5A

218. Thus, as corroborated by *Dorogusker*, a POSITA would have understood that a volume adjustment control from an external device may be done relative to a current setting or by providing a new volume setting. A POSITA would have understood that *Patel*'s Figure 7 embodiment expressly

suggests the latter implementation because the received digital value is used to "set the gain" of the amplifier. A POSITA also would have understood that the latter implementation would be advantageous because it would provide the user with the ability to implement pre-programmed volume settings, which are convenient to users. Thus, a POSITA would have considered it obvious, when implementing a control signal to turn down the environmental sounds, to include a gain value to be applied to the audio signal including the environmental sounds.

219. In addition, for the same reasons discussed in §X.A.1.e(i), *Patel* at least renders obvious transmitting the control signal to turn down environmental sounds from the same external device that transmits audio signals and equalizer settings.

220. Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest an external device with a processor executing instructions to send a control signal including a gain value to be applied to the audio signal including the environmental sounds ("sending … an ambient sound gain value to the earpiece").

f. *[1.6]: wherein the earpiece processor applies the audio content gain value to the audio content signal to generate a modified audio content signal,*

221. *Patel* in combination with *Topholm* discloses or suggests this feature.

222. As explained in §X.A.1.d, a POSITA would have understood *Patel* to disclose sending an audio content gain value by sending the amplitude settings for each filter within a filter bank. In addition, *Patel* discloses that "the audio characteristics of the left and right audio feeds 81, 82 may be ***altered by the digital signal processing blocks 76, 78***." *Patel* ¶[0060]. And as discussed in §X.A.1.c, *Patel* discloses implementing the digital processing blocks with a single digital signal processor. A POSITA would have understood that after the audio content signal is processed or "altered by the digital signal processing blocks 76, 78" it is a "modified audio content signal."

223. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest that the digital signal processor of the head-mounted device ("the earpiece processor") applies the equalizer settings provided by the user ("applies the audio content gain value") to the audio signal ("to the audio content signal") to alter the audio characteristics of the audio signal ("generate a modified audio content signal").

g. *[1.7]: wherein the earpiece processor applies the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal, and*

224. *Patel* in combination with *Topholm* discloses or suggests this feature.

225. *Patel* discloses that "[b]y electronic (e.g., a control signal sent by a wireless or direct link) or manual means via the control input ***to the digital signal processor*** 28, the user may turn down or turn off the environmental sounds, for example when the head-mounted device is in a communication mode (e.g., when a cellular call is initiated or in progress.)." *Patel* ¶[0028].

226. *Patel* explains that the environmental sounds are "audio signals from outside of the ear canal" and "may be received by the microphone 27." *Patel* ¶[0028]. This is consistent with how a POSITA would have understood "ambient sound signal." Thus, a POSITA would have understood *Patel* to disclose that the digital signal processor of the head-mounted device that applies the transmitted ambient sound gain value to an ambient sound signal.

227. A POSITA would have understood that the gain-reduced ambient sound signal constitutes a "modified ambient sound signal" because reducing the gain of the signal modifies it.

228. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest that the digital signal processer of the head-

mounted device ("the earpiece processor") applies the gain value included in the control signal ("applies the ambient sound gain value") to the environmental sounds reproduced to the user ("an ambient sound signal") such that the gain of the signal representing the environmental sounds reproduced to the user is adjusted ("generate a modified ambient sound signal").

h. *[1.8]: wherein the earpiece processor mixes the modified ambient sound signal with the modified audio content signal to generate a mixed signal.*

229. *Patel* in combination with *Topholm* discloses or suggests this feature.

230. *Patel* discloses that, when in communication mode, audio signals from an external device are output to the speaker in the user's ear canal. *Id*. ¶¶[0019], [0027]-[0028], [0032]. *Patel* further discloses that "[e]nvironmental sounds … may be received by the microphone 27 and introduced by the signal processor 28 into the audio signal being transmitted into the ear canal" and that the environmental sounds can be "turn[ed] down … ***when the head-mounted device is in a communication mode*** (e.g., when a cellular call is initiated or in progress.)." *Patel* ¶[0028]. *Patel* describes this as a "stereo mode." *Id.* ¶[0060]. A POSITA would have understood that *Patel*'s disclosure of providing environmental sounds to the user "in a communication mode" applies to *Patel*'s "stereo mode" as well. *Patel* makes it clear that "communication mode"

generally refers to a mode in which the head-mounted device is in communication with a remote device because, for example, *Patel* discloses various types of external devices with which the head-mounted device may "communicate," including an MP3 player and other devices that are not cellular phones. *Patel* ¶[0018] ("Other types of wireless links could also be utilized, and the device may be configured to communicate with a variety of different external devices, such as cellular phones, PDAs, radios, MP3 players, CD players, portable game machines, etc."). Thus, a POSITA would have understood *Patel*'s reference to "stereo mode" to be a type of communication mode in which audio is played stereophonically, and that *Patel*'s "communication mode" broadly refers to a mode in which the headset is exchanging signals with a remote device, including for purposes other than a phone call. And therefore, a POSITA would have understood *Patel*'s disclosure in paragraph [0028] to encompass introducing environmental sounds into a high fidelity audio signal (e.g., from an MP3 player or other communication device) provided to the user

231. A POSITA would have understood that the environmental sounds are "introduced" into the audio signal by "mixing" them with the audio signal because that is how signals are "introduced" into each other. Therefore, a POSITA would have understood *Patel* to disclose that the processor mixes

environmental sounds into an audio signal from an external device when in a communication mode.

232. *Patel* discloses that the environmental sounds may be "turn[ed] down" while in a communication mode. *Patel* ¶[0028]. Thus, a POSITA would have understood *Patel* to disclose that the volume adjustment to the audio signal of the environmental sounds was made ***prior to*** mixing it with the audio signal from the external device such that when the processor introduces the environmental sounds into the audio signal delivered to the ear canal, it is the "modified ambient sound signal" (i.e., the turned-down environmental sound signal) that is being introduced/mixed.

233. As detailed in §X.A.1.f, a POSITA would have understood that the "modified audio content signal" is the signal that has been processed through the processor's programmable filter banks. Because *Patel* discloses that the programmable filter banks adjust the "audio characteristics of the left and right audio feeds" and *Patel* ***does not*** disclose that the programmable filter banks adjust the audio characteristics of the microphone input environmental sounds, a POSITA would have understood *Patel* to disclose that the audio signals from the external device are processed/filtered before the environmental sound is introduced into the audio signal. Therefore, a POSITA would have further understood *Patel* to disclose or suggest that the mix included both the "modified

ambient sound signal" discussed above in §X.A.1.g and the "modified audio content signal" discussed above in §X.A.1.f.

234. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest that, when in a communication mode, the processor of the head-mounted device ("wherein the earpiece processor") introduces the turned-down environmental sound signal into the filtered audio signal from an external device ("mixes the modified ambient sound signal with the modified audio content signal to generate a mixed signal").

2. Claim 3 ("The device according to claim 1, wherein the earpiece processor generates the ambient sound signal from a first microphone signal from the first microphone"); Claim 5 ("The device according to claim 1, wherein the earpiece processor applies an ambient gain to a first microphone signal from the first microphone to generate the ambient sound signal.")

235. *Patel* in combination with *Topholm* discloses or suggests these claims.

236. Claims 3 and 5 add a requirement that the "ambient sound signal" is "generat[ed]" from "a first microphone signal." Claim 5 in particular requires "appl[ying] an ambient gain" to the "first microphone signal."

237. As explained in §§X.A.1.a and X.A.1.e(ii), *Patel* discloses a microphone input is used to provide environmental sounds to the user, and this is consistent with how the '473 patent describes an "ambient sound signal." *Patel*

discloses that before the microphone input is provided to the user, the microphone input is first digitized and then "processed through a filter bank 14 to optimize the overall frequency response." *Patel* ¶[0021], FIGs. 2, 5. A POSITA would have understood that this description applies to the head-mounted device of Figure 9 and that *Patel* discloses that the digital signal processor for use in Figure 9 also includes a filter bank for optimizing the overall frequency response of the microphone input.

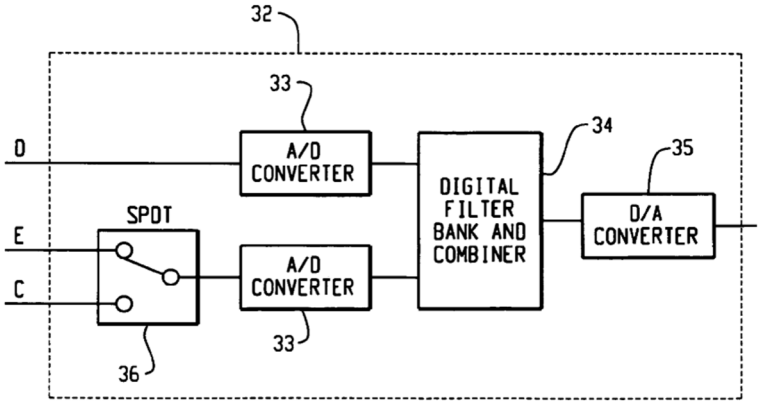


Fig. 5

Patel, FIG. 5

238. A POSITA would have understood that the analog microphone signal that is digitized constitutes a "first microphone signal" because it is the digital version of an analog microphone signal from one of the microphones in the head-mounted device. Because the digitized microphone signal is processed through a filter bank before being provided to the user, a POSITA would have

understood that the processor of the head-mounted device "generates the ambient sound signal from a first microphone signal from the first microphone."

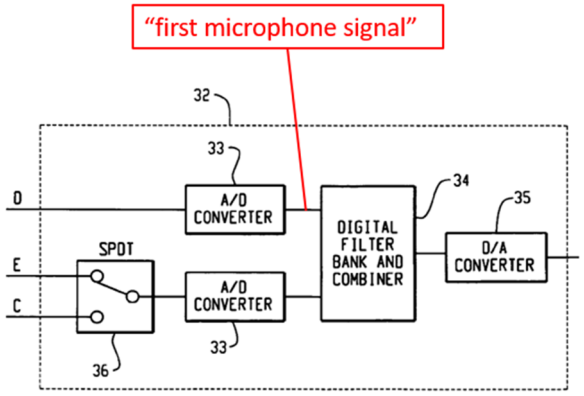


Fig. 5

Patel, FIG. 5 (annotated)

239. A POSITA would have understood that filtering the signal to optimize the overall frequency response includes applying a gain to the signal. A POSITA would have understood that optimization of the frequency response involves adjustments of the audio signal at various frequencies. For example, it is often desirable to prevent certain low frequencies from entering the signal processing path and so the digital filter bank would be configured to reduce the gain at those frequencies. That is, a POSITA would have understood that a digital filter adjusts the gain of one or more frequency components of the signal by applying a gain to those frequency components. This understanding is corroborated, for example, by *Bongiovi*, which explains that filters apply a gain to certain frequency components of the signal to optimize the frequency

response. EX1040 at 11 ("In one embodiment, each of the low-shelf filters 102, 105 is a filter that has a nominal gain of 0 dB for all frequencies above a certain frequency termed the corner frequency. For frequencies below the corner frequency, the low-shelving filter has a gain of ±G dB, depending on whether the low-shelving filter is in boost or cut mode, respectively. This is shown in FIG. 2."). It also corroborated by *Bohn* at 6:58-60, 1:20-25, 2:27-32, and 10:14-17.[9]

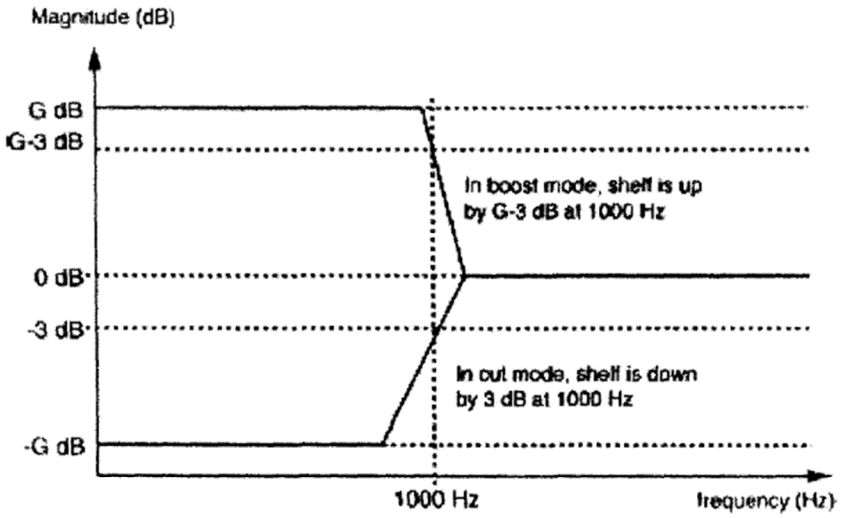


Figure 2: Low shelving filters with cutoff frequency at 1000 Hz and gains $\pm GdB$

FIG. 2

Bongiovi, FIG. 2

[9] To be clear, I am relying on *Bongiovi and Bohn* as evidence of the knowledge of a person having ordinary skill in the art and they are not part of the unpatentability ground 1A.

240. A POSITA would have understood that the gain applied to the microphone signal to constitute an "ambient gain" within the meaning of the '473 patent because it is applied to a signal that represents environmental (i.e., ambient) sounds.

241. In view of the above, a POSITA would have understood *Patel* to disclose an embodiment in which the head-mounted device is in a communication mode and outputting an audio signal that includes both environmental sounds and, for example, stereo audio. In that embodiment, *Patel* discloses that the microphone input signal including the environmental sounds is digitized ("first microphone signal"), filtered to optimize the frequency response ("ambient sound signal"), volume-adjusted (e.g., turned down) ("modified ambient sound signal"), and then mixed with an equalized stereo audio signal ("modified audio content signal").

242. To be clear, for purposes of my analysis for independent claim 1, the "ambient sound signal" may be mapped to the digitized microphone signal ***or*** the digitized and filtered microphone signal. For purposes of claims 3 and 5, the digitized microphone signal is a "first microphone signal" that is modified by the filter, which generates an "ambient sound signal" within the meaning of claims 3 and 5.

243. Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest generating that the processor of the head-mounted device applies a filter to digitized microphone signal representing environmental sounds to generate a filtered microphone signal with an optimized frequency response ("wherein the earpiece processor generates the ambient sound signal from a first microphone signal from the first microphone") and that the filter applies a gain to the digitized signal to generate filtered microphone signal with an optimized frequency response ("wherein the earpiece processor applies an ambient gain to a first microphone signal from the first microphone to generate the ambient sound signal.").

3. Claim 4 ("The device according to claim 1 further including a user interface.")

244. *Patel* in combination with *Topholm* discloses or suggests this claim.

245. *Patel* discloses various examples of the "external device," including "cellular phones, PDAs, radios, MP3 players, CD players, portable game machines." *Patel* ¶[0018]. As explained in §X.A.1.a, a POSITA would have understood this description to apply to the embodiment in Figure 9. *Patel* further discloses that the external device may be a device that "emulates a graphical equalizer" and "can be displayed on a computer screen and adjusted in real time under user control." *Patel* ¶[0060]. This is consistent with how a POSITA would have understood a "user interface" to provide user control of a device.

246. Accordingly, a POSITA would have understood the *Patel*-*Topholm* combination to disclose or suggest an external device with a user interface ("The device according to claim 1 further including a user interface.").

4. Claim 6 ("The device according to claim 1, wherein the device is a portable communication device"); Claim 7 ("The device according to claim 6, where the device portable communication device is at least one of a phone, a watch, a tablet, earpiece, a laptop computer or a combination thereof.")

247. *Patel* in combination with *Topholm* and *Beasley* discloses or suggests these claims.

248. *Patel* discloses various examples of the "external device," including "cellular phones, PDAs, radios, MP3 players, CD players, portable game machines." *Patel* ¶[0024]; *see also id*. ¶[0058] ("FIG. 8 shows an example of a communication head-mounted device that is configured to listen to a high fidelity external stereo audio source such as a CD player or MP3 player."). A POSITA would have understood that, in the 2007-2008 timeframe, each of these devices were implemented in a portable form factor. *See* EX1042 ("*Lee*") ¶¶[0005] ("Conventionally, portable electronic devices, such as ***cellular phones***, ***portable digital assistants*** or ***portable media players***, have provided visual clues regarding certain device status conditions."), [0035] ("The improved techniques are also well suited for use with ***portable electronic devices*** having audio playback capabilities, such as portable media devices (e.g., digital music player

or ***MP3 player***).").[10] A POSITA would have understood that, because *Patel* describes exchanging signals with these external devices, they are "communication devices." Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest that "the device is a portable communication device."

249. Moreover, because *Patel* discloses that the external device may be a "cellular phone," a POSITA would have understood *Patel* to disclose that the portable communication device is a "phone." To be clear, as noted above in §X.A.1.d, *Patel* discloses that the head-mounted device may be "configured to listen to a high fidelity external stereo audio source ***such as*** a CD player or MP3 player." *Patel* ¶[0058]. A POSITA would have understood *Patel* to disclose that the "high fidelity external stereo audio source" could be any of the exemplary "external devices" listed in paragraph [0018] even though *Patel* only lists devices "such as" a CD player or MP3 player in paragraph [0058]. Indeed, a POSITA would have understood that, by the earliest alleged priority date, it was well-known that cellular phones were a source of high fidelity external stereo audio and would be suitable to provide the audio feeds for the head-mounted device in Figure 9. *See, e.g.*, EX1043 ("*Mahini*") ¶[0020] ("One of the trendiest

[10] To be clear, I am relying on *Lee* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

and most desirable portable electronic device on the market today is the portable music player that can play music in a variety of data formats like MP3 or ATRAC among others. Such devices can and have been incorporated into mobile phones. Playback is typically accomplished via a set of headphones or via the mobile phone's relatively small speaker. Users of these devices greatly value their portability but would also like to listen to the music library in as high fidelity as possible. This can be achieved if the portable device can be coupled to a larger device for purposes of outputting the audio.").[11]

250. Accordingly, a POSITA would have understood the *Patel-Topholm* combination to disclose or suggest an external device, such as a cellular phone, PDA, radio, MP3 player, CD player, or portable game machine, that exchanges signals with the head-mounted device ("wherein the device is a portable communication device; where the device portable communication device is at least one of a phone").

[11] To be clear, I am relying on *Mahini* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 1A.

B. Ground 1B: *Patel*, *Topholm*, and *Williamson* Render Obvious Claim 2

1. Claim 2

a. *[2.1] The device according to claim 1, wherein the earpiece processor calculates a noise level based upon a first microphone signal from the first microphone*

251. *Patel* in combination with *Topholm* and *Williamson* discloses or suggests this feature.

252. *Patel* discloses that the head-mounted device may download "environment specific noise reduction algorithms (ESNRAs)" that are used to "filter environmental sounds from an audio signal(s)." *Patel* ¶[0064].

253. The ESNRAs may be applied to outgoing signals so that "the environmental sound present in the user's location can be reduced as it is heard by the other party to the call." *Id*. ¶[0067]. "With respect to incoming signals, the ESNRAs may be applied to (a) reduce environmental noise present on the other end of the call or (b) to filter noise from the user's environment. Thus, the user can benefit by having the environmental noise present in the other party's location reduced, or by filtering environmental noise present in the user's location. In another example, the ESNRAs could be used to reduce noise from both the user's location and the other party's location at the same time." *Id*.

254. Furthermore, "[i]n [a] noise reduction only mode[,] the ESNRA function is applied to environmental signals in the user's location. This mode, for

example, would aid users in hearing face-to-face communications in noisy environments. It could also be used, for example, to listen to television or music in a noisy environment." *Id*. ¶[0068].

255. *Patel* explains that the ESNRA that is used to filter the audio signals "may be automatically selected from memory by a device processor based on an analysis of the environmental noise present in the audio signal(s)." *Id*.; *see also id*. ¶[0066] ("In another example, the processor 260 may be able to automatically recognize which ESNRA would be most effective for a particular environment."). A POSITA would have understood this disclosure to mean that the ESNRA is selected based on an analysis of the noise present in the digitized signal captured by the microphone of the head-mounted device (i.e., the "first microphone signal"). This is because *Patel* discloses using the selected ESNRA to filter noise from the environmental sounds picked up by the microphone of the head-mounted device. *Patel* ¶[0067]; *see also id*. ¶[0068] ("In the noise reduction only mode the ESNRA function is applied to environmental signals in the user's location."), claims 22, 23, 32 (reciting a head-mounted device with "a microphone; wherein the processor is operable to receive an audio signal from the microphone, is operable to filter the audio signal by using the selected ESNRA, and is operable to transmit a filtered audio signal to a user via the speaker"). A POSITA would have understood that analyzing the digitized signal

picked up by the microphone would have been necessary to select the appropriate ESNRA to filter the noise out of that signal because that is how the extent and type of noise is determined.

256. *Patel* leaves the details of the "analysis" performed on the environmental sounds that is used to select the appropriate ESNRA to the knowledge of a POSITA. A POSITA looking for additional implementation details would have readily identified *Williamson*. *Williamson* is similar to *Patel* because it discloses analyzing the noise in a hearing aid microphone signal to adapt the filter used on the microphone signal. More particularly, *Williamson* discloses a "hearing aid system which is both programmable to fit the hearing deficit of a particular user and adaptive to the sound environment to maximize the intelligibility and quality of the audio signal provided to the user." *Williamson*, 1:57-62. The system includes a "digital filter having filter coefficients which can be varied on a quasi-real time basis … to best accommodate ambient signal levels and noise levels." *Id*., 3:36-41. "An estimate of the level of background noise is preferably made from the energy envelope of the input signal in various frequency ranges." *Id*., 2:67-3:1.

257. With reference to Figure 3, below, *Williamson* discloses that the microphone signal is subjected to "time varying and spectral shaping filtering" in block 66. *Williamson*, 8:6-14. *Williamson* explains that "[t]he filter

coefficients which define the characteristics of the spectral filter change over time to adapt to the magnitude and estimated noise content of the signal being processed." *Id*., 8:15-21. "More particularly, the system allows tracking of long term signal and noise levels and the use of estimates of these levels to maintain the output speech sound at a level which will be comfortable to the user while simultaneously controlling noise." *Id*., 8:36-41. An energy estimation algorithm "calculates the energy in a number (preferably 5) of frequency bands" and "[t]hese estimates are used to decide on the current gains or attenuations for each of the bands" of the filter. *Id*., 8:51-59.

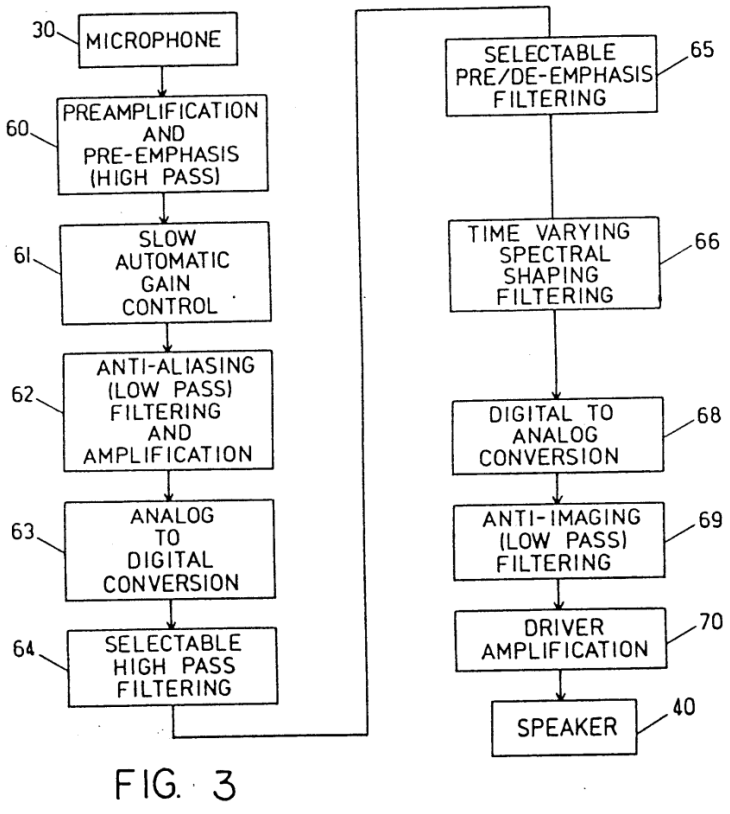


FIG. 3

Williamson, FIG. 3

258. In view of the above disclosures, a POSITA would have understood *Williamson* to disclose estimating a noise level based of a digitized microphone signal by calculating the energy envelope of the signal in a number of frequency bands and then using that calculation to adapt a filter to account for the ambient noise level.

259. *Williamson*'s description of estimating noise by calculating the energy envelope in a number of frequency bands is consistent with how the '473 patent describes "calculat[ing] a noise level":

> FIG. 8 is a flowchart of a method 800 for estimating background sound level. The method 800 can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method 800, reference will be made to components of FIG. 2 , although it is understood that the method 800 can be implemented in any other manner using other suitable components. The method 800 can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.
>
> Briefly, method 800 receives as its input 802 either or both the ASM signal from ASM 111 and a signal from the ECM 123. ***An audio buffer 804 of the input audio signal is accumulated (e.g. 10 ms of data), which is then processed by squaring step 806 to obtain the temporal envelope. The envelope is smoothed (e.g. an FIR-type low-pass digital filter) at step 808 using a smoothing window 810 stored in data***

> ***memory (e.g. a Hanning or Hamming shaped window). At step 812, transient peaks in the input buffer can be identified and removed to determine a "steady-state" Background Noise Level (BNL). At step 814 an average BNL 816 can be obtained (similar to, or the same as, the RMS) that is frequency dependent or a single value averaged over all frequencies.*** If the ECM 123 is used to determine the BNL, then decision step 818 adjusts the ambient BNL estimation to provide an equivalent ear-canal BNL SPL, by deducting an Earpiece Noise Reduction Rating 828 from the BNL estimate 826. Alternatively, if the ECM 123 is used, then the Audio Content SPL level (ACL) 822 of any reproduced Audio Content 820 is deducted from the ECM level at step 824. The updated BNL estimate is then converted to a Sound Pressure Level (SPL) equivalent 832 (i.e. substantially equal to the SPL at the ear-drum in which the earphone device is inserted) by taking into account the sensitivity (e.g. measured in V per dB) of either the ASM 111 or ECM 123 at steps 830 and 834 respectively. The resulting BNL SPL is then combined at step 842 with the previous BNL estimate 840, by averaging 838 a weighted previous BNL (weighted with coefficient 836), to give a new ear-canal BNL 844.

'473pat., 11:42-12:13.

260. A POSITA would have been motivated to implement *Williamson*'s teachings related to estimating an ambient noise level in the *Patel-Topholm* combination for the purpose of selecting the appropriate ESNRA used to filter the environmental noise and would have had a reasonable expectation of success

in doing so. *Patel* already discloses analyzing the environmental noise in a digitized microphone signal to select the appropriate ESNRA for use in the environment, and *Williamson* the details of such an analysis. In the combination, the digitized signal from *Patel*'s microphone that picks up environmental sounds ("first microphone signal") would be analyzed according to *Williamson*'s teachings to estimate a noise level, and the appropriate ESNRA would be selected according to the estimation.

261. The combination amounts to applying a known technique (estimating the ambient noise level to adapt a filter) to a known device (*Patel*'s head-mounted device) ready for improvement (*Patel*'s head-mounted device analyzes environmental noise to select an ESNRA for filtering) to yield predictable results (*Patel*'s head-mounted device that analyzes the environmental noise to estimate a noise level and select an appropriate ESNRA for filtering).

262. It also amounts to the use of a known technique (estimating a noise level) to improve similar devices (both *Patel* and *Williamson* describe head-mounted devices that filter microphone signals to account for ambient noise) in the same way (to adapt a filter based on the environmental noise).

263. Accordingly, a POSITA would have understood the *Patel-Topholm-Williamson* combination to disclose or suggest the processor of the

head-mounted device ("wherein the earpiece processor") estimates a noise level of a digitized microphone signal representing environmental sounds ("calculates a noise level based upon a first microphone signal from the first microphone") to select the appropriate ESNRA filtering to be applied to the signal.

b. *[2.2] wherein the earpiece processor modifies the mixed signal based upon the noise level.*

264. *Patel* in combination with *Topholm* and *Williamson* discloses or suggests this feature.

265. As detailed in §X.A.1.h, the *Patel-Topholm* combination discloses or suggests generating a mixed signal by introducing environmental sounds into the audio signal delivered to the user. *Patel* discloses that the ESNRAs are used to filter noise in the user's environment and that the filtered microphone signals may be provided to the user. *Patel* ¶[0067] ("With respect to incoming signals, the ESNRAs may be applied to … to filter noise from the user's environment. Thus, the user can benefit … by filtering environmental noise present in the user's location."); *see also id*. claims 22, 23, 32 (reciting a head-mounted device with "a microphone; wherein the processor is operable to receive an audio signal from the microphone, is operable to filter the audio signal by using the selected ESNRA, and is operable to transmit a filtered audio signal to a user via the speaker.").

266. A POSITA would have understood *Patel* to disclose that the ESNRA filter is applied to the environmental sound that is provided to the user while in a communication mode, i.e., to the environmental sound that is mixed with the audio signals from the external device, because *Patel*'s disclosure of filtering the environmental sounds to the user is not limited to when the head-mounted device is in a particular mode. *Patel* also explains that "the ESNRAs could be used to reduce noise from both the user's location and the other party's location at the same time" (*Patel* ¶[0067]), i.e., reduce noise in the received communication signal and in the microphone signal containing environmental sounds at the same time.

267. Thus, a POSITA would have understood that, in the *Patel-Topholm-Williamson* combination, the ESNRA-filtered microphone signal representing environmental sounds would be introduced into (mixed) into the audio content signal delivered to the user. A POSITA would have understood that this would have had the effect of modifying the mixed signal based on the noise level. More particularly, as the ESNRA filter is adapted according to *Williamson*'s noise-level adaptation teaching, this would have the effect of modifying the mixed signal because the portion of the mix corresponding to the environmental sounds was modified.

268. Accordingly, a POSITA would have understood the *Patel-Topholm-Williamson* combination to disclose or suggest the processor of the head-mounted device ("wherein the earpiece processor") applies the ESNRA to filter that was selected based on the noise level to the environmental sounds introduced into the audio signal from the external device ("modifies the mixed signal based upon the noise level").

C. Ground 2A: *Melanson* and *Rosenberg* Render Obvious Claims 8, 19, and 20

1. Claim 8

a. *[8.pre]: A device that can control ambient sound passthrough comprising:*

269. To the extent the preamble is limiting, *Melanson* discloses or suggests this feature. For example, *Melanson* discloses a "personal audio playback device that alters the gain of playback program material in response to environmental sounds." *Melanson*, Abstract. *Melanson* discloses implementing the playback device with "one or more microphone elements" to "sense environmental (ambient) sounds." *Id*., 2:17-24.

270. *Melanson* further discloses that the playback device has a "Personal Safety Mode" in which, upon detection of certain sounds, "[t]he microphone input can be provided at the headphone output and/or mixed with the program audio, so that the user is made more aware of the external environment while the sound is present." *Id*., 7:10-24; *see also id*., 8:52-60 ("If an event responsive

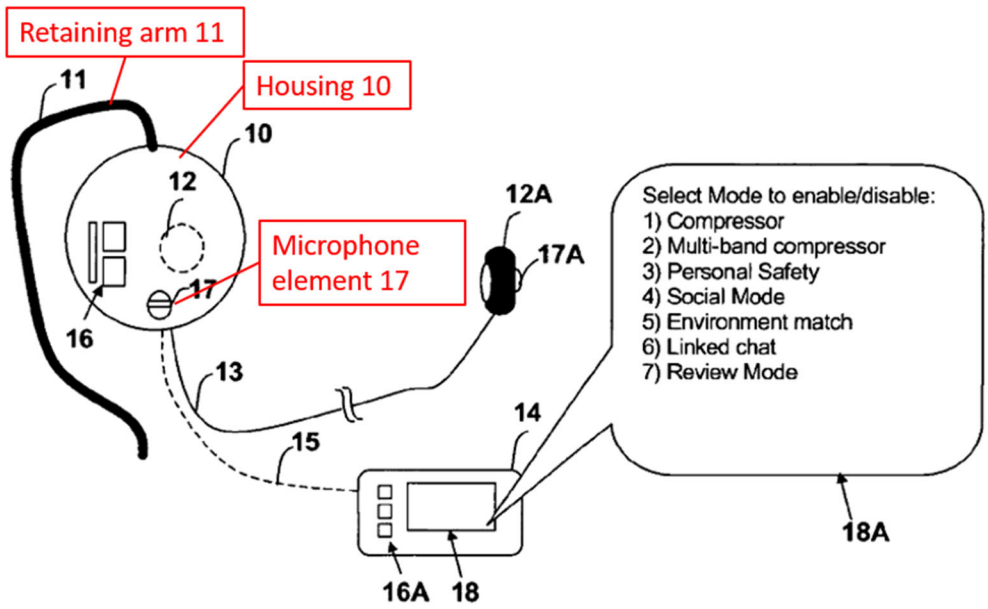
mode is selected (decision 57), then the device will respond to an ambient audio or wireless event by attenuating, pausing or discontinuing playback of the program material and replacing it in the audio output with event-related material (step 58). For example, … in social and personal safety modes, the program material is replaced with the sounds received by the device microphone(s).”). A POSITA would have understood that the “microphone input” referenced in the discussion of personal safety mode corresponds to the sensed “environmental (ambient) sounds” because *Melanson* discloses that the microphones are “provided to sense environmental (ambient) sounds.” *Id*., 2:19-21.

271. *Melanson*’s disclosure is consistent with how the ’473 patent describes ambient sound passthrough. *’473pat.*, 7:28-31 (“If at step 310, an acute sound is detected within the ambient sound, the earpiece 100 activates ‘sound pass-through’ to reproduce the ambient sound in the ear canal by way of the ECR 125.”).

272. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest a playback device that outputs ambient sounds to a user upon detection of certain sounds (“[a] device that can control ambient sound passthrough”).

b. *[8.1]: a first microphone that measures a first acoustic environment to generate a first microphone signal;*

273. *Melanson* discloses or suggests this feature. *Melanson* discloses a playback device in Figure 1 including a housing 10, a "retaining arm 11" for "retaining device housing 10 to a wearer's ear," and a "microphone element 17." *Melanson*, 3:34-43. The microphone element 17 "is disposed within housing 10 and acoustically coupled through the surface of housing 10 to detect environmental sounds." *Melanson*, 3:40-43; *see also id.*, FIG. 1. Figure 1 (annotated) shows microphone element 17:



Melanson, FIG. 1 (annotated)

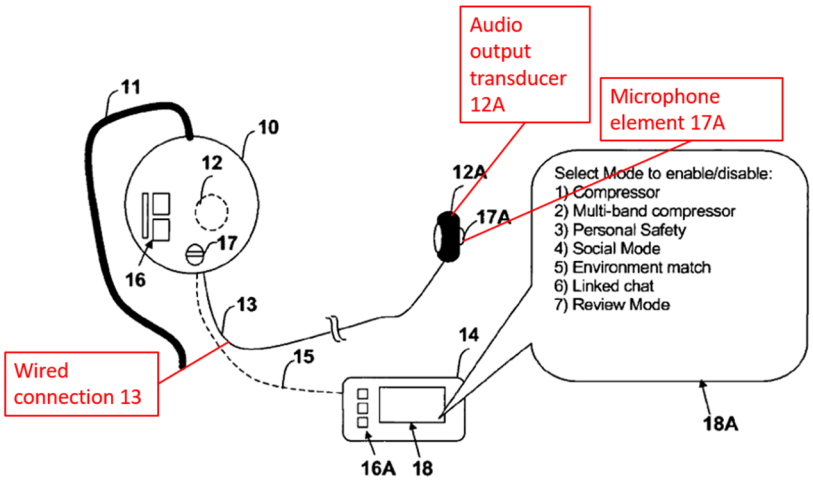
274. *Melanson* discloses that the "the processing circuit receives a signal from the microphone elements and adjusts the gain of the program material in

conformity with the microphone signal." *Id*., 2:19-24; *see also id*., 4:49-63. *Melanson*'s claims further describe the microphone as "producing an ambient measuring audio signal." *See id.*, 9:12-13 ("a microphone input for connection to a microphone element for sensing ambient sounds external to said housing and ***producing an ambient measuring audio signal***") (emphasis added), 10:56-58, 11:32-34, 12:19-21, 12:46-49. Because microphone element 17 detects environmental sounds and produces an "ambient measuring audio signal," a POSITA would have understood that microphone element 17 "measures a first acoustic environment to generate a first microphone signal."

275. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest a microphone 17 ("a first microphone") that detects environmental sounds and produces an ambient measuring signal ("that measures a first acoustic environment to generate a first microphone signal").

c. *[8.2]: a second microphone that is positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone generates a second microphone signal;*

276. *Melanson* discloses or suggests this feature. For example, *Melanson* discloses that housing 10 has an output transducer 12, and that it is connected by wire 13 to another output audio transducer 12A with "another microphone element 17A." *Melanson*, 3:49-51. Figure 1 (annotated) shows microphone element 17 and microphone element 17A:



Melanson, FIG. 1 (annotated)

277. As noted above, the retaining arm 11 connected to housing 10 is for retaining the housing on a wearer's ear. *Melanson*, 3:36-37. A POSITA would have understood that output transducer 12 and output transducer 12A were designed to be worn on different ears because a POSITA would have understood that each output transducer would be inserted in different ears. And, because output transducer 12A and microphone 17A are connected by wire 13 to housing 10, output transducer 12A microphone element 17A would be positioned on the ear opposite housing 10 and microphone element 17. Thus, a POSITA would have understood that microphone 17A is closer to one of the user's ears than microphone element 17 when the device is worn.

278. A POSITA would have further understood that each ear of the user has an ear canal. Thus, because microphone element 17A is closer to one of the user's ear canals, a POSITA would have understood that microphone element

17A "is positioned closer to a user's ear canal than the first microphone when the device is worn." This is consistent with the '473 patent's description that the alleged invention "can be implemented in a single earpiece, ***a pair of earpieces***, ***headphones***, or other suitable headset audio delivery devices." *'473pat.*, 6:37-39 (emphasis added); *see also id*., 9:3-5, 10:7-10, 10:50-52, 11:21-23, 11:48-51, 12:21-24, 13:16-19.

279. This interpretation is also consistent with how the Patent Office interpreted similar claim language during prosecution of U.S. Patent 11,244,666 (the "'666 patent") – a parent to the '473 patent that shares a specification with the '473 patent. During prosecution of the '666 patent, the pending claims recited:

> An earphone comprising:
> a first microphone configured to measure an ambient acoustic environment,
> a second microphone configured to measure an acoustic environment closer the [*sic*] ear canal of a wearer than that measured by the first microphone
> …

EX1008 at 157.

280. The examiner rejected the claims and mapped this claim language to the *Terlizzi* reference as follows:

> Terlizzi et al. teaches an earphone arrangement (the specific circuit arrangement of which can be seen in figure 2 for blending signals, which as discussed with respect to figure 9 can be included in the earpiece) comprising: a first microphone configured to measure an ambient acoustic environment (2250, capable of picking up ambient sounds); a second microphone (2106) configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone (note locations of microphones 2106 and 2250 are used at different locations relative to a user's ear and therefor ear canal)

EX1008 at 118-119.

281. I have annotated *Terlizzi*'s Figure 2, below, which confirms that the Patent Office mapped the claimed "first microphone" to microphone 2250 on the audio generation device 2200 and the claimed "a second microphone configured to measure an acoustic environment closer the [*sic*] ear canal of a wearer" to microphone 2106 on the listening device 2100 (i.e., headphones).

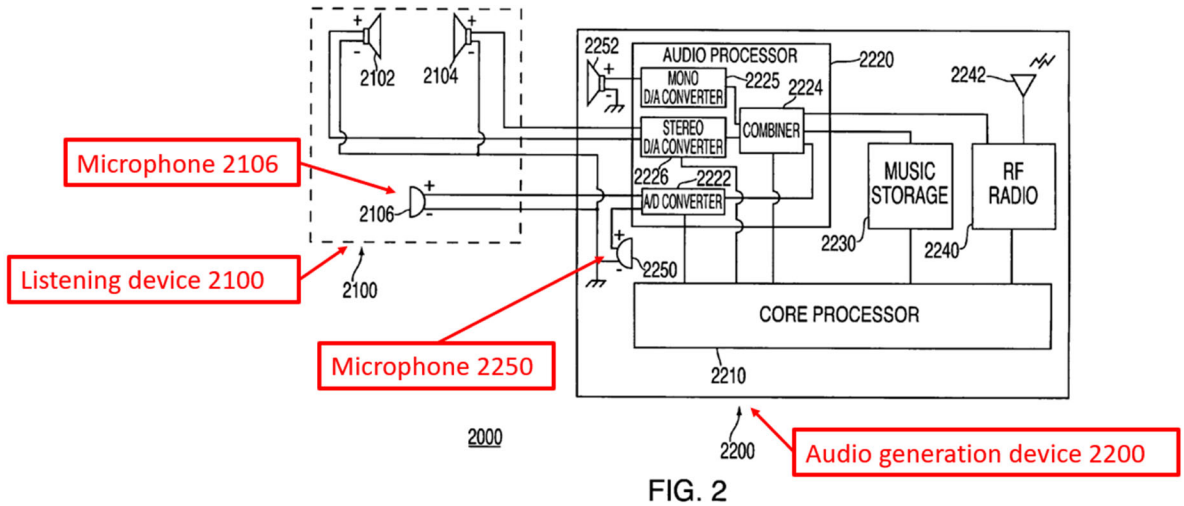


FIG. 2

EX1044, FIG. 2 (annotated)

282. In response to the examiner's rejection, the applicant amended the claims as show below:

Amended claim 1 currently states:

1.(currently amended) An earphone comprising:

a first microphone configured to measure an ambient acoustic environment, wherein the first microphone has a first microphone port that is configured to face away from a user when the earphone is inserted;

a second microphone configured to measure an acoustic environment closer the ear canal of a wearer than that measured by the first microphone, wherein the second microphone has a second microphone port that is configured to face toward the user when the earphone is inserted;

EX1008 at 108. The applicant explained that the amendment was meant to clarify that the "second microphone" was "an ear canal microphone (ECM)." *Id.* at 112.

283. The Patent Office's interpretation and the applicant's subsequent amendment support my interpretation of the "first" and "second" microphone of the '473 patent is accurate because the claim amendments that were added to the claims of the '666 patent to limit the "second microphone" to an ear canal microphone are not present in the claims of the '473 patent.

284. As discussed in §X.C.1.b, a POSITA would have understood *Melanson* to disclose that microphone elements 17 and 17A provide one or more electrical signals corresponding to environmental sounds. Thus, a POSITA

would have understood that microphone element 17A generates a second microphone signal.

285. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest a microphone element 17A ("a second microphone") positioned on the ear opposite microphone element 17 when the device is worn ("that is positioned closer to a user's ear canal than the first microphone when the device is worn") and that provides an electrical signal corresponding to environmental sounds ("where the second microphone generates a second microphone signal").

d. *[8.3]: a speaker;*

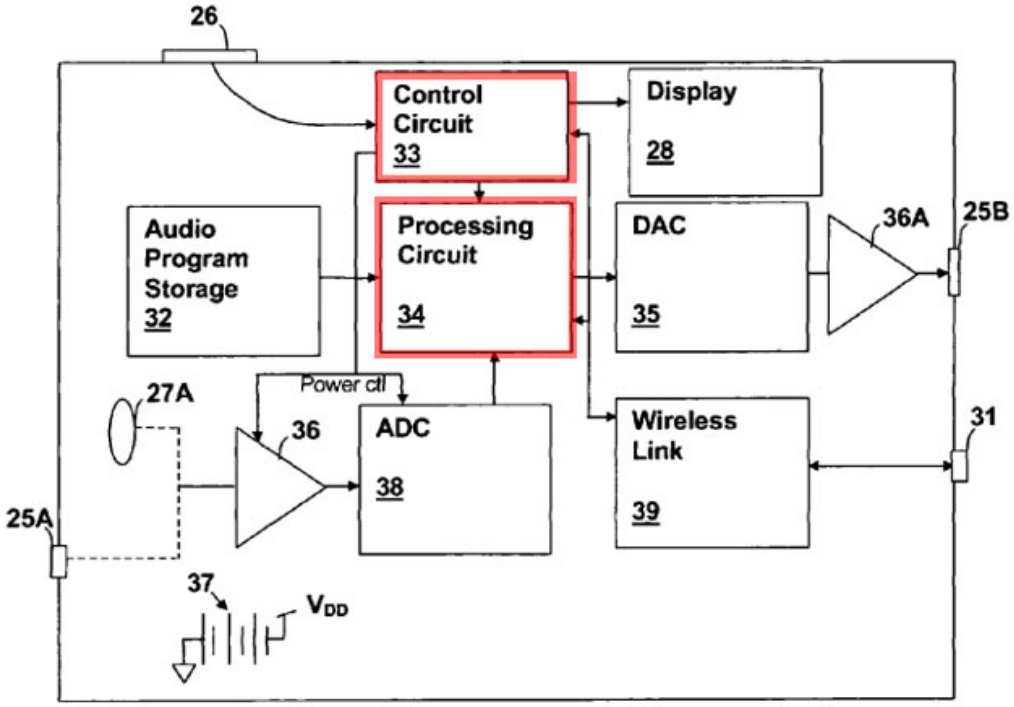
286. *Melanson* discloses or suggests this feature. For example, *Melanson* discloses that "[a]n output audio transducer 12 is included on housing 10 to provide audio program information to the wearer from program media stored within housing 10." *Melanson*, 3:38-40; *see also id.*, FIG. 1. This is consistent with how the '473 patent describes a "speaker." *'473pat.*, 5:18-20 ("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.").

287. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest an output audio transducer 12 ("a speaker").

e. *[8.4] and [8.5]: a memory that stores instructions; and a processor that executes the instructions to perform operations, the operations comprising:*

288. *Melanson* discloses or suggests this feature. For example, with reference to Figure 3, *Melanson* discloses that the playback device includes a "control circuit 33" and a "processing circuit 34" and that they may be implemented together as a single "digital signal processor with data and program memory." *Melanson*, 5:14-24. A POSITA would have understood that "program memory" refers to memory that stores programs comprised of software instructions. EX1034 (*Computer Dictionary*).[12]

[12] To be clear, I am relying on *Computer Dictionary* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2A.



Melanson, FIG. 3 (annotated)

289. A POSITA would have understood that *Melanson*'s description of Figure 3 applies to the playback device in Figure 1 because *Melanson* describes Figure 3 as pertaining to "the ***above-described*** consumer personal audio playback device**s**," (*Melanson*, 4:31-33 (emphasis added)), referring to both devices (plural) described "above" with reference to Figures 1 and 2.

290. *Melanson* generally discloses that "[i]ntelligent algorithms are provided in the processing circuit to analyze the environmental sounds and adjust the gain of the headphone outputs in one or more frequency bands in conformity with the results of the analysis." *Id*., 3:29-33. *Melanson* further discloses implementing its various playback modes with "a signal processing

algorithm according to program instructions stored within a memory of processing circuit 34." *See, e.g.*, *id*., 7:5-9 ("As mentioned above, the multi-band compression mode (or the single-band compression mode) can be implemented by a signal processing algorithm according to program instructions stored within a memory of processing circuit 34 or by dedicated circuits."), 7:10-13 (same with respect to "Personal Safety Mode"), 7:25-31 (same with respect to "social mode").

291. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest a control circuit 33 and processing circuit 34 implemented as a digital signal processor with a program memory that executes program instructions to analyze environmental sounds and adjust the gain of the headphone outputs ("memory that stores instructions and a processor that executes the instructions to perform operations").

f. *[8.6]: receiving an audio content signal;*

292. *Melanson* discloses or suggests this feature.

293. *Melanson* discloses that audio program storage 32 includes "the audio portion of video program material or computer program-generated audio sources such as game or Musical Instrument Device Interface (MIDI) information playback, provides the program source material." *Id.*, 4:33-38. This is consistent with how the '473 patent describes "audio content." *'473pat*., 5:59-

62 ("The earpiece 100 can include an audio interface 212 operatively coupled to the processor 206 to receive audio content, for example from a media player or cell phone, and deliver the audio content to the processor 206.").

294. *Melanson* explains that "[p]rocessing circuit 34 performs gain control and rendering of the program information ***retrieved from storage 32*** and provides the result to a digital-to-analog converter (DAC) 35 that converts the processed program information to analog form that is provided to a headphone amplifier 36A." *Melanson*, 4:41-48 (emphasis added); *see also id.*, FIG. 3, 2:15-19. A POSITA would have understood that to perform gain control and rendering of program information from storage 32, the program information is retrieved as a signal representing the program information.

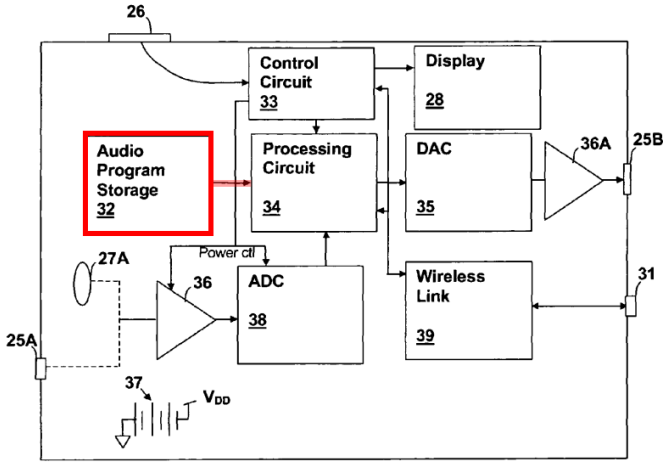


Fig. 3

Melanson, FIG. 3 (annotated)

295. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest processing circuit 34 retrieving a signal representing the audio portion of video program material or computer program-generated audio sources from storage 32 ("receiving an audio content signal").

g. *[8.7]: modifying the audio content signal to generate a modified audio content signal;*

296. *Melanson* discloses or suggests this feature. For example, *Melanson* discloses a "compressor mode" (*Melanson*, 5:41-64) and a "multi-band compressor mode" (*id.*, 5:65-7:9). In these modes, *Melanson* discloses adjusting the gain of the program audio (i.e., the audio content signal) based on the loudness of external sounds. *Id.*, 5:45-51 ("In the most basic 'compressor' mode, selected to overcome environmental sounds in general, the gain of low levels of program audio is adjusted upward in conformity with the detected loudness of the external sounds. Processing circuit 34 acts as an electronic volume control that increases the gain of the program audio as ambient noise increases."), 6:4-13 ("The detected levels of loudness are used to control compression of the program material gain in multiple frequency bands by splitting the program material into corresponding frequency bands and independently controlling the gain of the signal provided in each band to DAC 35. The loudness of the program material in each frequency band can also be detected, and a combination of the loudness of the program material and the detected

environmental noise level in each band can be used to control the gain of the signal supplied to DAC 35 in each frequency band."). A POSITA would have understood that adjusting the gain of the program audio modifies the signal because changing the gain of the signal modifies the signal.

297. Accordingly, a POSITA would have understood *Melanson* to disclose or suggest a compressor mode or multi-band compressor mode in which the gain of program audio is adjusted based on environmental noise levels ("modifying the audio content signal to generate a modified audio content signal").

h. *[8.8]: modifying the first microphone signal to generate a modified first microphone signal; [8.9]: modifying the second microphone signal to generate a modified second microphone signal;*

298. *Melanson* in combination with *Rosenberg* discloses or suggests this feature.

299. As explained above in §§X.C.1.b-X.C.1.c, *Melanson* discloses a processor receiving first and second microphone signals from microphone elements 17 and 17A. And as explained in §X.C.1.a, *Melanson* generally discloses detecting sounds such as "sudden changes in volume, or particular frequency patterns such as vehicle horns or loud voices" detected in the ambient sound picked up by the microphones and passing through the ambient sound to the user in response.

300. *Rosenberg* discloses an ambient sound responsive media player similar to *Melanson*. For example, *Rosenberg* similarly discloses mixing an audio content signal such as music with ambient sound if particular sounds or "characteristic forms" are detected in the ambient sound. *Rosenberg* ¶[0013].

301. *Rosenberg* provides additional details related to the processing of ambient microphone signals prior to detecting sounds in the signals. In particular, *Rosenberg*'s media player includes a microphone to capture ambient sound which, like *Melanson*, may be implemented as a "plurality of microphone elements." *Rosenberg* ¶¶[0014], [0039]. *Rosenberg* explains that the "ambient audio signals are generally captured as analog signals from the microphone element and then are digitized through an analog to digital conversion process." *Id*. ¶[0044]. In addition, *Rosenberg* explains that "noise reduction, filtering, and/or other commonly known signal processing steps may be performed upon the ambient signal." *Id.* ¶[0044].

302. *Rosenberg* further explains that "[t]he ambient audio signals, once converted to a final digital form, are generally stored in a temporary local memory of the portable media player" where it is accessible by the steps of *Rosenberg*'s algorithm, including for the purpose of sound recognition processing, speech recognition processing, and/or vocal identity recognition processing. *Id.* ¶¶[0044]-[0045].

303. A POSITA would have understood that performing such signal processing (e.g., filtering and/or noise reduction) on the ambient signals modifies the signals. Thus, in view of the above disclosures, a POSITA would have understood *Rosenberg* to disclose performing noise reduction, filtering, and/or other commonly known signal processing to generate modified microphone signals. A POSITA would have been motivated to implement *Rosenberg*'s teachings related to performing such common signal processing steps into *Melanson*'s playback device. A POSITA would have recognized, for example, that performing noise reduction and/or filtering of the ambient sound before analyzing it to detect, for example, "particular frequency patterns such as vehicle horns or loud voices," as taught by *Melanson*, would have increased the reliability of the detection. More particularly, a POSITA would have understood that pattern recognition algorithms are more prone to error if the signal being analyzed is too noisy, and so performing a noise-reduction and/or filtering operation on the signal would reduce the incidence of false recognition, leading to fewer errors in operation.

304. Therefore, a POSITA would have considered the combination to amount to the use of a known technique (performing noise reduction and/or filtering on the captured ambient sound signal prior to processing for sound/pattern detection) to improve similar devices (*Melanson* and *Rosenberg*

are both directed to processing ambient sound to detect particular sounds) in the same way (to increase the reliability and accuracy of *Melanson*'s sound detection processing).

305. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest processing *Melanson*'s first and second microphone signals by performing noise reduction and/or filtering on the signals ("modifying the first microphone signal to generate a modified first microphone signal; modifying the second microphone signal to generate a modified second microphone signal").

i. *[8.11]: detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;*

306. *Melanson* in combination with *Rosenberg* discloses or suggests this feature.

307. *Melanson* discloses a personal safety mode in which the audio provided to the user is changed when "sudden noises," "sudden change in volume" are detected in the microphone signals. *Melanson*, 7:10-24 ("In personal safety mode, sudden changes in volume, or particular frequency patterns such as vehicle horns or loud voices, are detected and processing circuit 34 either partially or completely attenuates the program material gain, so that a user of the personal audio playback device will hear the outside sounds."); *see*

also id., 2:34-37 ("The gain of the program material may be reduced when certain sounds are recognized, such as sudden noises, vehicle horns or a person attempting to talk to a user of the device."), 1:38-41 ("As another example, a pedestrian listening to audio via such a device may not notice a vehicle horn or siren that is alerting them to a hazard.").

308. *Melanson*'s disclosure of detecting "sudden noises" or a "sudden change in volume" is consistent with how the '473 patent describes detecting an "acute sound." *'473pat.*, 5:43-50 ("The processor 206 can monitor the ambient sound captured by the ASM 111 for acute sounds in the environment, such as an abrupt high energy sound corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., 'help', 'stop', 'police', etc.), or specific noise type (e.g., breaking glass, gunshot, etc.)."), 7:48-61 ("Notably, the processor 206 can search for on-sets (e.g., fast rising amplitude wave-front) of an acute sound or other abrupt feature characteristics without initially attempting to initially identify or recognize the sound source. That is, the processor 206 is actively listening for a presence of acute sounds before identifying the type of sound source."), 9:45-49 ("Briefly, if a transient, high-level sound (or acute sound) is detected in the ambient sound signal (ASM input signal), then it is converted to a level, and its magnitude compared with the BNL is calculated."), 16:5-8

("sending a notification signal to the speaker if the acute sound is a warning sound or siren").

309. As explained above in §X.C.1.h, *Rosenberg* discloses processing the signals captured from the microphone elements before processing them to identify sounds. Thus, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest processing the "modified" first and/or second microphone signals to detect the acute sound.

310. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest detecting sudden noises or sudden changes in volume ("detecting an acute sound") by analyzing a microphone signal that had been processed (e.g., filtered or noise reduced) ("by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof").

j. *[8.12]: modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal;*

311. *Melanson* in combination with *Rosenberg* discloses or suggests this feature.

312. As explained in §X.C.1.g, *Melanson* discloses a "compressor mode" and "multi-band compressor mode" that modifies the audio content signal based on the background noise level. As explained below, *Melanson*

discloses implementing a further modification to the audio content signal if an acute sound is detected.

313. *Melanson* discloses that its various playback modes, including the compressor modes and, for example, personal safety mode, are "not mutually exclusive" and "can be implemented simultaneously" and "selectively enabled." *Melanson*, 8:29-39. Thus, a POSITA would have understood *Melanson* to disclose that, for example, the multi-band compressor mode and personal safety mode may be enabled at the same time. In that scenario, the compressor mode performs an initial modification of the audio content signal, as explained in §X.C.1.g. Then, according to personal safety mode, when a sudden noise or change in volume ("acute sound") is detected, the "processing circuit 34 either partially or completely attenuates the program material gain." *Melanson*, 7:16-19. A POSITA would have understood that partially attenuating the program material gain modifies the audio content signal because it lowers the gain of the signal.

314. Thus, *Melanson* discloses a two-step modification process—first the audio content signal is modified by the compressor mode and then modified again by the personal safety mode. The two-step modification is illustrated in *Melanson*'s Figure 4, annotated below. In step 56, the compressor mode adjusts

the gain of the program material ("audio content signal"), and in steps 57 and 58, the program material is attenuated if an event is detected.

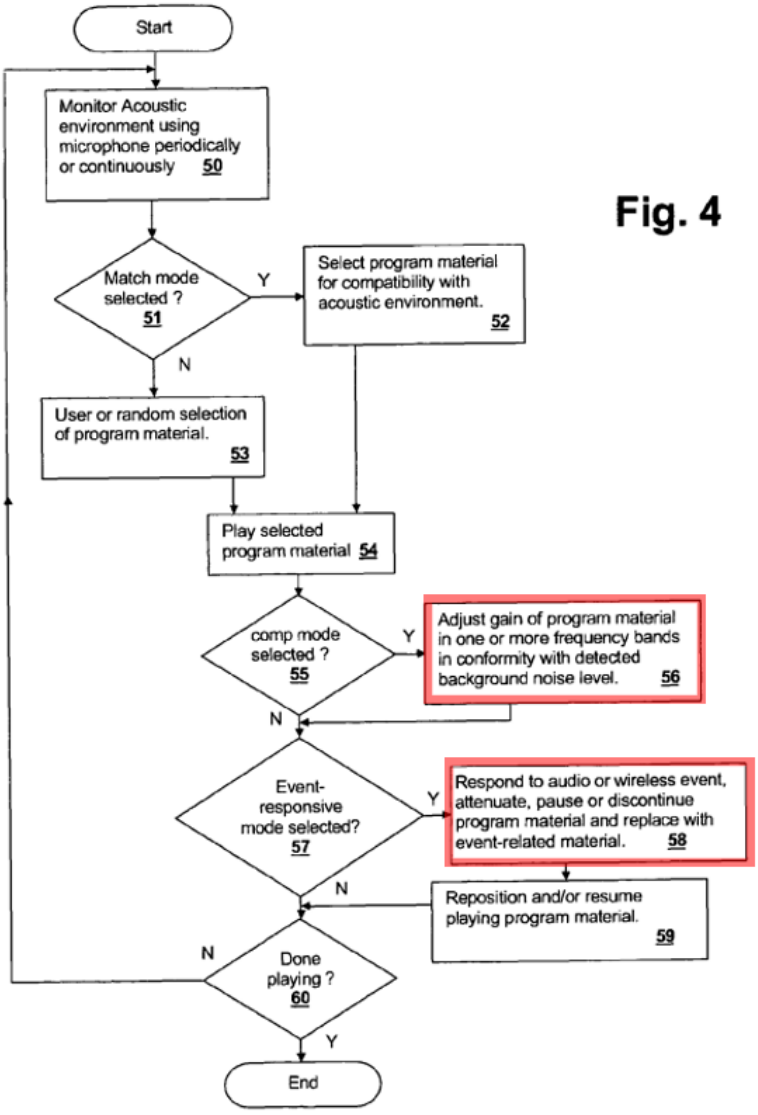


Fig. 4

Melanson, FIG. 4 (annotated)

315. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest partially attenuating the gain of the compressor-mode-modified program material ("modifying the modified audio content signal") if a sudden noise or particular frequency pattern is

detected ("if an acute sound is detected"), which yields an attenuated version of the compressor-mode-modified program material ("to generate a new modified audio content signal").

k. ***[8.10]: generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;***

316. *Melanson* in combination with *Rosenberg* discloses or suggests this feature.

317. *Melanson* generally discloses mixing the input from the microphone with program audio so that the user can be more aware of the environment and to facilitate conversations with others in noisy environments. *See generally Melanson*, 1:13-2:8. As discussed in §X.C.1.h, *Rosenberg* discloses an ambient sound responsive media player similar to *Melanson*. *Rosenberg* provides additional details related to the processing of ambient microphone signals for mixing with audio content and output to the user. *See Rosenberg* ¶¶[0054]-[0058]. In particular, *Rosenberg* explains that after the ambient microphone signals are digitized and processed as explained above in §X.C.1.h, "the media player is operative to mix musical audio content derived from a stored media file with ambient audio content captured from a microphone local to the user." *Id*. ¶[0054]; *see also id*. ¶[0013].

318. *Rosenberg* further notes that, in the mixed signal, "the ambient audio signal content may be filtered or otherwise processed to extract extraneous noise and/or sound content that is outside certain magnitude and/or frequency limits or thresholds." *Id*. ¶[0054].

319. In reading *Rosenberg* as a whole, a POSITA would have understood *Rosenberg* to disclose or suggest that the signal processing done on the ambient sound signal for mixing (described in *Rosenberg* paragraph [0054]) is ***in addition*** to the signal processing described in the context of capturing the ambient signals (described in *Rosenberg* paragraph [0044]). That is, a POSITA would have understood *Rosenberg* to disclose or suggest that the "modified first microphone signal" and "modified second microphone signal" generated by the processing described in paragraph [0044] are further processed before being mixed and output to the user.

320. *Rosenberg* explains that, after the "commonly known signal processing" is performed on the ambient signals, they are stored in a "shared memory space accessible by the steps of this program flow," i.e., the steps of the method shown in Figure 2. *Rosenberg* ¶[0044]. A POSITA would have understood *Rosenberg*'s teaching in paragraphs [0054] and [0055] relating to the mixing embodiment build on and modify the program flow of Figure 2 to include additional steps for implementing mixing because mixing is an

alternative technique to the merely reducing the volume of the audio content discussed in the context of Figure 2. Thus, where *Rosenberg* discloses in paragraph [0055] that the ambient audio signal may be processed, a POSITA would have understood *Rosenberg* to disclose that the ambient signal is accessed in the shared memory space, processed (e.g., to extract extraneous noise and/or sound content that is outside certain magnitude and/or frequency limits or thresholds), and then mixed with the audio content signal.

321. A POSITA would have understood that such further processing "generates an ambient sound signal from at least one of the modified first microphone signal [or] the modified second microphone signal" because the additional processing further modifies the modified first or second microphone signal into an ambient sound signal output to the user.

322. A POSITA would have been motivated to incorporate *Rosenberg*'s teaching related to further processing the ambient sound signal prior to mixing and outputting it to the user into *Melanson* and would have had a reasonable expectation of success. A POSITA would have understood that such processing was advantageous because, for example, it would provide the user with an ambient sound signal with a higher signal-to-noise ratio, allowing the user to hear more of the signal that they want to hear (e.g., the voice of the person to whom they are speaking) and less of other background noise. A POSITA would

have understood that this would have advanced *Melanson*'s objective to facilitate conversations in noisy environments.

323. Thus, a POSITA would have understood that the combination would have amounted to the use of a known technique (performing additional processing on the ambient sound signal prior to mixing and output to the user) to improve similar devices (*Melanson* and *Rosenberg* are both directed to mixing ambient sound with audio content) in the same way (to increase the signal-to-noise ratio of the ambient sound output to the user).

324. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest performing additional processing on the ambient sound signal ("generating an ambient sound signal from at least one of the modified first microphone signal [or] the modified second microphone signal").

l. *[8.13]: generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal; and [8.14] sending the mixed signal to the speaker*

325. *Melanson* in combination with *Rosenberg* discloses or suggests this feature.

326. *Melanson* discloses that, in personal safety mode, "sudden changes in volume, or particular frequency patterns such as vehicle horns or loud voices, are detected and processing circuit 34 either partially or completely attenuates

the program material gain, so that a user of the personal audio playback device will hear the outside sounds." *Melanson*, 7:10-24. *Melanson* also discloses that the "microphone input can be … mixed with the program audio, so that the user is made more aware of the external environment while the sound is present." *Id.* A POSITA would have understood that the "headphone output" refers to the output transducer (or "speaker"), and that this is how the user is "made more aware of the external environment."

327. As explained in §X.C.1.k, the *Melanson-Rosenberg* combination discloses or suggests generating an ambient sound signal for mixing with the audio content signal. And as explained in §X.C.1.k, the *Melanson-Rosenberg* combination discloses or suggests generating a new modified audio content signal. Thus, a POSITA would have understood that the *Melanson-Rosenberg* combination discloses or suggests a mixed signal that comprises the "ambient sound signal" and the "new modified audio content signal."

328. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest mixing an ambient sound signal with the new modified audio content signal ("generating a mixed signal by mixing the ambient sound signal with … the new modified audio content signal") and providing it at the headphone output ("sending the mixed signal to the speaker").

2. Claim 19: The device according to claim 8, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a user's voice is detected.

329. *Melanson* in combination with *Rosenberg* discloses or suggests this claim. *Melanson* discloses a "social mode" in which "speech of another person is detected and the program material is attenuated, so that a user of the personal audio playback device can converse with another person." *Melanson*, 7:25-31. *Melanson* further states that "[a] timer can be used to restore the program material after a period ***without speech by*** either ***the user*** or the other person ***is detected***." *Id.*, 7:34-37 (emphasis added). Thus, a POSITA would have understood *Melanson* to disclose detecting speech of the user.

330. *Rosenberg*, for a device similar to *Melanson*'s, expressly discloses modification of the audio output if a user's voice is detected, including by mixing the ambient sound with audio content and decreasing the volume of the audio content. In particular, *Rosenberg* discloses "mixing at least a portion of the ambient audio signal with a first output of a media player to generate a second output of the media player if the one or more characteristic forms are present within the ambient audio signal." *Rosenberg* ¶¶[0013], [0026] ("In some embodiments ambient sounds from the local environment are selectively mixed with digital media sounds such that their relative volumes are adjusted based at least in part upon detected ambient sound events within the user's local

environment."), [0057]; *see also id.* claim 19. *Rosenberg* discloses that one of the "characteristic forms" that may be detected is "the unique voice of the media player user." *Id*. ¶¶[0046], [0047], [0055], claim 21.

331. A POSITA would have been motivated to implement *Rosenberg*'s teaching related to modifying the audio output when the user's voice is detected into *Melanson*'s social mode and would have had a reasonable expectation of success in doing so. One of *Melanson*'s objectives is to facilitate conversation in noisy environments, and a POSITA would have understood that incorporating *Rosenberg*'s user-voice detection technique into *Melanson* to improve *Melanson*'s social mode would have advanced that objective by providing the user the ability to trigger social mode with his/her own voice. And as explained by *Rosenberg*, this would allow the "media player user to spontaneously begin [to] engage in a conversation and not talk too loud, because he or she can more easily hear himself or herself while talking." *Rosenberg* ¶[0061].

332. Moreover, techniques for uniquely identifying the user's voice were well-known, and *Melanson* already discloses the ability to detect the user's voice (*Melanson*, 7:34-38), so it would have been within the skill of a POSITA to modify *Melanson* in this way. Nor would the modification have frustrated *Melanson*'s objectives or purposes.

333. I note that *Melanson* also discloses a "Linked Chat Mode":

> Linked Chat Mode Similar to conversation mode, in chat mode, also implemented by a signal processing algorithm according to program instructions stored within a memory of processing circuit **34** or a dedicated circuit, speech of the user (wearer) is detected and transmitted to another device via wireless link **39**. The program material is paused or attenuated for both devices until a timer has expired and the other device can transmit audio information directly from the other device's microphone circuits to the device of FIG. 3, so that the other device-user's speech is sent to headphone amplifier **36**A and connected headphone elements.

Melanson, 7:50-60.

334. A POSITA would have understood that modifying *Melanson*'s "social mode" to be triggered upon detection of the user's voice would not have caused a design conflict or otherwise render inoperable *Melanson*'s other modes, including linked chat mode. First, a POSITA would have understood *Melanson*'s linked chat mode to require a wireless link with another device, so there would be no conflict with social mode, which does not involve a wireless link with another device. In other words, a POSITA would have understood that Linked Chat mode would be triggered by two conditions – both the user's voice and a wireless link to another device, whereas social mode would be triggered just by detection of the user's voice.

335. Second, *Melanson* discloses that the various modes can be "selectively enabled" at the same time, which suggests that less than all of the modes need to be enabled. Thus, even if the *Melanson-Rosenberg* modified "social mode" created a design conflict with "Linked Chat" mode, a POSITA would have understood to further modify the *Melanson-Rosenberg* combination so that "social mode" and "linked chat mode" could not be simultaneously enabled. Having to make that modification would not have dissuaded a POSITA from making the combination. A POSITA would have recognized the benefits to an improved "social mode" design (i.e., a social mode that was triggered by detection of the user's voice) would outweigh the drawback of not being able to have "social mode" and "linked chat mode" on at the same time. *Melanson* does not describe the ability to have all modes simultaneously enabled as a key or critical aspect of its disclosed system.

336. Thus, a POSITA would have understood that the combination would have amounted to using a known technique (modifying audio output based on user's voice detection) to improve similar devices (*Melanson* and *Rosenberg* are both directed to modifying audio output in response to detection of certain events) in the same way (to facilitate conversations with others in noisy environments).

337. *Melanson* discloses that when "social mode" is triggered, the program material may be attenuated. *Melanson*, 7:25-31; *id*, 8:1-9, 8:52-56, FIG. 4 (block 58). A POSITA would have understood that where *Melanson* discloses "attenuat[ing]" the program material, it means to attenuate or decrease the gain of the program material. *See id.*, 2:37-41 (equating "decreas[ing]" gain with "attenuat[ing]" gain).

338. A POSITA would have further understood that attenuating the gain of the program material would cause a decrease in the volume of the program material because the gain of the program material is directly related to the output volume of the program material (higher gain means higher volume, lower gain means lower volume). A POSITA would have understood that gain changes are applied to acoustical signals and these are perceived by the listener as changes in volume. This is corroborated by, for example, Patent Owner's statements to the Patent Office during prosecution of U.S. Patent Application No. 17/483,190, "[t]he term gain is a common acoustic DSP term, to refer to a value that is applied to the acoustic signal to either increase the volume (gain increases) or reduce the volume (gain reduces)." EX1045 at 15. It is also corroborated by *AudioCyclopedia*, which explains:

> In a feedback-type volume control, attenuation is controlled by the amount of feedback in the circuit. Gain of a closed-loop circuit is

$$A_c = A_o/(1 + \beta A_o) \qquad (16\text{-}50)$$

where,
A_c is the closed-loop gain,
A_o is the open-loop gain,
β is the feedback factor.

By varying β, the amplifier gain can be changed, hence, giving the effect of a volume control.

EX1038 at 561.[13]

339. As explained in §X.C.1.j, because the compressor mode may be simultaneously engaged with social mode, and the compressor mode generates a "modified audio content signal," so when social mode is triggered, it operates to attenuate the gain of the "modified audio content signal."

340. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest attenuating the gain of the program material that was already processed by the compressor mode ("decreasing a volume of the modified audio content signal") when social mode is triggered by the user's voice ("when a user's voice is detected.").

[13] To be clear, I am relying on *AudioCyclopedia* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2B.

3. Claim 20: The device according to claim 8, wherein the acute sound is a voice.

341. *Melanson* in combination with *Rosenberg* discloses or suggests this claim. As discussed in §X.C.1.i, *Melanson* discloses that detecting a sudden noise or change in volume ("acute sound") can trigger personal safety mode. *Melanson* also discloses triggering personal safety mode upon detection of particular frequency patterns, including "loud voices." *Melanson*, 7:10-24. A POSITA would have therefore understood *Melanson* to disclose or suggest that the sudden noise or change in volume may correspond to a "loud voice."

342. Accordingly, a POSITA would have understood the *Melanson-Rosenberg* combination to disclose or suggest that the sudden noise or change in volume ("wherein the acute sound") may be a loud voice ("is a voice").

D. Ground 2B: *Melanson*, *Rosenberg*, *Samadani*, and *Dijkstra* Render Obvious Claims 9, 13-18

1. Claim 9: The device according to claim 8, wherein the operations further comprise: determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics of at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof.

343. *Melanson* in combination with *Rosenberg*, *Samadani*, and *Dijkstra* discloses or suggests this claim.

344. As context, a POSITA would have understood that claim 9 narrows independent claim 8 in two ways: first, it requires a step for determining the type

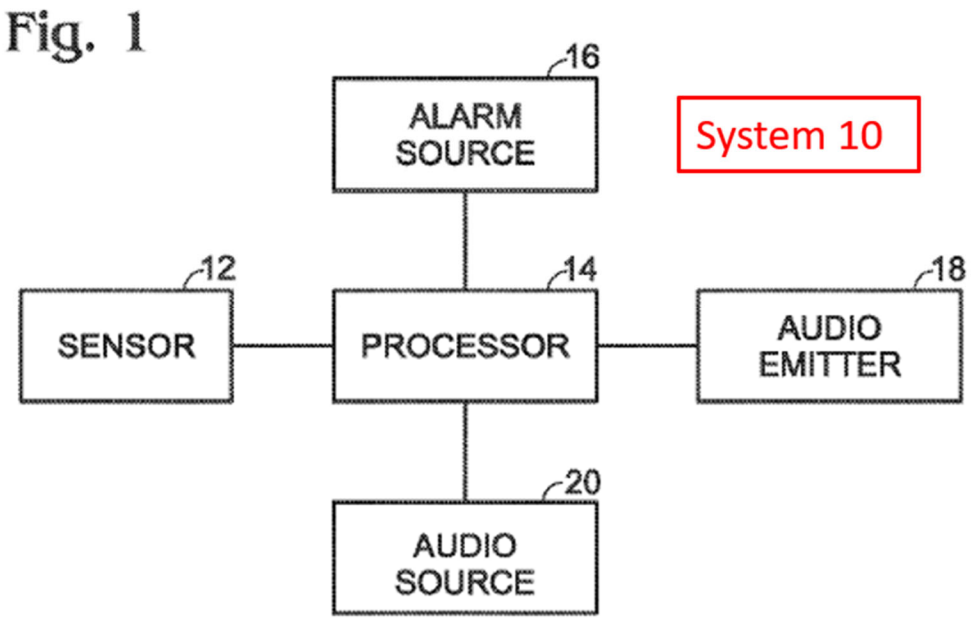
or source of the acute sound; and second, it specifically requires "determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics of at least one of the [modified microphone signals]." In other words, the claimed device must (1) detect an acute sound; and (2) determine the source or type of the acute sound (i.e., perform two-step acute sound detection), including by analyzing temporal and spectral characteristics to determine if it is the user's voice. As explained below, both of these aspects were known in the art and would have been obvious to incorporate into *Melanson*.

a. *Two-step acute sound detection*

345. *Melanson* discloses triggering personal safety mode upon detection of a sudden change in volume ***or*** "particular frequency patterns" such as "vehicle horns or loud voices." *Melanson*, 7:13-19. *Melanson* does not expressly disclose that personal safety mode is triggered by first detecting a sudden noise or change in volume ("acute sound") ***and then*** detecting the particular frequency pattern (i.e., the source of the sudden noise). Nevertheless, a POSITA would have considered it obvious to modify *Melanson* to implement that feature.

346. *Samadani* is similar to *Melanson* and discloses a two-step approach for triggering a notification to the user of hazards. *Samadani* discloses a system for a headset including microphones to monitor the external environment. *Samadani* ¶¶[0010]-[0011] ("System 10 includes a sensor 12. Sensor 12 is

configured to monitor the external environment surrounding the device user. According to one embodiment, sensor 12 may be or include one or more microphones configured to monitor external audio signals produced by the external environment. For the purposes of the present disclosure, the term "external audio signal(s)" refers to audio signals that can be heard in the environment surrounding the device user (though not necessarily by the user) and that are not produced by system 10 or any components thereof. Alternatively, sensor 12 may take the form of a video camera, sonar device, radar device, or any other device suitable for monitoring the proximity and velocity of various objects.")



Samadani, FIG. 1 (annotated)

347. *Samadani* explains that known devices do not differentiate between hazards that are approaching the user or not:

> A number of safety devices have been described that detect a signal that is emitted from a potentially hazardous object (such as a construction truck) and alert a device user when the user comes within a certain proximity of the object. However, the headphones are not able to alert the headphone wearer comes into proximity with objects that do not emit the signal. Moreover, these devices sound the alarm whether or not the hazardous object is approaching the device user.
>
> Other safety devices have been described that detect when a certain sounds, such as a siren. However, these safety devices do not determine whether or not the source of the sound is approaching the device user.

Id. ¶¶[0002]-[0003].

348. *Samadani* therefore discloses processing signals from the microphones to first detect a "signal of interest," and then determine if a "warning condition" exists. *Id.* ¶¶[0013]-[0014]; *see also id.* ¶¶[0023]-[0026], [0031]. *Samadani* discloses that the processor of the device "may be configured to measure the energy signature over time of each external audio signal and determine whether any of the monitored external energy signatures fit within the parameters of a signal of interest. … If an energy signature is determined to have

a value that is higher than the defined threshold, the signal could be identified as a signal of interest." *Id.* ¶[0026]. This is similar to *Melanson*'s disclosure of detecting a sudden noise or change in volume and is consistent with how the '473 patent describes detecting an "acute sound." *'473pat*., 5:43-50 ("The processor 206 can monitor the ambient sound captured by the ASM 111 ***for acute sounds in the environment, such as an abrupt high energy sound*** corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., 'help', 'stop', 'police', etc.), or specific noise type (e.g., breaking glass, gunshot, etc.)."); *see also id.*, 2:36-38, 3:2-7, 7:48-7:61, 9:45-49, 16:5-8.

349. *Samadani* further discloses that "once a signal of interest is identified, processor 14 could apply one or more filters or other processes to determine if the identified signal of interest indicates a warning condition." *Samadani* ¶[0031]. *Samadani* explains that skilled artisans would know of ways to "to filter out various types of noise and sounds from a given audio signal" so that "background noises and other uninteresting noises could be filtered out using any known means, either before or after signals are identified as being 'of interest.'" *Id.* ¶[0032]. *Samadani* itself discloses that "the spectral characteristics of an audio signal to filter out various unimportant signals" (*id*. ¶[0028]) and

suggests that the system "could be configured to detect the specific spectral characteristics of human speech" (*id*. ¶[0034]).

350. Thus, a POSITA would have understood *Samadani* to disclose a two-step approach for detecting a warning condition sufficient to alter output audio signals — first identify a signal of interest (e.g., by measuring the power signature over time) and then detect the specific spectral characteristics of the signal (e.g., by determining whether the signal includes human speech).

351. A POSITA would have been motivated to implement *Samadani*'s teaching of a two-step approach for detecting a warning condition in the *Melanson-Rosenberg* combination and would have a reasonable expectation of success. The combination discloses or suggests *Melanson*'s personal safety mode would be triggered by detecting a sudden noise or change in volume followed by detection of a particular frequency pattern (e.g., loud voices or vehicle horn). *Samadani* suggests that a two-step approach helps to avoid false detections of warning conditions. A POSITA would have understood that it was important for a device like *Melanson* to implement safeguards to false detections because a user would get annoyed by frequent undesired interruptions to the program material.

352. Moreover, it was known that it would be advantageous to give users the flexibility to define multiple conditions for when to trigger interruptions to

playback as with *Melanson*'s personal safety mode. This is corroborated, for example, by *Alley* and *Bostick*.[14] *Alley* "relates generally to a safety apparatus for use with audio devices (including cellular telephones) and, more specifically, controlling the audio output of the audio device in a pre-determined manner (including muting, if necessary), in response to a predetermined noise level (sound spectrum) in the ambient space." EX1047 ¶[0003]. *Alley* emphasizes the importance of giving the user the ability to "predetermine and set various parameters, including the level of muting, ***the type and/or loudness of the sound detected that will activate a warning***, and even the type of warning (visual, tactile)." *Id.* ¶[0014]; *see also id.* ¶¶[0023] ("Also, many prior art apparatus only perform one or a very limited function. The most common function is that when a triggering event (a siren that exceeds 115 dBs) is received by the prior apparatus, it automatically mutes the sound coming from the speakers. In the present invention, the user may choose to mute the sound upon the detection of an emergency siren, may choose to only lower the sound volume upon the detection of a back-up warning from a construction truck, may provide a simple notice (three beeps) if it detects the sound of an automobile horn, etc."), [0033] ("It cannot be overemphasized how prior art devices, upon sensing a triggering

[14] To be clear, I am relying on *Alley* and *Bostick* as evidence of the knowledge of a person having ordinary skill in the art and they are not part of the unpatentability ground 2B.

event (e.g. a siren) are designed to respond in one specific fashion, which is, usually, to automatically mute the output of the audio device. The present invention, allows the user to control the sounds upon which the safety device 10 will respond, and it allows multiple responses or outputs depending on the triggering event.").

353. *Bostick* similarly "relates to a method and system for automatically muting a headphone." EX1046 ¶[0002]. *Bostick* discloses a method that is based on ***both*** a "volume threshold" and a "frequency threshold." *Id.* ¶¶[0005], [0019] ("The automatic muting feature activates when headphone 110 detects a sound that is above the volume threshold and within the frequency threshold established by the user in configuration device 120."), [0020]("In the illustrative embodiment, the muting feature is active with the volume threshold set at 20 dB and the frequency threshold range set between 4,000 and 20,000 Hz."), [0030] ("Automuting device 208 also compares the volume of the captured sound against the volume threshold. If the frequency falls within the frequency threshold range and the volume is above the volume threshold, then automuting device 208 concludes that the captured sound is a sound of interest."), [0038] ("Automuting device 238 also receives a communication from configuration device 260 indicating the frequency threshold and the volume threshold either established by a user or established through the default settings in configuration

device 260. Automuting device 238 utilizes the received signal and communication to determine whether the sound captured by microphone 232 is a sound of interest.").

354. Thus, a POSITA also would have considered the combination to amount to applying a known technique (*Samadani*'s two-step approach to detecting a warning condition) to a known device (*Melanson*'s personal safety mode) ready for improvement (*Melanson* already discloses detecting sudden noises and particular frequency patterns) to yield predictable results (*Melanson*'s device with a personal safety mode triggered by sudden noises that are determined to have particular frequency patterns).

355. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani* combination to disclose or suggest detecting sudden noise ("acute sound") and performing a separate/additional step for determining the source or type of acute sound by, for example, analyzing the frequency pattern of the sudden noise, before triggering personal safety mode.

b. *"determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics" of one of the modified microphone signals*

356. The second limitation introduced by claim 9 is specifically determining that the acute sound is the user's voice by analyzing the temporal and spectral characteristics of one of the modified microphone signals.

357. *Melanson* discloses detecting the user's voice or the voice of another person. In particular, *Melanson* explains that that with multiple microphones, the system may determine whether a sound came from the user or elsewhere and activate its playback modes accordingly. *Melanson*, 8:10-22 ("In each of the above modes where the character of external environmental sounds is being determined, the use of multiple microphones, such as microphone elements 17 and 17A, provides for the possibility of determining the direction of the environmental sound source. In such an implementation, it is possible to determine whether a person speaking or other sound is coming from the space directly in front of the user/wearer and selection of activation of such modes as social mode or linked chat mode made in conformity with the determination. Also, the mode can be initiated in this manner, and then subsequently maintained, so that two persons can walk astride while the mode is continued.").

358. By the earliest alleged priority date, there were various known techniques for detecting whether the user of a headset was speaking. For example, *Rasmussen* discloses a technique for using cross correlation to determine if the user is speaking. EX1048 ¶[0022], FIG. 4.[15]

[15] To be clear, I am relying on *Rasmussen* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2B.

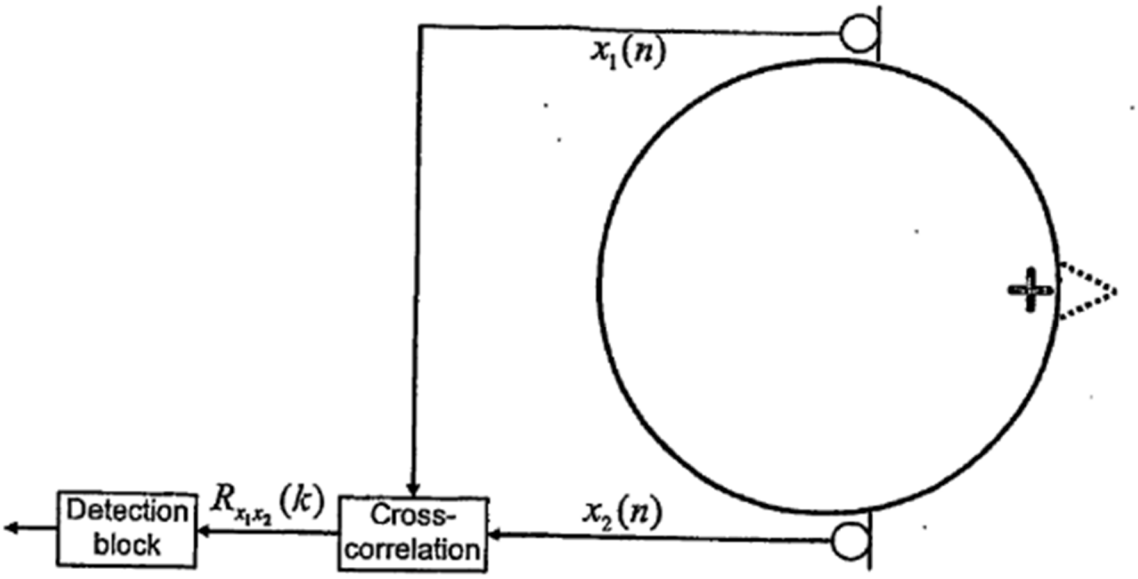


Figure 4.

EX1048 (*Rasmussen*), FIG. 4

359. As another example, *Kvaløy* discloses an earpiece that compares the level of an in-ear microphone and the level of an ambient microphone and generates "a difference signal representing the difference in signal strength from the two electroacoustic transducer elements." EX1049 ¶[0065].[16] "When the difference signal is less negative than a certain limit a decision is made that the user of the equipment is speaking. When the difference signal is more negative than this limit the user is considered not to be speaking." *Id*. ¶[0066].

360. *Dijkstra* discloses another device similar to *Melanson* in which a microphone signal is analyzed and, depending on the analysis, a mix of audio

[16] To be clear, I am relying on *Kvaløy* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2B.

content and ambient sounds is changed. More particularly, *Dijkstra* discloses a "hearing instrument" with a "classification unit" that analyzes the "frequency and/or temporal characteristics" of an earpiece microphone signal and classifies the current auditory situation into various pre-defined index values, one of which is "own voice," i.e., the user him/herself is speaking. *Dijkstra* ¶[0037]. Based on the classification index, the ratio of mix of audio content and ambient sound is changed. *Id*. ("The hearing instrument can contain various modules and units dedicated to signal processing and in particular a processing module 33 that processes signals received from the receiving unit 24, a classification unit 34 for determining the classification index based on the amplitude and/or frequency and/or temporal characteristics of the audio signals, and a central unit 35 for adaptation of the predefined amplitude and/or frequency ratio of the first audio signals relative to the second audio signals, based on the classification index.")

361. A POSITA would have understood that the "frequency characteristics" of the signal are spectral characteristics because spectral characteristics refers to the frequency content of the signal. This understanding is corroborated, for example, by *Mertens*. EX1050 ¶[0060] (equating signal analysis of "dominant frequencies" of an audio signal to "spectral analysis").[17]

[17] To be clear, I am relying on *Mertens* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2B.

Thus, a POSITA would have understood *Dijkstra* to disclose analyzing the spectral and temporal characteristics of an earpiece microphone signal and adjusting a mix of audio for playback.

362. A POSITA would have been motivated to implement *Dijkstra*'s spectral and temporal characteristics analysis technique for detecting the user's own voice in *Melanson* and would have had a reasonable expectation of success. As noted above, *Melanson* discloses that determining whether a sound originates from the user or elsewhere is useful to help determine which of its various modes to trigger, and a POSITA would have understood that *Dijkstra*'s frequency and temporal characteristics analysis was a suitable solution for that. Moreover, a POSITA would have understood that analyzing the temporal and spectral characteristics of the microphone signal as taught by *Dijkstra* would provide a robust method of detecting the user's voice. The combination would have been predictable because *Melanson*'s device already has the signal processing capability for analyzing the spectral and temporal characteristics of an earpiece microphone signal, as evidenced by *Melanson*'s ability to detect sudden noises and particular frequency patterns.

363. Thus, a POSITA would have considered the combination to amount to applying a known technique (detecting the user's own voice by analyzing spectral and temporal characteristics of the signal) to a known device

(*Melanson*) ready for improvement (*Melanson* already discloses the ability to analyze the frequency and temporal characteristics of signals) to yield predictable results (a device capable of analyzing the frequency and temporal characteristics of a sudden noise to determine if it is the user's voice).

364. A POSITA also would have considered the combination to amount to the simple substitution of one known element (*Melanson*'s directional technique for detecting whether the sound came from the user) for another (*Dijkstra*'s frequency and temporal characteristics analysis) to obtain predictable results (*Melanson*'s device that distinguishes between when the user is speaking or someone else by analyzing the frequency and temporal characteristics of the signal).

365. As discussed in §X.C.1.i, the *Melanson-Rosenberg* combination discloses or suggests detecting the acute sound by analyzing one of the first or second modified microphone signals. Thus, in the *Melanson-Rosenberg-Samadani-Dijkstra* combination, those are the signals that would be analyzed to detect the user's voice.

366. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest analyzing the sudden noise or change in volume to determine if it is the user's voice ("determining if the acute sound is a user's voice") by analyzing the frequency

and temporal characteristics of the processed microphone signal ("by analyzing the temporal and spectral characteristics of at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof").

2. Claim 13: The device according to claim 9, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the mixed signal if the user's voice is detected.

367. *Melanson* in combination with *Rosenberg, Samadani*, and *Dijkstra* discloses or suggests this claim.

368. As discussed in §X.C.2, the *Melanson-Rosenberg* combination discloses or suggests implementing *Melanson*'s social mode so that it is activated upon detection of the user's voice.

369. *Melanson* discloses that when social mode is activated, "[t]he microphone input can be provided at the headphone output ***and/or*** mixed with the program audio, so that the user is able to hear the other person quite well without requiring removal of headphone elements." *Melanson*, 7:31-34 (emphasis added). A POSITA would have understood that because *Melanson* discloses providing the microphone input at the headphone output ***or*** mixed with the program audio, that *Melanson* discloses that the microphone input can be exclusively output (i.e., without mixing). Accordingly, a POSITA would have

understood *Melanson* to disclose or suggest that, in social mode, the microphone input can be provided ***instead of*** the program content.

370. As explained in §X.C.1.k, the *Melanson-Rosenberg* combination discloses or suggests generating and outputting an "ambient sound signal." *Rosenber* also discloses an embodiment in which the ambient sound signal is provided to the user instead of the audio content. In particular, *Rosenberg* discloses an embodiment in which "the musical audio content is gradually decreased down to substantially zero while the ambient audio musical [*sic*] content is gradually increased up to the prior music volume level." *Rosenberg* ¶[0058]. "Such a cross-fade enables the music to fade out while the ambient audio content fades in." *Id.* Then, after a period of time, "the ambient audio content fad[es] out to zero volume and the musical content fading back to its pre-event nominal volume." *Id*.

371. A POSITA would have understood that adjusting the volume of the ambient sound signal "modifies" the ambient signal. A POSITA would have further understood that, after the cross-fade, the ambient sound signal is output to the speaker instead of the audio content. Thus, a POSITA would have understood *Rosenberg* to disclose outputting a volume-adjusted ambient sound signal to the speaker instead of the audio content signal.

372. A POSITA would have been motivated to implement *Rosenberg*'s teaching related to cross-fading in *Melanson* and would have had a reasonable expectation of success in doing so. For example, a POSITA would have recognized that the cross-fade would provide a more pleasant user experience than, e.g., abruptly cutting off the program material. Moreover, implementing such a cross-fade feature in *Melanson* would have been well within the skill of an ordinary artisan, as demonstrated by *Rosenberg*.

373. Thus, a POSITA would have considered the combination to amount to the use of a known technique (*Rosenberg*'s cross fade to output the ambient sound instead of the audio content) to improve similar devices (*Melanson* and *Rosenberg* both disclose outputting ambient sound instead of audio content upon detection of an event) in the same way (to provide the user with a more pleasant listening experience).

374. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest cross-fading the ambient sound signal in and the program material out ("sending a modified ambient signal to the speaker instead of the mixed signal") if the user's voice is detected ("if the user's voice is detected").

3. Claim 14: The device according to claim 9, wherein the operations further comprise: detecting whether the acute sound is at least one of a warning sound, siren, voice, specific noise type, or a combination, by analyzing the spectrum of at least one of the modified first microphone signal, the modified second microphone signal, the modified ambient sound signal, or a combination.

375. *Melanson* in combination with *Rosenberg, Samadani*, and *Dijkstra* discloses or suggests this claim. Claim 14 is similar claim 9 with respect to requiring a separate step for detecting the source or type of acute sound. Instead of detecting whether the acute sound is the user's voice, claim 14 requires detecting whether the acute sound is one of various other sound types "by analyzing the spectrum" of one of the modified microphone signals.

376. As discussed in §X.D.1.a, the *Melanson-Rosenberg-Samadani* combination discloses or suggests *Melanson*'s personal safety mode modified to be triggered by detecting a sudden noise or change in volume followed by detection of a particular frequency pattern (e.g., loud voices or vehicle horn). A POSITA would have understood that detecting a particular frequency pattern involves analyzing the "spectrum" of the signal because the "spectrum" of the signal refers to the frequency content of the signal. This is corroborated, for example, by *Mertens*. EX1050 ¶[0060] (equating signal analysis of "dominant

frequencies" of an audio signal to "spectral analysis").[18] Thus, detecting frequency patterns requires analyzing the spectrum, i.e., frequency content, of the signal for patterns.

377. As discussed in §X.C.1.i, the *Melanson-Rosenberg* combination discloses or suggests detecting the acute sound by analyzing one of the first or second modified microphone signals. Thus, in the *Melanson-Rosenberg-Samadani-Dijkstra* combination, those are the signals that would be analyzed to detect the frequency patterns for, e.g., loud voices or a vehicle horn.

378. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest analyzing the sudden noise or change in volume to determine if it is loud voices or a vehicle horn ("detecting whether the acute sound is at least one of a warning sound, siren, voice, specific noise type, or a combination") by analyzing the frequency patterns of the processed microphone signal ("by analyzing the spectrum of at least one of the modified first microphone signal, the modified second microphone signal, the modified ambient sound signal, or a combination").

[18] To be clear, I am relying on *Mertens* as evidence of the knowledge of a person having ordinary skill in the art and it is not part of the unpatentability ground 2B.

4. Claim 15: The device according to claim 14, wherein the operations further comprise: decreasing a volume of the modified audio content signal when a warning or siren is detected.

379. *Melanson* in combination with *Rosenberg*, *Samadani*, and *Dijkstra* discloses or suggests this claim.

380. *Melanson* discloses that, in personal safety mode, when "sudden changes in volume, or particular frequency patterns such as vehicle horns or loud voices, are detected," "processing circuit 34 either ***partially or completely attenuates the program material gain***, so that a user of the personal audio playback device will hear the outside sounds." *Melanson*, 7:10-24; *see also id.*, 1:38-41 ("As another example, a pedestrian listening to audio via such a device may not notice a vehicle horn or siren that is alerting them to a hazard."). This is consistent with how the '473 patent describes detecting a warning sound. *See '473pat*., 9:15-16 ("Block 412 presents a method for warning signal detection (e.g. car horns, klaxons)."). Thus, a POSITA would have understood that a vehicle horn is a type of warning sound in the context of the '473 patent.

381. As detailed in §X.C.1.j, a POSITA would have understood that attenuating the gain of the program material would decrease the volume of the program material.

382. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest attenuating

the gain of the program material ("decreasing a volume of the modified audio content signal") when a vehicle horn is detected ("when a warning … is detected").

5. Claim 16: The device according to claim 14, where the operations further comprise: sending a notification signal to the speaker if the acute sound is a warning sound or siren.

383. *Melanson* in combination with *Rosenberg*, *Samadani*, and *Dijkstra* discloses or suggests this claim.

384. As explained in §X.D.3, *Melanson* discloses that, in personal safety mode, if a vehicle horn is detected ("if the acute sound is a warning sound"), the output of the playback device will be adjusted. As explained in §X.D.1.a, *Samadani* discloses a system similar to *Melanson*. *Samadani* discloses outputting an alarm to the user's headphones when a warning condition is detected, including by mixing the alert signal into the audio output. *Samadani* ¶¶[0016] (" Once the processor has determined that a warning condition exists, processor 14 is configured to provide an alert to the user. The form of the alert may vary, as desired. For example, the alert may include an audible alarm, a visual alarm, a tangible alarm, or a combination of any of these."), [0022] ("For example, the present disclosure also contemplates a device, which may or may not be worn on the user's body, which detects a given signal and alerts the user by vibrating or flashing lights. ***Of course such a device could also incorporate***

headphones, in which case an alert could alternatively or additionally be delivered to the user via the headphones.”); *see also id*. ¶¶[0017]-[0018] (“Accordingly, alarm source 16 may be configured to produce or direct the production of the audible, visual, or tangible alarm. … Processor 14 may further include or be in electronic communication with a mixer (not shown). The mixer may be any device configured to receive multiple audio signals, such as those generated by alarm source 16 and audio source 20 and mix and/or alter the audio signals when a warning condition is detected.”).

385. A POSITA would have been motivated to implement *Samadani*’s teachings related to sending an alert signal in *Melanson*’s personal audio playback device and would have had a reasonable expectation of success in doing so. Both *Melanson* and *Samadani* are concerned with ensuring that users are aware of hazards in the environment when using headphones, and a POSITA would have recognized that implementing *Samadani*’s alert signal technique would have advanced that objective. Thus, a POSITA would have considered the combination to amount to the use of a known technique (*Samadani*’s alarm signal) to improve similar devices (*Melanson* and *Samadani* both disclose adjusting the output of a headset to make users aware of hazards in the environment) in the same way (to provide an additional signal to the user to make them aware of the hazards in the environment).

386. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest sending an alert signal to the speaker ("sending a notification signal to the speaker") if a vehicle horn is detected ("if the acute sound is a warning sound or siren"). `

6. Claim 17: The device according to claim 16, wherein the warning sound is at least one of a bell, an emergency vehicle, the sound of a security system or a combination thereof.

387. As detailed in §X.C.1.i, *Melanson* discloses or suggests detecting particular frequency patterns, including vehicle horns and sirens. *Samadani* expressly discloses "identify[ing] the characteristics of commonly used emergency vehicle sirens and determin[ing] when a signal matches an emergency vehicle siren that is approaching the car from behind." *Samadani* ¶[0035].

388. A POSITA would have been motivated to combine *Samadani*'s teachings related to detecting an emergency vehicle siren into the *Melanson-Rosenberg-Samadani-Dijkstra* combination and would have had a reasonable expectation of success in doing so. A POSITA would have understood that it would have advanced *Melanson*'s objectives for notifying the user of important events in the environment. Therefore, a POSITA would have considered the combination to amount to the use of a known technique (detecting emergency vehicle sirens) to improve similar devices (*Melanson* and *Samadani* are both

directed to adjusting the output of a media device in response to detecting particular sounds) in the same way (to make users aware of hazards in the environment).

389. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest detecting an acute sound that is an emergency vehicle siren ("wherein the warning sound is at least one of a bell, an emergency vehicle, the sound of a security system or a combination thereof").

7. Claim 18: The device according to claim 14, wherein if an alarm is detected it is at least one of a police siren, an ambulance siren, a car honking or a combination thereof.

390. *Melanson* in combination with *Rosenberg*, *Samadani*, and *Dijkstra* discloses or suggests this claim.

391. As discussed in §X.C.1.i, *Melanson* discloses detecting particular frequency patterns, including vehicle horns. A POSITA would have understood that detecting a vehicle horn is the same thing as detecting a car honking. This is consistent with the '473 patent insofar as the '473 patent discloses detecting "car horns" (*473pat*., 9:15-16), and does not separately disclose or otherwise distinguish that from detecting "car honking." Thus, a POSITA would have understood *Melanson* to disclose detecting a car honking, and thus also detecting an "alarm" in the context of claim 18.

392. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra* combination to disclose or suggest detecting a vehicle horn ("if an alarm is detected, it is at least one of … a car honking ….").

E. Ground 2C: *Melanson*, *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* Render Obvious Claims 10-12

1. Claim 10: The device according to claim 9, wherein the operations further comprise: generating a noise signal from at least one of the modified first microphone signal, the modified second microphone signal, or both; generating a noise reduction signal.

393. *Melanson* in combination with *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* discloses or suggests this claim.

394. *Melanson* discloses that the personal audio playback device may "include subtractive noise-cancellation as a selectable operating mode and, if so, the microphones used to provide input to the noise-cancellation process may be used to provide input to the processing circuit of the present invention." *Melanson*, 3:24-27. *Melanson* leaves the details for implementing noise-cancellation to the knowledge of a POSITA, and a POSITA looking for additional details for how to implement noise cancellation would have readily identified *Engle*.

395. *Engle* discloses headphones with active noise cancellation. *Engle*, 1:4-5 ("The present invention relates to headphones, and particularly to active noise cancellation in headphones."). With reference to Figures 4 (analog) and 5

(DSP), *Engle* discloses using a set of custom noise reduction filters that vary based on the type of noise that should be canceled (or not canceled) and the environment in which the headphones are worn:

> Another problem of a conventional noise reduction headphone is that fixed noise reduction filters are used, which limits the amount of noise reduction available, and does not allow a user to vary the noise canceling characteristics based on the external noise. For example, a user may wish that certain types of noise, such as voices or emergency sirens, not be canceled. The types of noise that should be canceled, and those that should not be canceled, may vary depending on the environment where the headphones are worn. Accordingly, filters that meet these needs would be desirable.

Engle, 2:11-19

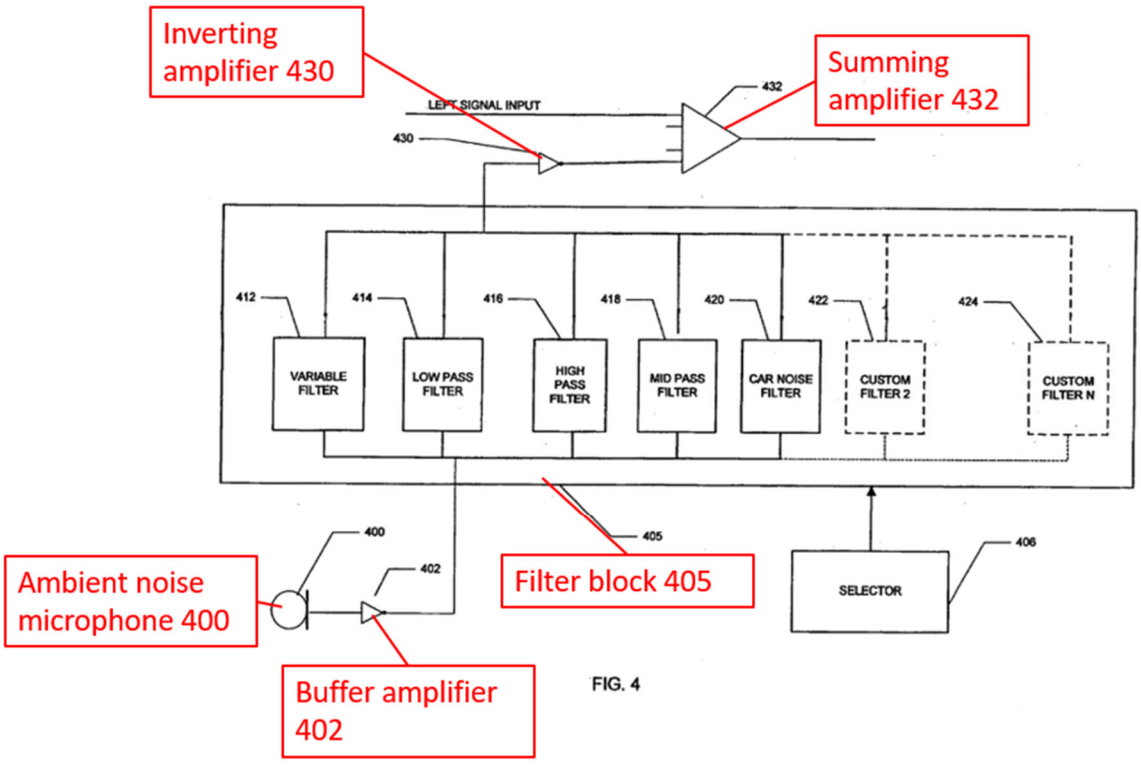
> In another embodiment of the invention, a headphone is provided with active noise reduction including active noise reduction filters applicable to one or more predefined frequency bands selectable by a user, and filters with frequency bands and/or gains selectable by a user, as illustrated in Figure 4.
>
> Referring to Figure 4, ambient noise from a microphone 400 is buffered and amplified by a buffer amplifier 402. The output from the buffer amplifier is fed to one of the filters in filter block 405, as selected by the user employing selector 406. Selector 406 may operate as set forth above in connection with Figs. 1 and 2. The particular filter may be selected by commands provided to the

microphones, in one of the manners described above, or by a switch of a different type. The output signal from a selected filter is inverted and amplified by an inverting amplifier 430. The output of the inverted noise signal is added to an input audio signal by a summing amplifier 432 and the sum signal is then fed to a speaker (not shown) in the headphone, so that the output signal from the speaker can cancel or reduce the ambient noise present at an output of the headphone.

Switchable frequency bands, and high pass, low pass, and mid pass filters, as shown in Figure 4, could be independently switched in. Two or more such bands may be activated at the same time. Variable filter 412 may include variable frequency, variable gain, and variable bandwidth, each of which parameters may be user-defined. These features permit a user great flexibility to cancel out a specific band of noise that is a problem. Low pass filter 414, high pass filter 416, and mid pass filter 418 may be individually tunable through a suitable user interface to permit users to select ranges of frequencies for active noise cancellation. Car filter 420, and custom filter 2 through custom filter N, . including as many custom filters as may be desired, may be provided. These custom filters may be specially tuned filters for an environment in general terms, such as the interior of a passenger car. Custom filters may be provided for specific variations on such an environment, such as specific makes and models of automobile, noise generated by various road surfaces, and such variables as open and closed windows, air conditioner noise, and/or fan noise. The custom

filters can be specially tuned filters for other environments, such as the cabin of a jet airplane, similarly configured passenger cabins of propeller planes, helicopter environment and associated noise, noises of motors found in a home, office or other building, such as air conditioner noise (e.g., window unit), vacuum cleaner, and industrial environments, such as warehouses and factory environments.

Id., 16:5-18:2



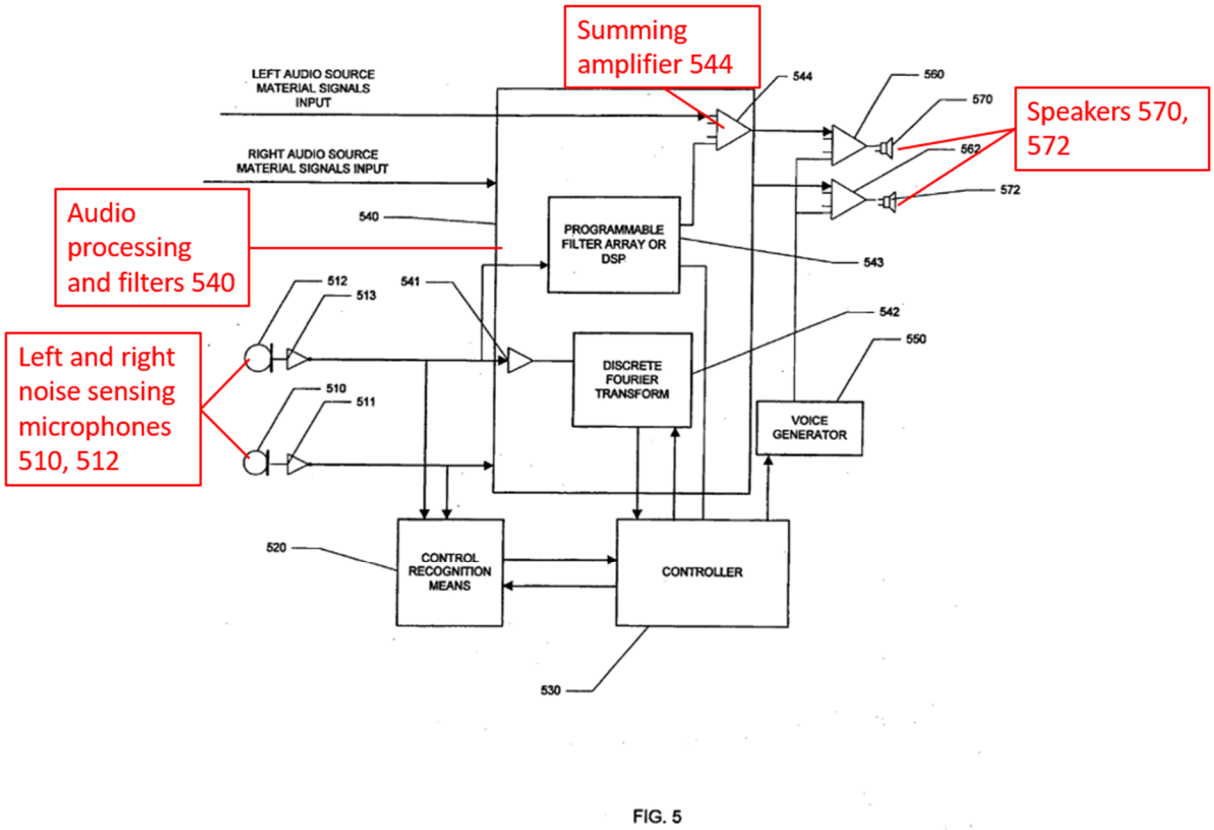
Engle, FIG. 4 (annotated)

Figure 5 illustrates another embodiment of the invention, where a digital signal processing (DSP) system is added to characterize the received noise, for example using spectral and amplitude information obtained through a spectral analysis, and program a

programmable filter array or a digital signal processor according to the characterization. In this way, noise cancellation or reduction can be dynamically optimized during transient conditions, such as passing vehicles, trees, trains, or aircraft. The noise processing of the left and right headphone is similar. For simplicity, Figure 5 shows only the processing of the left headphone. Ambient noise is received at left and right noise sensing microphones 510, 512, which are amplified by amplifiers 511 , 512. The received ambient noise signals are provided to control recognition means 520, similar to control recognition means 120 discussed above. The received ambient noise signals are provided to audio processing and filters 540, and are converted into digital signal by an analog-to-digital converter 541. The digitized noise signal is discrete Fourier transformed at block 542 to obtain the spectra and amplitudes of the ambient noise. The spectrum and amplitude information is sent to the controller 530, which stores information in a memory on those bands in the ambient noise that should not be cancelled. For example, bands corresponding to human speech and emergency sirens should not be cancelled. Based on the received spectrum and amplitude information and the information in the memory, the controller 530 determines one or more bands of ambient noise to be cancelled or reduced, and instructs the programmable filter array or DSP system 543 to construct filters to provide active noise cancellation for those bands that should be cancelled or reduced. The DSP system 543 performing the construction of the filter may be a programmable filter array or a digital signal processor. DSP system 543 may also invert the

filtered passed signal and output the inverted signal to a summing amplifier 544. Another input of summing amplifier 544 is coupled to left audio source materials input. Summing amplifier 544 sums the two received signals and the sum signal is digital to analog converted and input to one of buffer amplifiers 560, 562. Buffer amplifiers 560, 562, also receive signals from voice generator 550, which is similar to voice generator 150 of Fig. 1. Buffer amplifiers 560, 562 sum the received signals and provide an amplified output signal used to drive speakers 570, 572.

Engle, 18:3-19:22



Engle, FIG. 5 (annotated)

As an example of constructing the filters, assume that the controller decides that first and second frequency bands of the

ambient noise should be cancelled or reduced, the controller instructs a digital signal processor included in the DSP system to construct two band pass filters for passing the respective first and second bands. These two band pass filters should be arranged in parallel, so that both bands are present at the output. If a programmable filter array is used, the programmable array should includes many filters, each having a different passing band, and capable of being connected in parallel or in series in any combination.

If a digital signal processor is used to construct the filters, the digital signal processor should also perform the DFT as discussed above. In this embodiment, the ambient noise received by the microphone is converted to a digital signal by a second analog to digital converter, the digital signal is transformed into frequency domain, the digital signal in frequency domain is passed through the constructed filters, the filtered signal is inversely transformed into time domain, the filtered signal in time domain is inverted by an inverter, the inverted signal is converted to analog signal by a digital to analog converter, and the converted analog signal is then sent to the summing amplifier. The second analog to digital converter and the digital to analog converter are not shown for simplicity. The filters constructed by the digital signal processor can be MR or FIR filters as known in the art.

If a digital programmable filter array is used, the received ambient noise is converted into digital, transformed into time domain, and the filtered signal is inversely transformed to time domain,

inverted, and converted in a analog signal, as described above for using a digital signal processor.

Id., 19:17-20:21

396. *Engle* discloses that a microphone signal is filtered for particular environmental noise and then inverted before being mixed with audio source materials. *Id*., 19:6-11. In particular, *Engle* discloses "the ambient noise received by the microphone" is digitized and passed through a set of filters, and then "the filtered signal in time domain is inverted by an inverter," and then the inverted signal is converted to analog and summed with the audio program materials. *Id*., 20:4-13. *Engle* explains that the filters are constructed "to provide active noise cancellation for those bands that should be cancelled or reduced" so that bands corresponding to, for example, "human speech and emergency sirens" are not cancelled. *Id*., 18:10-19:6.

397. A POSITA would have understood that the filtered signal is a "noise signal" because it represents the noise to be cancelled. A POSITA would have understood that the inverted, filtered signal is a "noise-reduction signal" because when it is mixed with another signal, it will have the effect of reducing certain noise in that signal.

398. A POSITA would have been motivated to modify the *Melanson-Rosenberg-Samadani-Dijkstra* combination to include *Engle*'s teachings related

to generating a noise reduction signal and would have had a reasonable expectation of success in doing so.

399. A POSITA would have appreciated that *Melanson* and *Engle* disclose devices with very similar functionality. Similar to *Melanson*, *Engle* describes headphones that can detect an ambient sound, adjust the output of the media player based on the ambient audio signal, and provides for active noise cancellation. *Engle*, 1:4-5 ("The present invention relates to headphones, and particularly to active noise cancellation in headphones"), 4:1-5 ("According to another aspect, the invention provides a headphone, comprising: a microphone for receiving noise; a controller coupled to the microphone, wherein, in response to detection of a signal having a specified characteristic, the controller controls operation of the headphone in response to detection of the specified characteristic."). *Engle* further discloses embodiments that include "left" and "right" microphones and the ambient sound is detected by those microphones. *Id.*, 5:8-15 ("Figure 1 is a block diagram illustrating a headphone system according to an embodiment of the invention. Left noise sensing microphone 110 and right noise sensing microphone 112 detect signals, and in particular provide analog ambient noise signals to amplifiers 111 , 113, respectively. The amplified ambient noise signals are provided to control recognition means

(CRM) 120. CRM 120 determines if the noise signals have a specific, or particular, characteristic to determine if a user input signal is received.").

400. To the extent that *Melanson* does not expressly disclose the claimed generating a noise signal and generating a noise reduction signal, a POSITA looking for additional implementation details for *Melanson*'s subtractive noise-cancellation mode would have readily identified *Engle* and would have understood the applicability of *Engle*'s teachings to *Melanson*'s personal audio playback device. A POSITA would have recognized that *Engle*'s noise reduction technique based on programmable filters would improve *Melanson*'s personal audio playback device by making *Melanson*'s device more flexible and usable in various environments. That *Melanson* and *Engle* disclose similar devices with similar functions confirms that a POSITA would have understood that *Engle*'s teaching of a headphone device that generates a noise signal and a noise reduction signal could predictably be applied to *Melanson*'s disclosed personal audio playback device.

401. As explained in §X.C.1.h, the *Melanson-Rosenberg* combination discloses or suggests that the microphone signals are processed upon capture prior to additional processing.

402. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra-Engle* combination to disclose or suggest that the

captured and processed microphone signal is filtered ("generating a noise signal from at least one of the modified first microphone signal, the modified second microphone signal, or both") and that the filtered microphone signal is then inverted ("generating a noise reduction signal").

2. Claim 11: The device according to claim 10, wherein the operations further comprise: mixing the noise reduction signal with the mixed signal to generate a new mixed signal, and sending the new mixed signal to the speaker instead of the mixed signal.

403. *Melanson* in combination with *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* discloses or suggests this claim.

404. As noted above in §X.E.1, *Melanson* discloses that the personal audio playback device may include an additional selectable operating mode that "include[s] subtractive noise-cancellation." *Melanson*, 3:24-27. *Melanson* also discloses that each of the modes may be "selectively enabled." *Id.*, 8:33-39. Thus, a POSITA would have understood *Melanson* to disclose that the disclosed modes may be enabled and triggered in any order. Therefore, a POSITA would have understood *Melanson* to suggest a scenario in which personal safety mode is triggered first and noise-cancellation mode is triggered second. In that scenario, a POSITA would have understood that triggering the noise-cancellation mode would cause a noise-reduction signal to be mixed with the

mixed signal comprising the ambient and program audio created by personal safety mode.

405. Accordingly, a POSITA would have understood the *Melanson-Rosenberg-Samadani-Dijkstra-Engle* combination to disclose or suggest triggering the subtractive noise-cancellation mode after the personal safety mode is triggered ("mixing the noise reduction signal with the mixed signal to generate a new mixed signal, and sending the new mixed signal to the speaker instead of the mixed signal")

3. Claim 12: The device according to claim 11, wherein the operations further comprise: sending a modified ambient signal to the speaker instead of the new mixed signal if the user's voice is detected.

406. *Melanson* in combination with *Rosenberg*, *Samadani*, *Dijkstra*, and *Engle* discloses or suggests this claim for the reasons discussed in §X.D.2.

XI. SECONDARY CONSIDERATIONS

407. I am not aware of any evidence in the '473 patent's prosecution history or elsewhere supporting any secondary considerations arguments, or evidence of nexus of such alleged evidence to the challenged claims. To the extent Patent Owner asserts the existence of any secondary considerations in its responses, I reserve the right to address any such evidence.

XII. CONCLUSION

408. The findings and opinions set forth in this declaration are based on my work and examinations to date.

409. I may continue my examinations. I may also receive additional documentation and other factual evidence over the course of this IPR that will allow me to supplement and/or refine my opinions. I reserve the right to add to, alter, or delete my opinions and my declaration upon discovery of any additional information. I reserve the right to make such changes as may be deemed necessary.

410. In signing this declaration, I recognize that the declaration will be filed as evidence in an IPR before the PTAB. I also recognize that I may be subject to cross-examination in the case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

411. I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Date: ___2024/06/12_____



Chris Kyriakakis