

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

HARMAN INTERNATIONAL INDUSTRIES, INC.,

Petitioner,

v.

ST CASESTECH LLC,

Patent Owner.

Case IPR2024-01302

U.S. Patent No. 8,150,044

**DECLARATION OF DR. MICHAEL T. JOHNSON IN SUPPORT
OF PETITION FOR *INTER PARTES* REVIEW
OF U.S. PATENT NO. 8,150,044**

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I. INTRODUCTION

1. My name is Dr. Michael T. Johnson and I am a Full Professor in the Department of Electrical and Computer Engineering at the University of Kentucky in Lexington, Kentucky. I have been retained on behalf of Harman International Industries, Inc. (“Harman”), to provide my opinions on certain issues related to U.S. Patent No. 8,150,044 (“the ’044 patent”) in connection with the above-captioned inter partes review (IPR) proceeding. In particular, I have been asked to provide my insights, analysis, and opinions regarding whether claims 1-21 of the ’044 patent (“Challenged Claims”) are obvious over the prior art references identified below.

2. I understand that the ’044 patent is titled “Method and device configured for sound signature detection” names as inventor Steven Wayne Goldstein, Mark A. Clements, and Marc A. Boillot and is currently owned by ST CasesTech LLC. I have considered the ’044 patent. I understand that the ’044 patent has been provided as Ex. 1001.

3. I understand that the file history of the ’044 patent has been provided as Ex. 1002. I have considered this file history, and I will refer to it as “the ’044 File History.”

4. I understand that the ’044 patent claims priority to provisional application No. 60/883,013, filed on Dec. 31, 2006. I understand that the file history

for the provisional applications No. 60/883,013 has been provided as Exhibit (1014). I have considered this file history.

5. I understand that the '044 patent claims a priority date of December 31, 2006, the filing date of provisional application No. 60/883,013. I have been asked to assume that priority date for purposes of rendering my opinions in this declaration. I have not been asked to opine on whether the claims of the '044 are entitled to this priority date, and do not provide any opinion on that subject.

6. My analysis and conclusions set forth in this declaration are based on my educational background and experiences in the field (see Section II). I am qualified to opine concerning what a person of ordinary skill in the art would have known and understood during that time. Before December 31, 2006, my level of skill in the art was at least that of a person of ordinary skill. I am qualified to opine concerning what a person of ordinary skill in the art would have known and understood at the time of the '044 patent's purported invention. My analysis and conclusions herein are from the perspective of a person of ordinary skill in the art as of December 31, 2006.

7. 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 '044 patent; my own knowledge and

experiences gained from my work experience in the field of the '044 patent and related disciplines; and my experience in working with others involved in this field and related disciplines.

8. I have considered the following non-exhaustive list of references in preparing my declaration. I have been instructed to assume that each of these references qualifies as prior art.

- **U.S. Patent Application Publication No. 2001/0046304** to Rodger H. Rast (“Rast”), titled “System and method for selective control of acoustic isolation in headsets” filed on April 24, 2001, and published November 29, 2001. I understand that a copy of this reference has been designated as Exhibit 1004.
- **European Patent Application No. 1,519,625A2** to Thomas A. Victorian, David A. Preves, and J. Virgil Bradley (“Victorian”), titled “External ear canal voice detection” filed on September 13, 2004, and published March 30, 2005. I understand that a copy of this reference has been designated as Exhibit 1005.
- **U.S. Patent Application Publication No. 2007/0038395** to David Green, Dean Beale, Andrew Hume, Jason Senoir (“Green”), titled “Sensor apparatus and system” filed on Sep. 2, 2004, and published February 15, 2007. I understand that a copy of this reference has been designated as Exhibit 1006.
- **U.S. Patent Application Publication No. 2009/0085873** to William L. Betts, and Carol Betts (“Betts”), titled “Sensory enhancement systems and methods in personal electronic devices” filed on February 1, 2006, and published April 2, 2009. I understand that a copy of this reference has been designated as Exhibit 1007.
- **U.S. Patent Application Publication No. 2006/0293882** to David Giesbrecht, and Phillip Hetherington (“Giesbrecht”), titled “System and method for adaptive enhancement of speech signals” filed on June 28,

2005, and published December 28, 2006. I understand that a copy of this reference has been designated as Exhibit 1008.

- **U.S. Patent No. 4,325,142** to Kunihiro Nakazawa (“Nakazawa”), titled “Portable radio set with a carrying holder” filed on July 25, 21980, and published April 13, 1982. I understand that a copy of this reference has been designated as Exhibit 1009.

9. I understand that there is a pending district court litigation between Petitioner and ST CasesTech, LLC (“Harman Litigation”) involving the ’044 patent.

10. I am being compensated by Harman at my standard hourly consulting rate of \$750 for my time on this matter. My compensation is not dependent on the outcome of this proceeding or any other proceeding between the parties.

11. In forming my opinions, I relied on the documents cited in this declaration and the documents identified in the attached Appendix B. These documents comprise patents, file histories, printed publications, and other related documents. As discussed below, each document is a type that experts in my field would have reasonably relied upon when forming their opinions. Further, experts in my field would have had access to each document either through the applicable patent offices and/or well-known libraries, conferences, or publications in the field. My opinions are also based upon my personal and professional experience.

II. QUALIFICATIONS

12. I am qualified by education and experience to testify as an expert in the field of audio devices and audio signal processing. Additionally, the following

overview of my background pertains to my qualifications for providing expert testimony in this matter. I am over the age of 18 and am competent to write this declaration. I have personal knowledge, or have developed knowledge, of these technologies based upon education, training, or experience of the matters set forth herein.

13. My educational background includes undergraduate degrees both in Computer Science Engineering and in Electrical Engineering from LeTourneau University in Longview, TX, in 1989 and 1990, respectively, and a Master's degree in Electrical Engineering from the University of Texas at San Antonio in 1994. I earned a PhD from Purdue University in West Lafayette, Indiana, in 2000, with a doctoral dissertation focused on speech processing and automatic speech recognition. Prior to receiving my PhD, I worked in industry as a design engineer and engineering manager for more than 5 years, primarily in the telecommunications industry.

14. I am currently a Full Professor in the Department of Electrical and Computer Engineering at the University of Kentucky (UK) in Lexington, Kentucky. I also currently serve as the Associate Dean for Undergraduate Education and Student Success in the Stanley and Karen Pigman College of Engineering at the UK, a role I have held since July 2023.

15. I previously served as the Electrical and Computer Engineering Department Chair from 2016 to 2023.

16. Prior to my faculty role at UK, I was a faculty member at Marquette University in Milwaukee, WI from 2000 to 2016, during which time I also spent two separate sabbatical years as a Visiting Professor at Tsinghua University in Beijing, China.

17. The focus of my research work is Speech and Signal Processing, and I regularly conduct and publish original research articles within this field, with a publication record including over 50 peer-reviewed journal articles and more than 150 total peer-reviewed publications over the past three decades. My specific expertise includes automatic speech recognition, speech production and kinematics, bioacoustics, and machine learning. I have received multiple research grant awards for my work in this area, including from the National Science Foundation and the National Institutes of Health.

18. As a faculty member, I regularly teach both undergraduate and graduate courses in electrical and computer engineering, primarily in the areas of computers, signal processing, and machine learning. This includes graduate level courses in speech and signal processing, including topics directly related to acoustics and acoustic devices such as those at issue in these proceedings.

19. My professional service includes active membership in the Institute for Electrical and Electronic Engineers (IEEE), the IEEE Signal Processing Society, the Acoustical Society of America (ASA), and the Electrical and Computer Engineering Department Heads Association (ECEDHA). I served as a member of the IEEE Signal Processing Society Speech and Language Technical Committee from 2014 to 2017, and regularly serve as a reviewer for journals, conferences, and workshops in the area of speech and signal processing, including IEEE Transactions on Acoustics, Speech, and Signal Processing, IEEE Transactions on Signal Processing, the Journal of the Acoustical Society of America, Speech Communication, the International Conference on Acoustics, Speech, and Signal Processing, the IEEE Workshop on Automatic Speech Recognition, and Interspeech. I have reviewed funding proposals for a number of public and private agencies and foundations, including for multiple programs and directorates at the National Science Foundation. In addition to these manuscript and grant proposal reviewing activities, I also currently serve as Area Editor for the journal Speech Communication in the area of speech enhancement.

20. During my academic career, I have been honored to receive a number of awards for my teaching and research work. Notable awards have included the College of Engineering Researcher of the Year at Marquette University in 2007, Milwaukee Young Engineer of the Year in 2006, and Electrical and Computer Engineering Eta Kappa Nu Teacher of the Year in 2005 and 2014. In 2008, my work

applying speech recognition technology to animal vocalizations was featured in the Milwaukee Journal-Sentinel series “Groundbreaking Thinkers in Wisconsin.”

21. A detailed record of my professional qualifications, including a list of publications, awards, and professional activities, is set forth in my curriculum vitae attached to this report as Appendix A.

III. LEGAL UNDERSTANDING

A. My Understanding of Claim Construction

22. 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.

23. I have been informed that patent claims are construed from the viewpoint of a person of ordinary skill in the art of the patent at the time of the invention. I have been informed that patent claims generally should be interpreted consistent with their plain and ordinary meaning as understood by a person of ordinary skill in the art in the relevant time period (i.e., at the time of the purported invention, or the so called “effective filing date” of the patent application), after reviewing the patent claim language, the specification, and the prosecution history (i.e., the intrinsic record).

24. I have further been informed that a person of ordinary skill in the art must read the claim terms in the context of the claim itself, as well as in the context

of the entire patent specification. I understand that in the specification and prosecution history, the patentee may specifically define a claim term in a way that differs from the plain and ordinary meaning. I understand that the prosecution history of the patent is a record of the proceedings before the U.S. Patent and Trademark Office and may contain explicit representations or definitions made during prosecution that affect the scope of the patent claims. I understand that an applicant may, during the course of prosecuting the patent application, limit the scope of the claims to overcome prior art or to overcome an examiner's rejection, by clearly and unambiguously arguing to overcome or distinguish a prior art reference, or clearly and unambiguously disavowing claim coverage.

25. In interpreting the meaning of the claim language, I understand that a person of ordinary skill in the art may also consider "extrinsic" evidence, including expert testimony, inventor testimony, dictionaries, technical treatises, other patents, and scholarly publications. I understand this evidence is considered to ensure that a claim is construed in a way that is consistent with the understanding of those of ordinary skill in the art at the time of the claimed invention. For example, this can be useful for a technical term whose meaning may differ from its ordinary English meaning. I understand that extrinsic evidence may not be relied on if it contradicts or varies the meaning of claim language provided by the intrinsic evidence, particularly if the applicant has explicitly defined a term in the intrinsic record.

B. My Understanding of Anticipation

26. I understand that a patent claim is invalid as “anticipated” if all the limitations of the claim are present, either expressly or inherently, in a single prior art publication or patent.

27. I also understand that a prior art reference can disclose a claimed feature because the feature is expressly described or because the feature is inherent in the disclosure. I understand that something is inherent in a prior art reference if the missing descriptive matter must necessarily be present, not merely probably or possibly present, and it would be so recognized by a person of ordinary skill in the art as of the priority date of the patent. I understand that it is acceptable to examine evidence outside of the prior art reference (i.e., extrinsic evidence) in determining whether a feature, while not expressly discussed in the reference, is necessarily present in it.

C. My Understanding of Obviousness

28. I understand that a claim may be invalid if the subject matter described by the claim as a whole would have been obvious to a hypothetical person of ordinary skill in the art in view of a prior art reference or in view of a combination of references at the time the claimed invention was made. Therefore, I understand that obviousness is determined from the perspective of a hypothetical person of ordinary skill in the art and that the asserted claims of the patent should be read from

the point of view of such a person at the time the claimed invention was made. 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 in suit.

29. 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 have also been advised that an analysis of whether a claimed invention would have been obvious should be considered in light of the scope and content of the prior art, the differences (if any) between the prior art and the claimed invention,

and the level of ordinary skill in the pertinent art involved. I understand as well that a prior art reference should be viewed as a whole.

31. I have also been advised that in considering whether a claimed invention would have been obvious over a combination of prior art references, I may assess whether there are apparent reasons to combine known elements in the prior art in the manner claimed in view of interrelated teachings of multiple prior art references, the effects of demands known to the design community or present in the marketplace, and/or the background knowledge possessed by a person of ordinary skill in the art. I understand that other principles may be relied on in evaluating whether a claimed invention would have been obvious, and that these principles include the following:

- A combination of familiar elements according to known methods is likely to be obvious when it does no more than yield predictable results;
- When a device or technology is available in one field of endeavor, design incentives and other market forces can prompt variations of it, either in the same field or in a different one, so that if a person of ordinary skill can implement a predictable variation, the variation is likely obvious;
- If a technique has been used to improve one device, and a person of ordinary skill in the art would recognize that it would improve similar devices in the same way, using the technique is obvious unless its actual application is beyond their skill;
- An explicit or implicit teaching, suggestion, or motivation to combine two prior art references to form the claimed combination may demonstrate obviousness, but proof of obviousness does not depend on or require showing a teaching, suggestion, or motivation to combine;

- Market demand, rather than scientific literature, can drive design trends and may show obviousness;
- In determining whether the subject matter of a patent claim would have been obvious, neither the particular motivation nor the avowed purpose of the named inventor controls whether the claim is obvious;
- One of the ways in which a patent's subject can be proved obvious is by noting that there existed at the time of invention a known problem for which there was an obvious solution encompassed by the patent's claims;
- Any need or problem known in the field of endeavor at the time of invention and addressed by the patent can provide a reason for combining the elements in the manner claimed;
- "Common sense" teaches that familiar items may have obvious uses beyond their primary purposes, and in many cases a person of ordinary skill will be able to fit the teachings of multiple patents together like pieces of a puzzle;
- A person of ordinary skill in the art is also a person of ordinary creativity, and is not an automaton;
- A patent claim can be proved obvious by showing that the claimed combination of elements was "obvious to try," particularly when there is a design need or market pressure to solve a problem and there are a finite number of identified, predictable solutions such that a person of ordinary skill in the art would have had good reason to pursue the known options within his or her technical grasp; and
- One should be cautious of using hindsight in evaluating whether a claimed invention would have been obvious.

32. 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.

IV. BACKGROUND OF THE TECHNOLOGY

33. As discussed in Section V, the '044 patent is directed to an acoustic device for detecting target sounds in the environment. E.g., '044 patent, Abstract, 1:53-58. A processor coupled to an ambient microphone and receiver monitors target sounds within ambient sound, and creates a “sound signature” of the features of the target sound. '044 patent, 5:7-20. The processor compares the sound signatures to learned models to identify corresponding target sounds, and adjust the audio delivered to the user accordingly. '044 patent, 5:28-30, 5:38-39. For example, in one embodiment, when the target sound is detected, the audio is adjusted by way of the ear canal receiver passing, amplifying, attenuation or reproducing the target sound to the user's ear. '044 patent, 2:1-8. The purported inventive concept of the

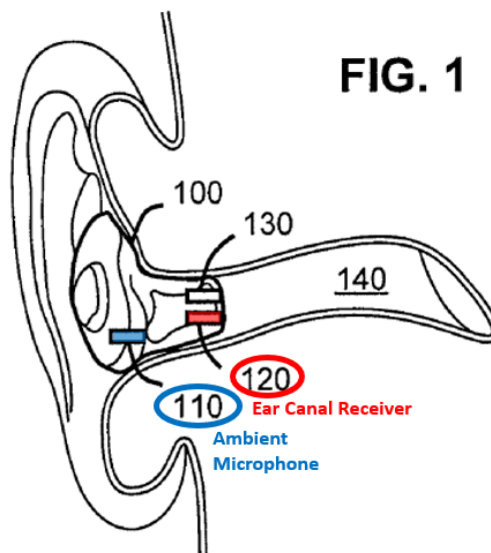
'044 patent is the comparing of the sound signature to learned models to detect a target sound. '044 patent, 2:40-44.

34. The technology at issue in the '044 patent was developed by others before the '044 patent. For example, Rast teaches an acoustic device wherein the processor compares a sound bite collected from the ambient sound to a stored sound, and when a match is found, adjusts the audio accordingly. *See e.g.*, Rast, [0051], [0053], [0059], [0060].

V. THE '044 PATENT

A. '044 Patent Overview

35. The '044 patent describes an acoustic device that “monitors” ambient sound for “target sounds.” '044 patent, 1-14-17. Figure one (below) depicts an earpiece with an **ear canal receiver** and **microphone**.



'044, Fig. 1, annotated

36. An **ambient sound microphone (ASM) 100** “captures ambient sound” “such as an alarm (e.g., bell, emergency vehicle, security system, etc.), siren (e.g. police car, ambulance, etc.), voice (e.g., ‘help’, ‘stop’, ‘police’. etc.), or specific noise type (e.g., breaking glass, gunshot, etc.).” ’044 patent, 3:36-40, 4:10-16. The processor 206 is “operatively coupled to the **ASM** and...**ECR** to monitor target sounds in the ambient sound.” ’044 patent, 1:53-58. Processor 206 detects “target sounds” within the environment by comparing learned models to the “sound signature” made up of identifiable features that characterize the sound. ’044 patent, 5:8-22, 11:42-47. Sound signatures are described as “certain identifiable features that characterize the sound,” such as “statistical properties or parametric properties,” like “prominent frequencies” or “principal components identifying the most likely recognizable features of a target sound.” ’044 patent, 5:10-20. The processor adjusts the audio delivered to the user through the **ear canal receiver (ECR) 120** based on the processor’s mode or state. ’044 patent, 5:38-40, 9:24-28. For example, in one embodiment wherein the device is in the “listening” state, the processor amplifies the detected target sound. ’044 patent, 5:39-42. In another example, the processor attenuates the audio content delivered to the ear canal, and amplifies the target sounds in the ear canal. ’044 patent, 5:46-49. The device “monitor[s] target sounds in the environment, present[s] notifications based on identified target sounds, adjust[s] audio content levels with respect to ambient sound levels, and filter[s]

sound” based on the mode or state of the processor as defined by the user. ’044 patent, 3:46-51. For example, the device may provide an “audible notification” to the user once a target sound is detected. ’044 patent, 10:44-46.

37. The ’044 patent’s alleged inventive concept is that the processor compares a sound signature to the learned models to identify the corresponding target sound. ’044 patent, 5:21-30.

B. The Prosecution History

38. The ’044 patent claims priority from application No. 11/966,457, filed 12/28/2007, which claims priority to Provisional application No. 60/883,013, filed 12/31/2006.

39. On 8/17/2011, the Examiner issued a non-final rejection, rejecting claims 1-3, 5-13, 15-18, 20 and 23, and objecting to claims 4, 14, 19, 21, 22, 24 and 25. ’044FH, 77-85.

40. Claims 1-3, 5, 713, 15-18, 20, and 23 were rejected under pre-AIA §102(b) as being anticipated by U.S. Patent Application Publication No. 2005/0111683 Chabries (“Chabries”). ’044FH, 81. The Examiner stated that Chabries discloses all limitations of claim 1: an acoustic device, comprising an ambient sound microphone (ASM) configured to capture ambient sound, at least one an ear canal receiver (ECR) configured to deliver audio to an ear canal, a processor operatively coupled to the ambient sound microphone where the processor monitors

ambient sound for a target sound. '044FH, 80 (citing Chabries, [0062], [0062], [0066], and [0072]). For similar reasons, claims 7, 12, 17, and 20 were rejected. '044FH, 81-83.

41. Claim 6 was rejected under pre-AIA §103(a) obviousness as being unpatentable over Chabries. '044FH, 84. The Examiner stated it would have been “obvious that the audible signals can come from anything such as a cell phone.” '044FH, 84.

42. Claims 4, 14, 19, 21, 22, 24, and 25 were objected to as being dependent upon a rejected base claim. '044FH, 84-85.

43. In response, Applicant amended claim 1 to add that the following limitation of the processor:

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

'044FH, 95.

44. Applicant also amended independent claims 7, 17, 20 and 23 to include additional limitations. '044FH, 95-98. Applicant amended independent claim 7 to include “by generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.”

'044FH, 95. Applicant amended claim 17 to include the limitation “where a voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.” '044FH, 97. Applicant amended claim 20 to include “where learned models are retrieved from a database, the sound signature is compared to the learned models, and the target sound is identified from the learned models in view of the comparison.” *Id.* Applicant amended claim 23 to include “an ear canal microphone,” and the limitation of “where the ear canal microphone is configured to emit an auditory warning when the processor identifies the onset.” '044FH, 97-98. Applicant canceled original claims 2, 19, 21 and 24. '044FH, 95-98.

45. Applicant argued that amendments to claims 17, 20 and 23 with the indicated allowable features of claims 19, 21 and 24 put them in condition for allowance. '044FH, 101. Applicant argued that amendment to claims 1 and 7 to include features similar to indicated allowable features of claim 21 put them in a condition of allowance. '044FH, 101.

46. Applicant argued claims 1 and 7 are allowable because Chabries “does not compare the sound signature generated from the ambient sound to a plurality of learned signature models to detect the target sound, as required by claims 1 and 7.” '044FH, 102. Applicant argued that Chabries instead “examines the constancy of the envelope of the input signature in several frequency bands.” '044FH, 102.

Applicant argued that thus, Chabries does not include all the features of claim 1 and 7. '044FH, 102.

47. Subsequently, claims 1, 3-18, 20-23, and 25 were allowed. '044FH, 111.

VI. LEVEL OF ORDINARY SKILL IN THE ART

48. I have been asked to assume the Claims are entitled to the December 31, 2006 filing date of provisional application No. 60/883,013 for the purposes of this declaration only.

49. I have been informed by Petitioner's 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.

50. Based on my knowledge and experience in the field and my review of the '044 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 engineering, audio engineering or the equivalent, and at least two years of experience with audio devices and/or audio signal processing, such as earphones, hearing aids and/or hearing protectors. A greater amount of education could compensate for fewer years of work experience, and vice versa.

51. 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.

VII. CLAIM CONSTRUCTION

52. I have been instructed that the Challenged Claims of the '044 patent are to be given their ordinary and customary meaning as understood by one of ordinary skill in the art at the time of the alleged invention in light of the specification and the prosecution history pertaining to the patent. I have followed these principles in my analysis set forth in this document.

53. I understand that the parties have not proposed constructions for the '044 patent in the Harman Litigation.

VIII. GROUNDS OF UNPATENTABILITY

54. As I explain above in Section IV and V, an audio device that detects target sounds within the ambient sound and adjusts audio was well-known in the art.

55. In this section, I will explain why the teachings or combinations of teachings from the following references render obvious each of the Challenged Claims:

Name	Ex.	Publication/ Patent No.	Filed	Published/ Issued
Rast	1004	U.S. 2001/0046304	4/24/2001	11/29/2001
Victorian	1005	EP 1,519,625A2	9/13/2004	3/30/2005
Green	1006	U.S. 2007/0038395	9/2/2004	2/15/2007
Betts	1007	U.S. 2009/0085873	2/1/2006	4/2/2009
Giesbrecht	1008	U.S. 2006/0293882	6/28/2005	12/28/2006
Nakazawa	1009	U.S. 4,325,142	7/25/1980	4/13/1982

I have been instructed to assume that each of these references qualifies as prior art.

56. As I explain below, the Challenged Claims are invalid based upon the following prior art grounds:

- **Ground 1:** Challenged Claims 1-11 and 13-18 of the '044 patent are anticipated and rendered obvious by Rast.
- **Ground 2:** Challenged Claims 2-6, 11, 17 18 and 21 of the '044 patent are rendered obvious by Rast in view of Victorian.
- **Ground 3:** Challenged Claim 12 of the '044 patent is rendered obvious by Rast in view of Green.
- **Ground 4:** Challenged Claim 14 of the '044 patent is rendered obvious by Rast in view of Betts.
- **Ground 5:** Challenged Claim 19 of the '044 patent is rendered obvious by Rast in view of Giesbrecht.
- **Ground 6:** Challenged Claims 20 and 21 of the '044 patent are rendered obvious by Rast in view of Nakazawa.

57. These Grounds above are not substantively the same as the art or arguments raised in the prosecution of the '044 patent. None of the references of the

above grounds was considered during prosecution. And the disclosures considered by the Examiner are not substantially the same as any of the references relied upon in the Grounds of this Petition. For example, none of the art cited during prosecution taught a processor coupled to a microphone and speaker, where the processor compares captured ambient sounds to stored sounds to detect a target sound, as Rast does. *See, e.g.*, Rast, [0017], [0051].

IX. SPECIFIC GROUNDS FOR CHALLENGE

58. As I explain above in Section IV and V, although the '044 patent purports to have invented an acoustic device and method of detecting target sounds in the environment through comparison, such devices and methods were well-known in the art. As described below, the cited prior art provides these teachings.

59. In this section, I explain why claims 1-11 and 13-18 of the '044 patent are anticipated and rendered obvious by the teachings of Rast (Ex. 1004), why claims 2-6, 11, 17-18 and 21 are further rendered obvious by the teachings of Rast in view of Victorian (Ex. 1005), why claim 12 is rendered obvious by the teachings of Rast in view of Green (Ex. 1006), why claim 14 is rendered obvious by the teachings of Rast in view of Betts (Ex. 1007), why claim 19 is rendered obvious by the teachings of Rast in view of Giesbrecht (Ex. 1008), and why claims 20-21 are rendered obvious by the teachings of Rast in view of Victorian and Nakazawa (Ex. 1009). I note that, in this discussion, I may focus on the particular language in specific claims, but it

should be understood that my analysis applies to the corresponding limitations of all claims containing identical or substantially identical language.

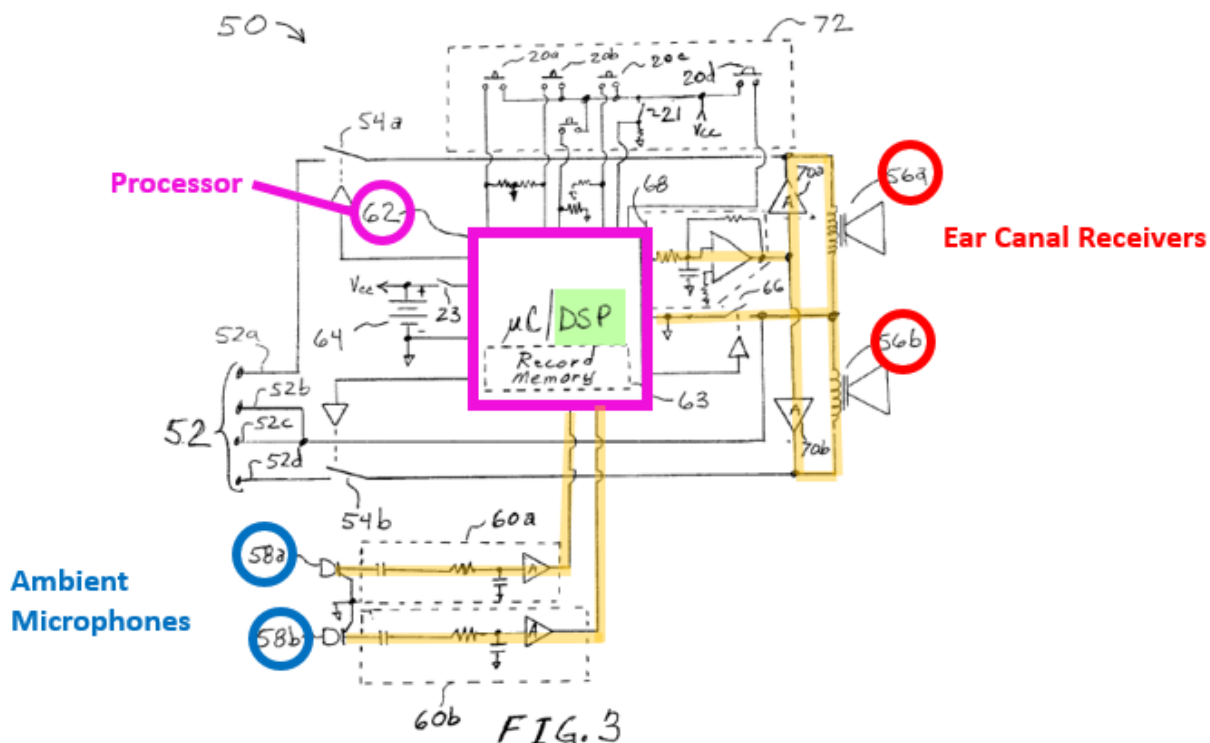
A. GROUND 1 (Rast, Claims 1-11, 13-18)

1. Overview of Rast

60. Rast discloses an apparatus and method for “sound selective acoustic isolation.” Rast, [0010]. **Ambient “[m]icrophones 58a, 58b**... convert sounds within the external acoustical environment to electrical signals” which are “pass[ed] through...into...inputs of a combined digital signal processor.” Rast, [0053]; Fig. 3. **Ambient microphones 58a, 58b** are “attached, or integrated, into the exterior of each earpiece.” Rast, [0014]. Rast teaches its system “is applicable to...earbud style devices” and “earbuds,” which are inserted into the ear. Rast, [0013]. Rast also teaches its system is applicable to “a pair of earpieces,” or a “single sided headset.” Rast, [0013], [0049]

61. “[**S**]peakers within the earpiece” such as “**conventional audio conversion devices 56 a, 56 b**” “deliver[] environmental audio” “toward the ear canal.” Rast, [0022], [0050], [0053].

62. “The **DSP/microcontroller 62** is capable of processing the audio signals from the **microphones 58 a, 58 b** and correlating these signals with a set of audio selection criterion which are stored in the sound characterization memory 63,” and outputs to the speaker per the mode of the device. Rast, [0053], Fig. 3.



Rast, Fig. 3

63. The **DSP** “continually compare[s] [sounds from the external environment] with the sounds and sound characteristics programmed in the sound correlation unit of the headset” by “correlat[ing] the external sounds and characteristics with programmed or recorded sounds or characteristics.” Rast, [0051], [0060], Fig 5. Rast refers to the stored information corresponding to the target sounds as “sound characterizations,” “sound characterization data,” “correlation sounds,” or “stored sound.” Rast, [0016], [0059]-[0060]. For example, the stored information can be an “alarm,” such as a “smoke alarm siren” or “fire alarm,” or “screams...[and] spoken phrases (i.e. ‘Hey!’, ‘Hello’, ‘Wearer's Name’,

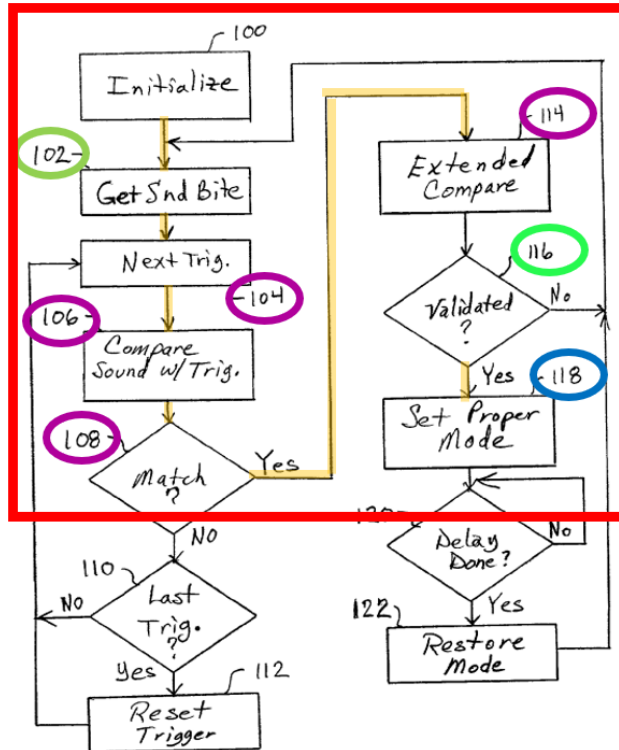
‘Help’),” which includes spoken words such as “Hey!” and “Help” that are associated with verbal warnings. Rast, [0021], [0059], [0062].

64. The **DSP** generates a “sound bite” including “waveforms, waveform characteristics, or characteristics of sound” from ambient sound recorded by the **microphones** and stored in the buffer. Rast, [0060]. The sound bite is saved in a “sound characterization memory 63” in the form of “random access memory (RAM) [that] has the ability to store sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated.” Rast, [0053]. The user presses “program button 88” to initiate “recording and downloading of sounds and sound characterization information for correlation with the external environment.” Rast, [0057].

65. “Segments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit of the headset.” Rast, [0051]. The **DSP** compares the sound bite to the stored sound characteristics to detect if the sound bite “matches” the stored “correlation sound.” Rast, [0059-0060], Fig. 5.

Generating a sound signature from the ambient sound

Comparing the sound signature to a plurality of learned signature models



Monitors the ambient sound for a target sound

Detecting the target sound

Adjusts the audio delivered to the ear canal based on the target sound

FIG. 5

Rast, Fig. 5

66. If a match is detected, the **DSP** adjusts the audio delivered through the **speaker** based on the mode of the headset set by the user. Rast, [0014], [0051]. For example, while the user is listening to music, when the ambient sound “correlates to a sufficiently high degree with the stored audio parameters,” “if the correlation sound was set for a sound that the user wished to be alerted to,” the DSP “passe[s]” the sound from the **microphones** to the listener. Rast, [0051], [0053], [0060]. Further examples of adjustments to the audio played to the user offered by **Rast** include generating an “alert” such as “warning tones,” “attenuat[ing], or

disconnect[ing]” the audio source, amplifying the microphone sounds, and repeating the **microphone** sounds (“echo mode”). Rast, [0051], [0059].

2. Invalidity of Claims 1-11, 13-18 Over Rast

67. As I discuss below, Claims 1-11 and 13-18 are disclosed and/or rendered obvious by Rast. In addition to my discussion in Sections I, IV, and V, the following discussion identifies example disclosures followed by bulleted excerpts that each disclose and/or render obvious Claims 1-11 and 13-18.

a. Element [1.pre]

68. **Element [1.pre]** recites “**An acoustic device, comprising.**” This element is met by Rast.

69. To the extent the preamble is limiting, Rast discloses an acoustic device (e.g., “apparatus which is acoustically coupled to the ear(s)”). *See, e.g.*, Rast:

- Fig. 1

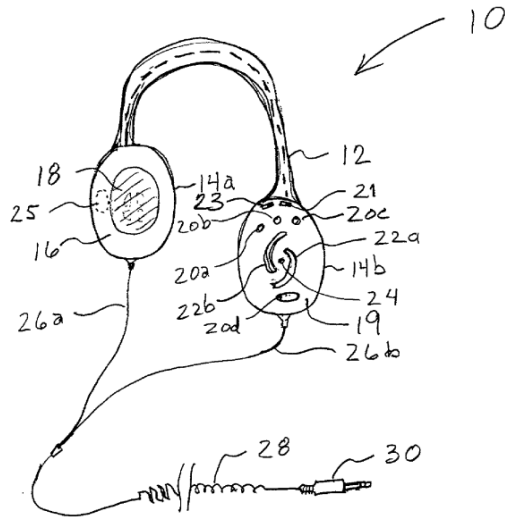


FIG. 1

- [0010]: “The present invention of an **apparatus for sound selective acoustic isolation** provides a number of benefits for users of audio related devices. By way of example the apparatus is exemplified as an environmentally responsive **apparatus which is acoustically coupled to the ear(s)** of a wearer to provide sound selective acoustical isolation from the environment. The **apparatus is typically worn by and acoustically coupled to a user**, referred to as the wearer.”
- [0013]: “**The apparatus is readily embodied within various forms of headsets, such as audio, communication, noise abatement, noise cancelation, and so forth.**”
- [0049]: “The headsets are exemplified for listening to an audio source over a wired connection, however, it will be appreciated that the present invention is **applicable to wireless headphones, communication headsets, noise abatement headsets, headsets incorporated into headgear such as helmets, single ear headsets, earbud style devices**, and so forth.”

70. Rast discloses an “apparatus which is acoustically coupled to the ear(s) of a wearer to provide sound selective acoustical isolation from the environment.”

Rast, [0010]. Figure 1 illustrates one such embodiment, though Rast describes

several applicable forms such as “wireless headphones,” and “earbud style devices.”

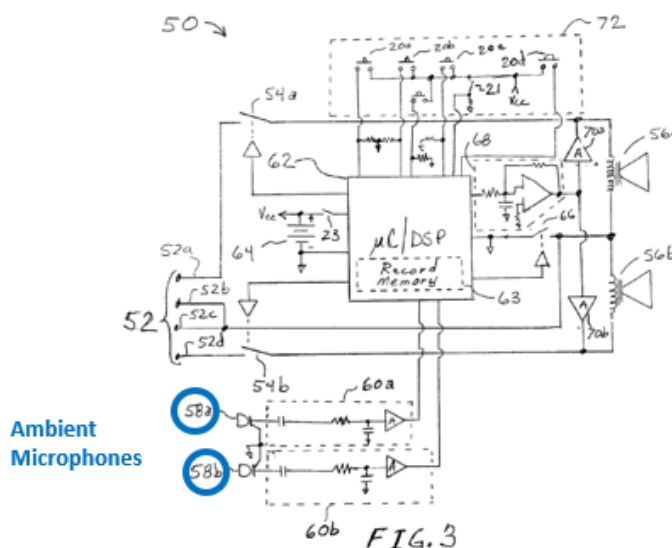
Rast, [0049].

b. Element [1.a]

71. Element [1.a] recites “an Ambient Sound Microphone (ASM) configured to capture ambient sound.” This element is met by Rast.

72. Rast discloses an Ambient Sound Microphone (ASM) configured to capture ambient sound (e.g., “[m]icrophones 58 a, 58 b...convert sounds within the external acoustical environment to electrical signals”). See, e.g., Rast:

- Fig. 3 (annotated)



- [Abstract]: “In response to correlation of the stored sound characteristics with sounds in the external environment the headset decreases acoustic isolation by coupling signals from one or more **external microphones** to the audio conversion elements within the earpieces.”
- [0014]: “**Sounds from the external environment are preferably registered by a microphones, with one microphone attached, or**

integrated, into the exterior of each earpiece. Each microphone converts the received external sound energy to an external sound signal.”

- [0053]: “FIG. 3 is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention...***Microphones 58 a, 58 b***, or similar audio conversion devices, ***convert sounds within the external acoustical environment to electrical signals*** which pass through the conditioning circuitry 60 a, 60 b...The DSP/microcontroller 62 is capable of processing the ***audio signals from the microphones 58 a, 58 b*** and correlating these signals with a set of audio selection criterion which are stored in the sound characterization memory 63...Once correlation occurs, ***the environmental sounds collected from the microphones*** are preferably passed through to the headset wearer for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the “hearthrough” button 20 d as shown in FIG. 1, to keep listening to the ***environmental audio***, or allow the duration to time out and return to listening to their original programming.”

73. Rast discloses “[m]icrophones 58 a, 58 b...convert sounds within the external acoustical environment to electrical signals,” and are “attached, or integrated, into the exterior of each earpiece.” Rast, [0014], [0053]. The “external acoustical environment” is ambient sound. Rast, [0053].

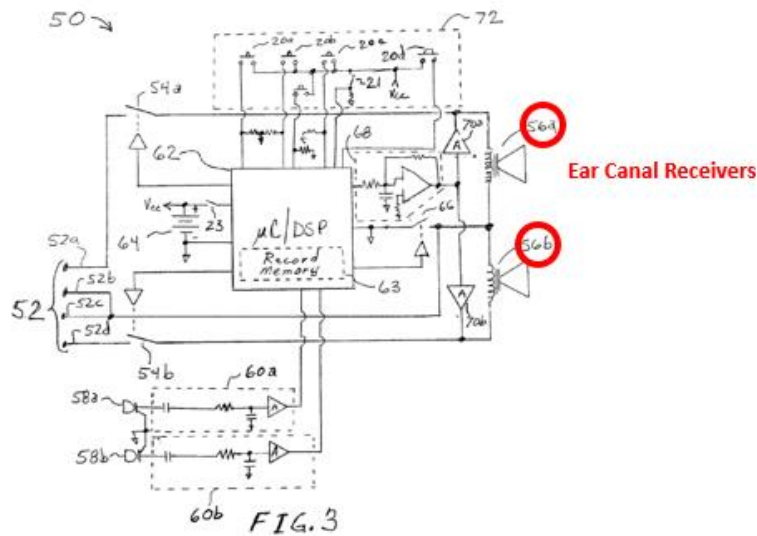
c. Element [1.b]

74. **Element [1.b]** recites “**at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal.**” This element is met by Rast.

75. Rast discloses an Ear Canal Receiver (ECR) (e.g., “speakers within the earpieces,” “conventional audio conversion devices 56 a, 56 b, such as speakers”)

configured to deliver audio to an ear canal (e.g., “delivers environmental audio to the wearer” “toward the ear canal”). See, e.g., Rast:

- Fig. 3 (annotated)



- [0242]: “In the audio device 10D, the **microphone 124D can be disposed in the frame 42D**. In particular, the microphone 124D can be disposed in the bridge portion 52D.”
- [0013]: “It will be appreciated, therefore, that **the apparatus referred to as a headset may be configured in various mechanical configurations**, such as mounted as earpieces within a helmet, single sided headset, **earbuds**, and so forth.”
- [0022]: “For example, one preferred method **delivers environmental audio** to the wearer during a predetermined interval by amplifying the signal from the microphone and playing it over **the speakers within the earpieces**.”
- [0023]: “One preferred method of changing the mode of the headphone is to temporarily couple the sound received from the microphone to the audio transducers, or **speakers, within the headset earpieces**.”

- [0049]: “The headsets are exemplified for listening to an audio source over a wired connection, however, ***it will be appreciated that the present invention is applicable to*** wireless headphones, communication headsets, noise abatement headsets, headsets incorporated into headgear such as helmets, single ear headsets, ***earbud style devices***, and so forth.”
- [0050]: “The audio collection grooves 22 a, 22 b and/or ridges which surround the microphone preferably mimic the acoustical function of the structures surrounding the exterior of the human ear, ***wherein sound is redirected toward the ear canal.***”
- [0053]: “FIG. 3 is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention. Audio source connections 52 a-52 d provide an analog audio channel which is routed through a pair of switches 54 a, 54 b to ***conventional audio conversion devices 56 a, 56 b, such as speakers.*** When listening to the audio source, the switches 54 a, 54 b are in the closed position. ...It will be recognized that the ***sound from the microphones is thereby selectively passed through to the listener*** when it correlates to a sufficiently high degree with the stored audio parameters. Once correlation occurs, the ***environmental sounds collected from the microphones are preferably passed through to the headset wearer*** for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the “hearthrough” button 20 d as shown in FIG. 1, to keep listening to the environmental audio, or allow the duration to time out and return to listening to their original programming.”

76. Rast discloses “speakers within the earpiece” such as “conventional audio conversion devices 56 a, 56 b” that “deliver[] environmental audio” “toward the ear canal.” Rast, [0022], [0050], [0053]. A person of ordinary skill in the art would have understood an ECR is a speaker that delivers audio to an ear canal, as an “ear canal receiver,” as Rast’s description states that it delivers environmental audio “toward the ear canal.” Rast, [0022], [0050]. Further, in one embodiment, the

'044 patent describes the earpiece as “partially or fully occluded in an ear canal,” but never describes the ear canal receiver as being occluded in the ear. *See, e.g.,* '044 patent, 2:16. Thus, a person of ordinary skill in the art would have understood that the ECR does not need to be within an ear canal.

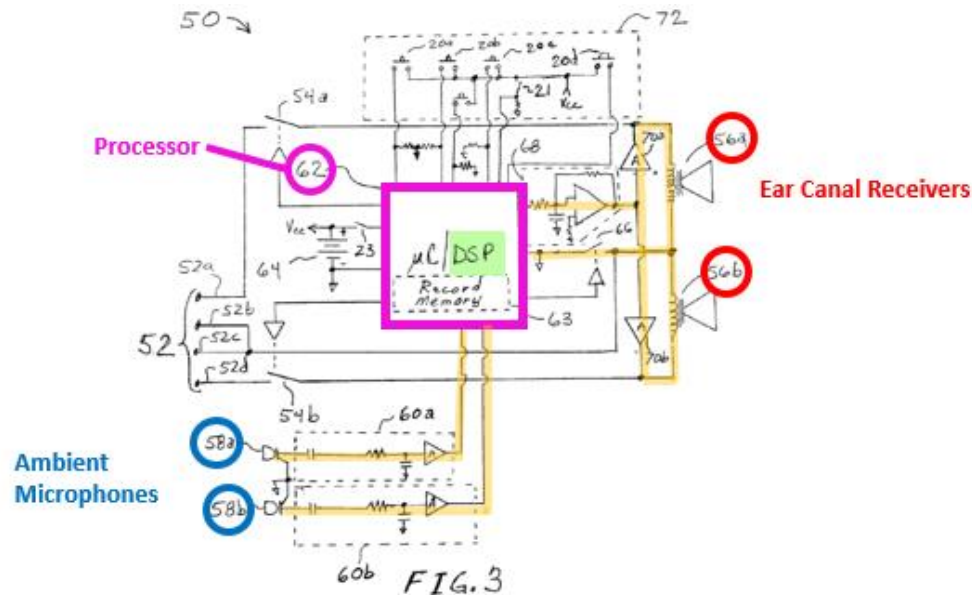
77. Regardless, Rast teaches its system “is applicable to...earbud style devices,” which are inserted into the ear. Rast, [0049]; *see also* Rast, [0013] (“It will be appreciated, therefore, that the apparatus referred to as a headset may be configured in various mechanical configurations, such as... earbuds”); *see also* Jaakkola, 1:28-2:5 (“Another prior art earbud headset is illustrated...wherein the earplug is inserted into the ear canal...”).

d. Element [1.c.1]

78. **Element [1.c.1] recites “a processor operatively coupled to the ASM and the ECR.”** This element is met by Rast.

79. Rast discloses a processor (*e.g.*, “digital signal processor” (“DSP”)) operatively coupled to the ASM and the ECR (*e.g.*, “processing the audio signals from the microphones” and “outputting...the sound collected...to earpiece speakers 56 a, 56 b”). *See, e.g.,* Rast:

- Fig. 3 (annotated)



- [0053]: “[A] combined **digital signal processor** and microcontroller 62 which contains an internal program store (ROM), random access memory (RAM), and has the ability to store sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated...The **DSP/microcontroller 62** is **capable of processing the audio signals from the microphones 58 a, 58 b** and...then the **DSP/microcontroller 62 within the present embodiment preferably switches off the input sources by opening switches 54 a, 54 b, engaging switch 66 to tie into the ground of the DSP with ground for the speakers 56 a, 56 b, and then outputting via a digital-to-analog converter the sound collected from the microphones to an amplifier circuit 68 which is then buffered by amplifiers 70 a, 70 b to the separate channels of the earpiece speakers 56 a, 56 b.** It will be appreciated that the correlation of sound and speech patterns is well known in the art. It will be recognized that the sound from the microphones is thereby selectively passed through to the listener when it correlates to a sufficiently high degree with the stored audio parameters. Once correlation occurs, the environmental sounds collected from the microphones are preferably passed through to the headset wearer for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the “hearthrough” button 20 d as shown in FIG. 1, to keep listening to the environmental audio, or allow the duration to time out and return to listening to their original programming. The use of the “hearthrough” button to control hearthrough may be configured in a variety of ways. One such way is to

interpret a quick button press as a change of state; from non-hearthrough mode to a short period of hearthrough, or from a hearthrough mode back to normal non-hearthrough mode. A button press of an intermediate duration, can be used to indicate that hearthrough mode should be entered for a slightly longer than conventional interval, regardless of original state. A button press of long duration may then be interpreted as a signal to lock the headset into hearthrough mode until a quick press of the “hearthrough” button occur.”

- [0063]: “Accordingly, it will be seen that this invention can be employed within various equipment, particularly headsets, for a variety of applications in acoustic environments. An analog headset embodiment has been shown and described, however, it will be recognized that digital headsets, which may provide class D output to the audio conversion elements, such as speakers, are anticipated within the invention. Specifically utilizing the invention with a digital input source simplifies the system as audio input is routed through the **DSP**/microcontroller on the way to the **speakers**, and power may be derived easily from the digital signal or a bias level embedded within that signal, as well as from battery sources. It will be appreciated that the circuits described for implementing the environmentally responsive headsets are but a single design embodiment to carry out the inventive teachings described, and that one of ordinary skill in the art can render innumerable variations without creative effort.”

80. See my discussion for element [1.a]-[1.b] above (§IX.A).

81. Rast further discloses DSP/microcontroller 62 is connected to the microphones (i.e., ASM) (see [1.a]) and speakers (i.e., ECR) (see [1.b]) because it “process[es] the audio signals from the microphones 58a, 58b” and “output[s]” it to the “speakers 56a, 56b,” as depicted in Figure 3. Rast, [0053].

e. Element [1.c.2]

82. **Element [1.c.2] recites “where the processor monitors the ambient sound for a target sound and adjusts the audio delivered to the ear canal based**

on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.” This element is met by Rast.

83. Rast discloses the processor (*see* [1.c.1]) monitors the ambient sound (*e.g.*, “[s]egments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit”) for a target sound (*e.g.*, for sound represented by “sound characterization”; “sound characterization data”; “correlation sound”; “stored sound”) and adjusts the audio delivered to the ear canal based on the target sound (*e.g.*, “sound from the microphones is...selectively passed through to the listener when it correlates to a sufficiently high degree with the stored audio parameters”; “the detected sound [is] repeated (echo mode),” “an alert [is] generated,” and/or “sounds...may be amplified” based on “the mode of the headset according to the configuration as set by the user”), the processor generating a sound signature from the ambient sound (*e.g.*, a portion of “sound being recorded from the microphones...placed in a circular buffer” is retrieved by “the processor” as a “sound bite” in the form of “waveforms, waveform characteristics, or characteristics of sound”) and comparing the sound signature to a plurality of learned signature models (*e.g.*, “[a] pointer to the correlation sound...is advanced at block 104 and the sound bite is compared at block 106” to “stored audio parameters” or “stored sounds”

or “correlation sound” or “sound characteristics”) to detect the target sound (e.g. a “determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics”). See, e.g., Rast:

- Fig. 5 (annotated)

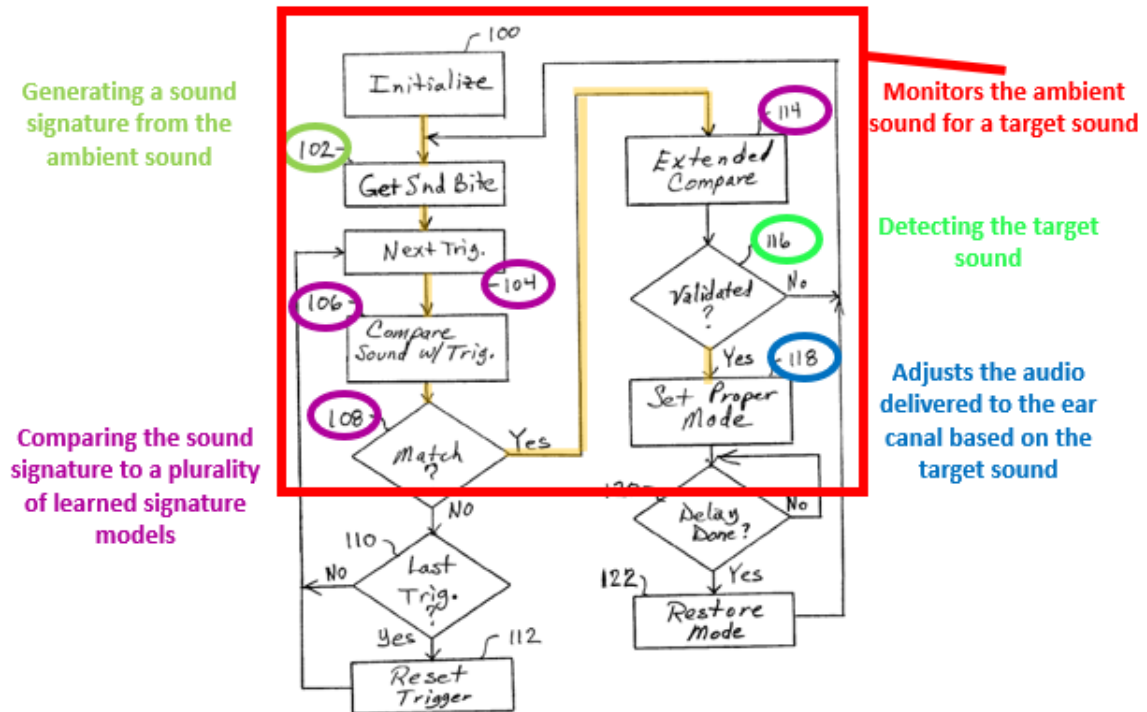


FIG. 5

- [0051]: “*Segments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit of the headset.*”
- Fig. 2

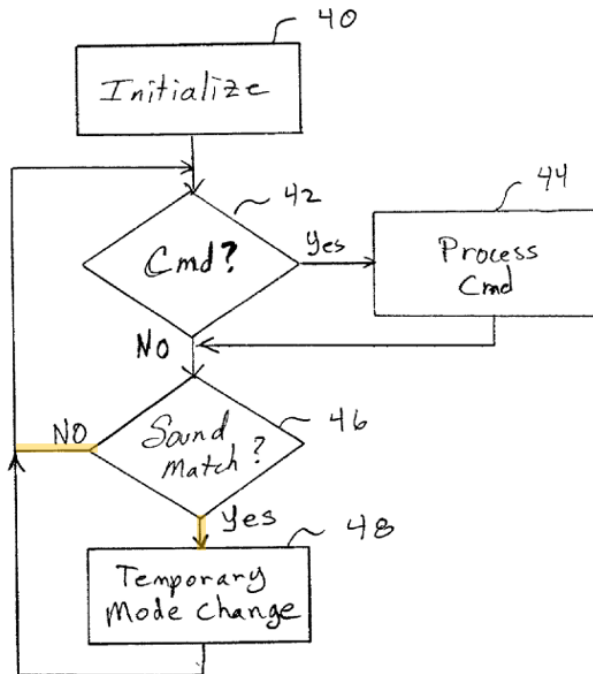


FIG. 2

- [0053]: “FIG. 3 is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention. Audio source connections 52 a-52 d provide an analog audio channel which is routed through a pair of switches 54 a, 54 b to **conventional audio conversion devices 56 a, 56 b, such as speakers**. When listening to the audio source, the switches 54 a, 54 b are in the closed position. ...It will be recognized that the **sound from the microphones is thereby selectively passed through to the listener when it correlates to a sufficiently high degree with the stored audio parameters**. Once correlation occurs, the **environmental sounds collected from the microphones are preferably passed through to the headset wearer** for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the “hearthrough” button 20 d as shown in FIG. 1, to keep listening to the environmental audio, or allow the duration to time out and return to listening to their original programming. The use of the “hearthrough” button to control hearthrough may be configured in a variety of ways. One such way is to interpret a quick button press as a change of state; from non-hearthrough mode to a short period of hearthrough, or from a hearthrough mode back to normal non-hearthrough mode. A button press

- of an intermediate duration, can be used to indicate that hearthrough mode should be entered for a slightly longer than conventional interval, regardless of original state. A button press of long duration may then be interpreted as a signal to lock the headset into hearthrough mode until a quick press of the “hearthrough” button occur.”
- [0058]: “Referring now to FIG. 4 a user interface 80 is shown for an environmentally responsive headset according to an aspect of the present invention. The interface allows the user to program the desired functionality of the headsets, such as operation, modes changes, as well as the selection of preprogrammed sounds, and both the **recording and downloading of sounds and sound characterization information for correlation with the external environment**. ... A program button 88 allows for the activation of **sound recording, or characterization**. It should be recognized that the duration of sound recorded is selected by the user in response to the retention of the program button, such that the user can select the amount of sound which is to be correlated prior to the occurrence of a mode change by the headset. Program button 88 is shown recessed to prevent inadvertent programming. A select button 90 allows selection of an item within a scrolling list shown on display 82. A clear button 92 allows for clearing a selected response, and for traversing in reverse order the prior screens. A set of up 94 and down 96 scroll buttons allow moving a selection up and down within the display for making user choices. A scrolling wheel 98, provides for altering the sensitivity of the sound discrimination circuit to the programmed sounds. The sensitivity preferably is a combination of both the amplitude of the sounds as well as the accuracy of the sounds in relation to the sounds and sound characteristics stored, and recorded within the environmentally responsive headset.”
 - [0059]: “Referring now to **Table 1**, by way of example and not of limitation, lists a series of commands which are preferably implemented within the headset. **The user is preferably capable of selecting all operating parameters of the selective acoustical isolation apparatus** through the user interface. The method of powering the device may be selectable depending on the specific form of headset being utilized, for example, power may in some cases be drawn from a wired connection that has a bias voltage, or from a rectified portion of the signal. In addition, the mode of activation and shutdown, for conserving battery power should be selectable. The mode of selective acoustical isolation to

be provided by the headset apparatus as well as ***the sounds can preferably be programmed***. A set of default characteristics which are to apply to the sounds which are to be heard in the environment may be selected. These may be overridden for any particular sound, however, the specification of a default reduces unnecessary selections by the user. Similarly, default characteristics for how to treat sounds that are to be blocked. It will be appreciated that sounds to be blocked may be ongoing “events”, wherein the selective acoustical isolation device continues to apply the attenuation to the given sound. The headset preferably is received by the user with a wide variety of pre-programmed sound characterizations, such as a telephone ring, smoke alarm siren, the spoken word “Hello”, and so forth. These sound characterizations are configured to provide matching with a variety of devices that need not produce the exact same sound. For example, it will be appreciated that telephone ringing cadence in the united states is fixed, wherein the discernment of ringing may be generally provided by a preprogrammed sound selection criterion. The user preferably can select which of these they wish to trigger a reduction in acoustical isolation. Similarly the user can select from preprogrammed sounds they wish to block, such as a car burglar alarm. ***The user may also record sounds that are specific to their environment which are to be used to modify the isolation provided by the headset. Prior to programming a sound that the user wants to be alerted to, the user may enter information about how to process the sound. For example, the user can select how the headset is to respond to the given sound, for instance should hearthrough be activated so that they may listen to the external environment, should the detected sound be repeated (echo mode), should an alert be generated.*** To improve correlation, the type of sound to be correlated can be selected by the user prior to recording. By selecting the type, the apparatus can store the proper sound characteristics to provide adequate correlation. For example ***sound characterization data stored*** for a spoken word is different than characterization data stored for a sound, or an alarm sound...***The user can also preferably select what is to be done with the audio source being received.*** The headset may be utilized for listening to music or for communication, wherein the headset is receiving signals that are being converted to audio by the audio conversion devices within the earpieces. ***The user can select if these signals are to be attenuated in response to the detection of a hearthrough match, for example, none (no attenuation), block (completely block), -10 dB (attenuate by 10 dB), -20 dB (attenuate by 20 dB), -40 dB (attenuate by 40 dB).*** The

duration of hearthrough can preferably be selected by the user, such as a duration in seconds. The relative volume to which hearthrough is to be configured is preferable set by the user. ***The user may then select to record a given sound, or to perform an audio “download” of sounds from an audio player device.*** User selection with regard to blocking selected sounds are similar in nature to the above although more limited in scope.”

- [0060]: ***“FIG. 5 is a flowchart of the method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics and changing the mode of the headset accordingly.*** The flowchart exemplifies a processor main loop, wherein command processing is considered to be in a separate module whose execution is in response to a processor interrupt. The electronics for the environmentally responsive headset are initialized for proper operation at block 100. This may take place after power up, or in response to the headsets being in a state of operation, such as positioned on the head of a user. The sound correlation method described is similar to that utilized in performing text string searches, in that the first character is compared for a match, and upon matching the first character, each subsequent character is checked until either a mismatch is found, or the search string is fully matched, at which time a match is returned. In the flowchart, the sound or characteristics thereof are matched a piece at a time. Preferably, distinctive sound elements, or characteristics are searched for thereby reducing the amount of processor overhead. ***The sound being recorded from the microphones is considered to be placed in a circular buffer, such as by a periodic interrupt, wherein the processor is capable of operating on information from the buffer. A sound bite is retrieved from the buffer at block 102.*** It will be appreciated that ***the sound bite may comprise waveforms, waveform characteristics, or characteristics of sound***, depending on the algorithms that are being utilized for providing correlation. A pointer to the correlation sound, or trigger condition, is advanced at block 104 and the sound bite is compared at block 106. ***If the sound does not match this particular correlation sound (stored sound selection criterion) then a check is made to determine if this is the last correlation sound in the list at block 110. If this is not the last correlation sound, then processing returns to block 104 which advances the correlation sound, trigger condition, to the next condition, and sound comparisons continue.*** The pointer to the correlation sound is reset at block 112 when all correlation sounds have

been checked against the sound bite. ***Upon the sound bite matching the correlation sound as detected at block 108, execution reaches an extended comparison routine, wherein subsequent portions, or characteristics, of the external sound are compared with subsequent portions of the stored correlation sound at block 114. A determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics.*** If the sound is found not to match the correlation sound, then execution returns to processing the sound bite with other correlation sounds, as shown at block 102. ***If the sound is validated then block 118 is reached which sets the mode of the headset according to the configuration as set by the user.*** For example, if the correlation sound was set for a sound that the user wished to be alerted to then the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound.”

- [0061]: “It will be appreciated that the invention can be implemented in a variety of ways for a number of applications without departing from the inventive teachings. One somewhat unique example involves the configuration of the headphones for use by infantrymen on the battlefield. The unit can be integrated into a helmet, or worn as a headset. The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while sounds such as voices, troop movements and so forth are amplified to provide a hearing advantage to the soldier. The headset itself would preferably be configured with a radio so that each infantryman may be retained in communication contact, such as with the squad leader. The headset preferably mixes the incoming audio with the ambient sounds so that the radio communication function is realized without the wearer losing audio contact with the environment. ***Upon detection of selected sounds in the external environment the incoming communication is blocked or attenuated and the external sounds are amplified to improve the safety of the infantryman.*** In addition, at the time of sound correlation, a radio signal may be forwarded to the squad leader, or otherwise utilized to signal the external event. To enhance the device for use within this application, the mechanical housing which surrounds the microphones should be preferably configured to direct sound in a similar pattern to the human ear, although with perhaps a larger surface area, wherein the wearer could still discern directional characteristics of ambient sounds in

a conventional manner. A similar utilization exist for a combat pilot, wherein the headset, typically with the earpieces incorporated into the flight helmet, provides for remote communication, the accentuation of critical audio, and the blocking of select noises, such as the noise of the particular engine. A second example would provide coaches, such as football, with a communication headset that allows them to still hear and communicate with players, while blocking crowd noise. Examples of use abound for those in high ambient noise environments, and those wearing communication headsets, such as pilots. In addition, it will be appreciated that disc-jockeys may find that headsets according to the invention provide the benefits of a studio quality sound reproduction headset while not preventing them from hearing necessary studio sounds, such as telephones, alerts, co-workers, and so forth. Perhaps the most common use may be for those persons wearing headphones to listen to music at work or at home, wherein they will be provided with a studio quality sound experience without the need of worrying about missing telephone calls, alarms, doorbells, or the utterances of another individual such as a spouse. Furthermore, the wearer can listen to the external environment, as triggered by sounds and speech, or as triggered by activating a hearthrough option, such as by pressing a button. The wearer is able to listen to the environment in these instances without the necessity of removing the headset from their head, or turning down the music.

- Table 1 (showing preferred command list)

TABLE 1

Preferred Command List
Setup-configure use
Options for: Power-off, Display use, Hear button duration, etc.
Alert settings-tone, volume, duration, etc.
Default, Hear-Default settings for sounds to be heard
Default, Block-Default settings for sounds to attenuate
Sounds Prerecorded, Hear: select any from an audio list (i.e. phone, siren, scream, doorbell, Hello, danger sounds . . .)
Sounds Prerecorded, Block: select any from an audio list (i.e. car alarm . . .)
Sounds 1 - n:
Mode: Through/Block
IF Through:
Hear by: Through, Repeat, Alert (check box)
Sound type: Spoken/Sound/Sound repetitive/Alarm (select)
Accuracy: required fit of sound (1 - n)
Amplification: (1 - n)
Audio source: NC, Block, -10 dB, -20 dB, -40 dB
Duration: time for pass through (1 - n)
Volume: (1 - n)
Record (start recording)
Download sound/algorithm (start recording)
IF Block:
Accuracy: required fit of sound (1 - n)
Attenuation: (1 - n)
Record (start sound recording)
Download sound/algorithm (start recording)

- [0010]: “The present invention of an apparatus for ***sound selective acoustic isolation*** provides a number of benefits for users of audio related devices. By way of example the apparatus is exemplified as an environmentally responsive apparatus which is acoustically coupled to the ear(s) of a wearer to ***provide sound selective acoustical isolation from the environment***. The apparatus is typically worn by and acoustically coupled to a user, referred to as the wearer. The acoustical isolation being characterized by the amount of sound attenuation in decibels which occurs as a result of the acoustical coupling of the device to the ear(s) of a wearer. ***An apparatus and method are described by which the acoustic isolation of the user is modified either manually based on user input, or automatically based on the sounds registered within the external environment and correlated to a set of stored sound characteristics.***”
- [0014]: “The environmentally responsive headsets exemplified in the present invention are capable of ***changing isolation mode in response to*** a manual trigger, or ***automatically in response to matching a sound from the external environment with a stored element of sound characterization information which is programmed within the apparatus***. Sounds from the external environment are preferably registered by a microphones, with one microphone attached, or

integrated, into the exterior of each earpiece. Each microphone converts the received external sound energy to an external sound signal. ***The mode change of the apparatus generally comprises altering the level of acoustical isolation accorded the wearer by the headset. The mode of the headset may be changed by any of a number of mechanisms, for example, by temporarily routing external sound to earpieces (a “hearthrough mode”), replaying the detected sound (temporally displaced sound—an “echo back”), generating an alert signal (“user alert tone”), attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation (“block matched sound”), amplifying the ambient sounds after extracting spurious noises, accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants), combinations of the preceding modes, and so forth.*** Furthermore, if the headset is configured for listening to a programmed sound, such as that being received over a wired connection, or wireless receiver, then the audio amplitude of the programmed audio source may be temporarily attenuated, or disconnected, such that the external sound may be more easily discerned by the wearer.”

- [0016]: “The apparatus of the present invention is capable of retaining information about acoustic events (***stored sounds***) ***to which external acoustic events (external sounds) may be correlated.*** For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”, however, it is to be recognized that these “stored sounds” may comprise any combination of sound representations, sound characterizations, sound related algorithms, sound correlation programming, sound scripting, sound signaturing, along with additional methods of storing information about audio waveforms.”
- [0017]: “One aspect of the present invention is a “hearthrough” mode which routes the acoustics of the environment, preferably at a user selected amplification level, for some period of time to the headset wearer when a hearthrough trigger occurs. The hearthrough trigger may comprise the manual activation of a hearthrough button by the wearer, or by automatic sound detection within the apparatus. In automatic sound detection a signal processing circuit receives signals from one or more microphones, or similar acoustic conversion devices that are preferably incorporated on the exterior portions of the earpieces, or otherwise attached to the headset. ***The signal processing unit compares the microphone signals against a set of stored selection criterion and upon***

finding a match alters the amount of acoustical isolation provided by the headset, and/or provides an alerting audio signal to the wearer. It is anticipated that the wearer will typically choose to have the headset temporarily change to a hearthrough mode which may be augmented with sound echo, alerts, and so forth. The wearer is thereby directly provided with audio from the external environment.”

- [0018]: “*The sounds to be correlated may be loaded into the apparatus as elements within a set of programmed sound selection criterion by way utilizing the apparatus to record sounds, or characteristics sounds, which are to be correlated.* Many of the sounds for which the user desires reduced acoustic isolation from the environment will be unique to their particular environment, for example their spoken name, the sound of the doorbell, and so forth. Alternatively, preprogrammed sounds may be selected for the selection criterion, loaded from a storage media such as a CD-ROM, or downloaded into the unit, either directly, as a sound, as a set of sound characterizations, as a program entity (algorithm), or as a combination thereof. For example, a single complex selection criterion of “siren” may comprise an algorithm that is capable of discerning a wide variety of common sirens in use. Utilizing this as a preprogrammed algorithm, or a downloaded algorithm would provide a broad discernment of sirens without the need to record one or more specific siren sounds.”
- [0030]: “Another object of the invention is to provide a wearer with increased *acoustical isolation from selected external sounds that sufficiently correlate with a one or more stored sounds for which the user is to be isolated* (blocked).”
- [0051]: “FIG. 2 is a flowchart of basic sound discrimination within the headsets of the present invention. The flowchart assumes that a series of sounds and sound patterns to be discriminated has previously been programmed. Upon power-up of the sound discriminator, such as by use of a power-on switch or the detection of head compression by hidden detection switch 25 of FIG. 1, the circuit is initialized at block 40 wherein the hardware and sounds are prepared for discerning the external events. A check is made to detect command entry at block 42, because the user at any time may elect to program a new sound or alter the mode characteristics of the headset. If a command entry is detected, such as the pressing of a button, then a command processing routine is entered, block

44, which interfaces with the user. ***Segments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit of the headset.*** To speed discernment, a small segment of audio is compared with all the stored trigger conditions, and upon meeting a first phase of correlation, the discernment process continues to check further portions of the sound against the stored sounds and characteristics. ***If the external sound correlates with the programmed sound at block 46, then the mode of operation of the headphone is changed at block 48 to that which has been programmed by the user.*** It will be appreciated that mode changes should generally comprise changing the amount of acoustical isolation from the environment. Such as by entering a hearthrough mode wherein sound registered by the microphones is conditioned and directed to the audio annunciators within the earcups. Additionally, the audio source being listened to may be either attenuated, or disconnected, such that the amplified external sounds from the microphones may be more clearly heard by the wearer. It will be appreciated, however, that the wearer may alternatively, or additionally be alerted to the correlated sound in a number of ways, such as alerts, and playing of the correlated sound, or the recorded correlation sound.

- [0052]: “Alternatively, when the correlated sound is one selected as a sound to be blocked by the headset, the mode change can provide ***enhanced attenuation of the pattern of sound in a number of ways.*** The sound pattern matching may be utilized to modulate an active noise cancelation circuit on headsets so equipped. ***Upon detecting the sound to be blocked, the headset may increase the selected level of noise cancelation from a low, or intermediate value, to a higher level of noise cancelation.*** The noise cancelation algorithms themselves can also accept the sound pattern as input, wherein the noise cancelation may be tuned to the sound to be blocked. In this way the noise cancelation becomes “selective sound noise cancelation” and is capable of providing high degrees of isolation to selected acoustic signature patterns without the necessity of attempting to cancel all external noise. It will be appreciated that the user can, therefore, configure an active noise cancelation circuit ***to attenuate all sounds by a given amount, while using the selective attenuation provided by the present invention to specifically attenuate (block) sounds which correlate to the stored sound.***”

- [0053]: “FIG. 3 is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention. Audio source connections 52 a-52 d provide an analog audio channel which is routed through a pair of switches 54 a, 54 b to conventional audio conversion devices 56 a, 56 b, such as speakers. When listening to the audio source, the switches 54 a, 54 b are in the closed position. Microphones 58 a, 58 b, or similar audio conversion devices, convert sounds within the external acoustical environment to electrical signals which pass through the conditioning circuitry 60 a, 60 b into analog-to-digital converter (ADC) inputs of a combined digital signal processor and microcontroller 62 which contains an internal program store (ROM), random access memory (RAM), and has the ability to store sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated. Power for the DSP/microcontroller is exemplified as battery power 64, although the power may be derived from a variety of sources, such as the bias voltage or signal voltage as provided on the input audio source connections 52 a-52 d. ***The DSP/microcontroller 62 is capable of processing the audio signals from the microphones 58 a, 58 b and correlating these signals with a set of audio selection criterion which are stored in the sound characterization memory 63. When the microphone input correlates with the stored parameters, such as detecting the word “Hello!”, then the DSP/microcontroller 62 within the present embodiment preferably switches off the input sources by opening switches 54 a, 54 b, engaging switch 66 to tie into the ground of the DSP with ground for the speakers 56 a, 56 b, and then outputting via a digital-to-analog converter the sound collected from the microphones to an amplifier circuit 68 which is then buffered by amplifiers 70 a, 70 b to the separate channels of the earpiece speakers 56 a, 56 b.*** It will be appreciated that the correlation of sound and speech patterns is well known in the art. It will be recognized that the sound from the microphones is thereby selectively passed through to the listener when it correlates to a sufficiently high degree with the stored audio parameters. Once correlation occurs, the environmental sounds collected from the microphones are preferably passed through to the headset wearer for a period of time, for example three to five to seconds (3 S-5 S), during which the user can decide to press and hold the “hearthrough” button 20 d as shown in FIG. 1, to keep listening to the environmental audio, or allow the duration to time out and return to listening to their original programming. The use of the “hearthrough” button to control

hearthrough may be configured in a variety of ways. One such way is to interpret a quick button press as a change of state; from non-hearthrough mode to a short period of hearthrough, or from a hearthrough mode back to normal non-hearthrough mode. A button press of an intermediate duration, can be used to indicate that hearthrough mode should be entered for a slightly longer than conventional interval, regardless of original state. A button press of long duration may then be interpreted as a signal to lock the headset into hearthrough mode until a quick press of the “hearthrough” button occur.”

- [0054]: “To aid in understanding the ***correlation of external sounds with sounds stored within the apparatus***, an example follows for a stored spoken sound of “hello”. ***Speech recognition characteristics relating to the spoken sound of the word “Hello!” are stored within the headset apparatus and the characteristics are compared with the sounds registered by the microphones and fed to the signal processing unit. Thereafter, if the spoken word “Hello” is registered by the microphone from the external environment, the headphone preferably switches to a hearthrough mode to communicate environmental sounds to the wearer.*** In addition, as it will be appreciated that the spoken word “hello” was blocked during correlation, the sound “hello” received from the environment and just correlated may be played, in essence as an echo upon which an additional alert tone may be superimposed to alert the wearer that a hearthrough correlation has occurred and to indicate that the replayed sound has been reproduced (thereby suffering from a temporal shift) so that the user will not be confused if the echo of “hello” is heard in concert with continued verbiage from the person who uttered the “hello”.”

84. See my discussion for element [1.a] and [1.c.1] above (§IX.A).

85. Rast further discloses the processor monitors the ambient sound for a target sound. For example, the DSP (see element [1.c.1]) “continually compare[s] [sounds from the external environment] with the sounds and sound characteristics programmed in the sound correlation unit of the headset” by “correlat[ing] the external sounds and characteristics with programmed or recorded sounds or

characteristics.” Rast, [0051], [0060], Fig 5. Note, Rast refers to the stored information corresponding to the target sounds as “sound characterizations,” “sound characterization data,” “correlation sounds,” or “stored sound.” Rast, [0016], [0059]-[0060].

86. Rast further discloses the processor adjusts the audio delivered to the ear canal based on the target sound. For example, while the user is listening to an audio source, perhaps music, when the ambient sound “correlates to a sufficiently high degree with the stored audio parameters” (target sound), the DSP “passe[s]” the sound from the microphones to the listener “if the correlation sound was set for a sound that the user wished to be alerted to.” Rast, [0051], [0053], [0060]. Rast discloses other adjustments to the audio played to the user, including generating an “alert” such as “warning tones,” “attenuat[ing], or disconnect[ing]” the audio source, amplifying the microphone sounds, and repeating the microphone sounds (“echo mode”). Rast, [0051], [0059]. Rast shares other examples: “block (completely block),” the sound, “-10 dB (attenuate by 10 dB),” “-20 dB (attenuate by 20 dB),” and “-40 dB (attenuate by 40 dB).” Rast, [0059]. The duration of hearthrough can be selected by the user, such as a duration in seconds, varying from a “short period of hearthrough” to a “long duration.” Rast, [0053]. Another example Rast provides; the user’s “music” may be “attenuated or disconnected during the period of time the environmental audio is being played.” Rast, [0022].

87. Rast further discloses the processor generates a sound signature from the ambient sound. For example, the DSP generates a “sound bite” including “waveforms, waveform characteristics, or characteristics of sound” from sound recorded by the microphones (see element [1.a]) and stored in the buffer. Rast, [0060]. As described in element [1.a], the processor “operat[es]” on the sound in the buffer, retrieving said “sound bite.” Rast, [0060].

88. Rast further discloses the processor compares the sound signature to a plurality of learned signature models to detect the target sound. The DSP compares the sound bite (i.e., the sound signature) to the stored sound characteristics (i.e., the learned signature models) to detect if the sound bite “matches” the stored “correlation sound” (also referred to as “stored sounds,” “sound characterizations,” “sound characteristic,” in Rast). Rast, [0059-0060], Fig. 5. This is learning as understood in the ’044 patent. For example, the ’044 patent describes “learn mode” where the user “initiate[s] a recording of ambient sounds in the environment,” which is “stored in the user defined database 504.” ’044 patent, 7:11-26. Thus, a person of ordinary skill in the art would have understood that Rast’s stored sound characteristics are “learned signature models” because, like the ’044 patent’s “learn mode,” Rast learns sound characterization information for target sounds by, for example, recording sounds from the environment and storing them. Rast, [0058-0060]; ’044, 7:11-26.

f. Claim [2]

89. Claim [2] recites “The acoustic device of claim 1, where the acoustic device is an earpiece, wherein the target sound is at least one among an alarm, a horn, and a noise.” See my discussion of the elements of claim 1 above (§IX.A). The added limitation of claim [2] is met by Rast.

90. Rast further discloses the acoustic device is an earpiece (e.g., “earpiece[]”), wherein the target sound (e.g., sound represented by “sound characterization”; “sound characterization data”; “correlation sound”; “stored sound,” see [1.c.2]) is at least one among an alarm, a horn, and a noise (e.g., “alarm”). See, e.g., Rast:

- [0013]: “The apparatus is readily embodied within various forms of headsets...A headset generally comprises a pair of **earpieces** that are capable of being retained in a position proximal the ear of a wearer. Typically, headsets (or headphones) are utilized for modifying the personal acoustic environment for a single wearer ... It will be appreciated, therefore, that the apparatus referred to as a headset may be configured in various mechanical configurations, such as mounted as earpieces within a helmet, **single sided headset**, **earbuds**, and so forth.”
- [0059]: “The headset preferably is received by the user with a wide variety of **pre-programmed sound characterizations**, such as a telephone ring, **smoke alarm siren**, the spoken word “Hello”, and so forth. These **sound characterizations** are configured to provide matching with a variety of devices that need not produce the exact same sound...For example **sound characterization data** stored for a spoken word is different than characterization data stored for a sound, or an **alarm sound**.”

- [0062]: “The microphone correlates a stored sound, or set of sound characteristics, such as screams, a siren, a fire alarm, and so forth, with sounds in the environment.”
- [0010]: “The present invention of an apparatus for sound selective acoustic isolation provides a number of benefits for users of audio related devices. By way of example the apparatus is exemplified as an environmentally responsive apparatus which is ***acoustically coupled to the ear(s) of a wearer*** to provide sound selective acoustical isolation from the environment. The apparatus is typically worn by and acoustically coupled to a user, referred to as the wearer. The acoustical isolation being characterized by the amount of sound attenuation in decibels which occurs as a result of the acoustical coupling of the device to the ear(s) of a wearer. An apparatus and method are described by which the acoustic isolation of the user is modified either manually based on user input, or automatically based on the sounds registered within the external environment and correlated to a set of stored sound characteristics.”
- [0020]: “To increase the accuracy of sound discernment it is preferable that the user select a sound “type” prior to recording a sound, options may include such types as ***general sound, speech, loud sounds, repetitive sounds, alarms, telephone rings***. The apparatus is thereby capable of extracting proper sound characterization from the sound being “recorded” from the environment. The improved characterization allows for greater discernment accuracy relative to wearer purpose. As an example, consider that correlation performed for the ***spoken sound “Robert”*** can be discerned from the external environment in a speaker independent manner utilizing speech recognition characterization of the sound. In contrast, if the sound were to be stored only as a general sound it would not be characterized as speech, wherein speaker independence is lost and discernment would be restricted to a very narrow interpretation.”
- [0021]: “A wide range of selection criterion may be used upon which to correlate audio signals received from the external environment. The environmental audio may be analyzed by numerous current methods, including parametric analysis, waveform comparisons, Fourier analysis, and in a larger context voice recognition. The selection criterion provides the ability for the signal processing unit to discern a variety of environmental acoustics, which by way of example and not of limitation

comprise: ***alarms, screams, horns, screeching brakes, phones ringing, spoken phrases*** (i.e. “Hey!”, “Hello”, “Wearer’s Name”, “Help”), ***along with additional acoustic patterns.***”

- [0049]: “FIG. 1. is a headset 10 according to an embodiment of the present invention whose audio attenuation of external audio sounds can be selectively changed in response to the correlation of those external sounds with sounds and sound characterizations stored within the headset apparatus. ***The headsets are exemplified for listening to an audio source over a wired connection, however, it will be appreciated that the present invention is applicable to wireless headphones, communication headsets, noise abatement headsets, headsets incorporated into headgear such as helmets, single ear headsets, earbud style devices, and so forth.*** The headset 10 is shown comprising an earpiece positioning member 12, earpieces 14 a, 14 b, with cushioned inner faces 16, an inner cup portion 18 which retains an audio annunciator directed toward, and acoustically coupled to, the ear canal. An exterior surface 19 comprises control buttons 20 a-20 d, programming switch 21, audio collection grooves 22 a, 22 b, power switch 23, microphone 24, and hidden enablement switch 25.”

91. See my discussion for element [1.c.2] and claim [1] above (§IX.A).

92. Rast further discloses its apparatus as “a pair of earpieces.” Rast, [0013]. Rast further discloses that its device may be in the form of a “single sided headset.” Rast, [0013]. Rast also discloses that “the present invention is applicable to...single ear headsets.” Thus, the Rast device may be implemented as a single sided device. A person of ordinary skill in the art would have understood, and at least found it obvious, that when the Rast device is implemented as a single sided headset, it would embody a single earpiece with a single speaker delivering audio based on control by the DSP. Further, because Rast describes that its device, as exemplified as a “headset” in Figure 1, is fully implementable as a single sided

device, and/or a “earbud[],” a person of ordinary skill in the art would have understood (or at least found it obvious) that when embodied in a single earpiece, the DSP (*see* [1.c.1]) (and all other aspects) would be included in the earpiece to advantageously make the audio device more compact, including by minimizing the wiring between the microphone and speaker coupled to the DSP. *See* Rast, Fig. 3, [0049].

93. Rast discloses the “sound characterization,” “sound characterization data,” “correlation sound” or “stored sound” includes an “alarm,” such as a “smoke alarm siren” or “fire alarm.” Rast, [0059], [0062].

g. Claim [3]

94. **Claim [3] recites “The acoustic device of claim 1, where the acoustic device is an earpiece, wherein the processor monitors the ambient sound for spoken words associated with verbal warnings.”** See my discussion of claim [1] above (§IX.A). The added limitation of claim [3] is met by Rast.

95. Rast further discloses the acoustic device is an earpiece (*see* [2]), wherein the processor (*see* [1.c.1]) monitors the ambient sound (*e.g.*, “sound bites” from “sound being recorded from the microphones” are “compared” to “sound characterizations”) for spoken words associated with verbal warning (*e.g.*, “spoken phrases (i.e. ‘Hey!’,...‘Help’)”).

- [0060]: “FIG. 5 is a flowchart of the method of correlating the external sounds and characteristics with programmed or recorded sounds or

characteristics and changing the mode of the headset accordingly. The flowchart exemplifies a processor main loop, wherein command processing is considered to be in a separate module whose execution is in response to a processor interrupt. The electronics for the environmentally responsive headset are initialized for proper operation at block 100. This may take place after power up, or in response to the headsets being in a state of operation, such as positioned on the head of a user. The sound correlation method described is similar to that utilized in performing text string searches, in that the first character is compared for a match, and upon matching the first character, each subsequent character is checked until either a mismatch is found, or the search string is fully matched, at which time a match is returned. In the flowchart, the sound or characteristics thereof are matched a piece at a time. Preferably, distinctive sound elements, or characteristics are searched for thereby reducing the amount of processor overhead. ***The sound being recorded from the microphones is considered to be placed in a circular buffer, such as by a periodic interrupt, wherein the processor is capable of operating on information from the buffer. A sound bite is retrieved from the buffer at block 102.*** It will be appreciated that the sound bite may comprise waveforms, waveform characteristics, or characteristics of sound, depending on the algorithms that are being utilized for providing correlation. A pointer to the correlation sound, or trigger condition, is advanced at block 104 and the sound bite is compared at block 106. If the sound does not match this particular correlation sound (stored sound selection criterion) then a check is made to determine if this is the last correlation sound in the list at block 110. If this is not the last correlation sound, then processing returns to block 104 which advances the correlation sound, trigger condition, to the next condition, and sound comparisons continue. The pointer to the correlation sound is reset at block 112 when all correlation sounds have been checked against the sound bite. Upon the sound bite matching the correlation sound as detected at block 108, execution reaches an extended comparison routine, wherein subsequent portions, or characteristics, of the external sound are compared with subsequent portions of the stored correlation sound at block 114. ***A determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics.*** If the sound is found not to match the correlation sound, then execution returns to processing the sound bite with other correlation sounds, as shown at block 102. If the sound is validated then block 118 is reached which sets the mode of the headset according to the

configuration as set by the user. For example, if the correlation sound was set for a sound that the user wished to be alerted to then the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound. The audio volume of programmed sounds being output from the audio transducers is preferably attenuated to a given decibel level in relation to the sound detected, so that the wearer can easily hear the external sounds. In addition, the wearer may be alerted with additional warning tones, or a playback of the sound taken from the environment or the correlation sound. If the mode change is a temporary one, such as to allow the wearer to hear the external environment for a period of time, then a delay is checked for completion at block 120. At the conclusion of the delay the mode of the headset may be restored to the original mode at block 122. It will be appreciated that the preceding flowchart is a simplified description of the sound discrimination program that will allow one of ordinary skill in programming sound discrimination systems to code the program.”

- [0021]: “The selection criterion provides the ability for the signal processing unit to discern a variety of environmental acoustics, which by way of example and not of limitation comprise: alarms, screams, horns, screeching brakes, phones ringing, spoken phrases (i.e. “**Hey!**”, “Hello”, “Wearer's Name”, “**Help**”), along with additional acoustic patterns.”
- [0008]: “It should be appreciated, however, that being isolated from every sound within the external acoustical environment can often be inconvenient, or even dangerous. A flight crew member may be unable to hear the siren of an emergency vehicle, **a scream, a shout, or an associate loudly calling his/her name**, while a party listening to music may miss an important phone call, be unable to hear the door bell, or miss their name being called out. Furthermore, the nonselective nature of the majority of noise cancelation devices provides insufficient attenuation of bothersome noises, such as the roar of an aircraft APU, while it strives to attenuate all external sounds.”
- [0012]: “***In automatic mode the apparatus monitors acoustics within the environment against a set of selection criterion and upon finding a match***, changes the mode of the headset wherein the amount of acoustical isolation provided is altered temporarily, or until a “reset condition” is received. Automatic changes in acoustic isolation may be similarly

coupled with attenuation of the incoming audio signals, as well as the generation of alerts, matching sound “echos”, and so forth to further **enhance user cognition of the external environment**.

- [0015]: “The selective acoustical isolation provided by the present invention preferably augments the operation of active noise cancelation circuitry within the headset to provide a number of additional benefits. It will be appreciated that noise canceling headphones also require the incorporation of a signal processing circuit coupled to one or more microphones so that the signal registered by the microphone(s) is modified into a negative feedback signal which modifies the program audio signal being received to thereby reduce the apparent receipt of noise acoustically coupled through the housing of the headset. The present invention **increases headset safety** and convenience by providing a hearthrough mode for the headset wearer, while **allowing the wearer to select which sounds are to be heard and which are to be blocked**.”
- [0016]: “**The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external acoustic events (external sounds) may be correlated**. For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”, however, it is to be recognized that these “stored sounds” may comprise any combination of sound representations, sound characterizations, sound related algorithms, sound correlation programming, sound scripting, sound signaturing, along with additional methods of storing information about audio waveforms.”
- [0017]: “One aspect of the present invention is a “hearthrough” mode which routes the acoustics of the environment, preferably at a user selected amplification level, for some period of time to the headset wearer when a hearthrough trigger occurs. The hearthrough trigger may comprise the manual activation of a hearthrough button by the wearer, or **by automatic sound detection within the apparatus. In automatic sound detection a signal processing circuit receives signals from one or more microphones**, or similar acoustic conversion devices that are preferably incorporated on the exterior portions of the earpieces, or otherwise attached to the headset. **The signal processing unit compares the microphone signals against a set of stored selection criterion** and upon finding a match alters the amount of acoustical isolation provided by the headset, and/or provides an alerting audio signal to the wearer. It is

anticipated that the wearer will typically choose to have the headset temporarily change to a hearthrough mode which may be augmented with sound echo, alerts, and so forth. The wearer is thereby directly provided with audio from the external environment.”

- [0020]: “To increase the accuracy of **sound discernment** it is preferable that the user select a sound “type” prior to recording a sound, options may include such types as general sound, speech, loud sounds, repetitive sounds, alarms, telephone rings. The apparatus is thereby capable of extracting proper sound characterization from the sound being “recorded” from the environment. The improved characterization allows for **greater discernment accuracy relative** to wearer purpose. **As an example, consider that correlation performed for the spoken sound “Robert” can be discerned from the external environment in a speaker independent manner utilizing speech recognition characterization of the sound.** In contrast, if the sound were to be stored only as a general sound it would not be characterized as speech, wherein speaker independence is lost and discernment would be restricted to a very narrow interpretation.
- [0054]: “To aid in understanding the correlation of external sounds with sounds stored within the apparatus, an example follows for a stored spoken sound of “**hello**”. Speech recognition characteristics relating to the **spoken sound of the word “Hello!”** are stored within the headset apparatus and the characteristics are compared with the sounds registered by the microphones and fed to the signal processing unit. Thereafter, if the spoken word “Hello” is registered by the microphone from the external environment, the headphone preferably switches to a hearthrough mode to communicate environmental sounds to the wearer. In addition, as it will be appreciated that the spoken word “hello” was blocked during correlation, the sound “hello” received from the environment and just correlated may be played, in essence as an echo upon which an additional alert tone may be superimposed to alert the wearer that a hearthrough correlation has occurred and to indicate that the replayed sound has been reproduced (thereby suffering from a temporal shift) so that the user will not be confused if the echo of “hello” is heard in concert with continued verbiage from the person who uttered the “hello”.
- [0059]: “The headset preferably is received by the user with a wide variety of pre-programmed **sound characterizations, such as a telephone**

ring, smoke alarm siren, the spoken word “Hello”, and so forth. These sound characterizations are configured to provide matching with a variety of devices that need not produce the exact same sound. For example, it will be appreciated that telephone ringing cadence in the united states is fixed, wherein the discernment of ringing may be generally provided by a preprogrammed sound selection criterion. ...***The user may also record sounds that are specific to their environment which are to be used to modify the isolation provided by the headset. ...***”.

- [0061]: “It will be appreciated that the invention can be implemented in a variety of ways for a number of applications without departing from the inventive teachings. ...The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while ***sounds such as voices***, troop movements and so forth are amplified to provide a hearing advantage to the soldier.... A second example would provide coaches, such as football, with a communication headset that ***allows them to still hear and communicate with players, while blocking crowd noise***. Examples of use abound for those in high ambient noise environments, and those wearing communication headsets, such as pilots. In addition, it will be appreciated that disc-jockeys may find that headsets according to the invention provide the benefits of a studio quality sound reproduction headset while not preventing them from hearing necessary studio sounds, such as telephones, ***alerts, co-workers***, and so forth. Perhaps the most common use may be for those persons wearing headphones to listen to music at work or at home, wherein they will be provided with a studio quality sound experience without the need of worrying about missing telephone calls, ***alarms***, doorbells, or the ***utterances of another individual such as a spouse...***”

96. See my discussion for elements [1.c.1]-[1.c.2] and claim [2] above (§IX.A).

97. Rast further discloses the DSP “operat[es] on information from the buffer,” including the “sound bite...retrieved from the buffer at block 102.” Rast, [0060]. The DSP compares this sound bite recorded from the environmental sound

to the sound characteristics set by the user, to find a match (*see* [1.c.1], [1.c.2]). Sound characterizations to which the received sound is compared also include “spoken phrases” such as “Hey!,” and “Help,” which are associated with verbal warnings. Rast, [0021]. Rast also discloses other spoken phrases such as “Hello,” and the “Wearer's Name.” Rast, [0021]. Other examples Rast provides are “voices” on the battlefield as those are likely associated with verbal warnings, and the “utterances of another individual.” Rast, [0061].

h. Claim [4]

98. **Claim [4] recites “The acoustic device of claim 1, where the acoustic device is an earpiece, further comprising a memory to store, responsive to a directive by a user of the acoustic device, at least one target sound captured by the ASM for learning.”** See my analysis of claim [1] above. The added limitation of claim [4] is met by Rast.

99. Rast further discloses that the acoustic device is an earpiece (*see* [2]), further comprising a memory to store (*e.g.*, “random access memory (RAM), [that] has the ability to store sound waveforms, sound characteristics, and sound profiles 63”; “sound characterization memory 63”) responsive to a directive by a user of the acoustic device, at least one target sound captured by the ASM for learning (*e.g.*, “user interface 80” allows “recording...of sounds and sound characterization

information for correlation with the external environment” using “program button 88”). See, e.g., Rast:

- [0016]: “***The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external acoustic events (external sounds) may be correlated.*** For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”....”
- [0018]: “The sounds to be correlated may be loaded into the apparatus as elements within a set of programmed sound selection criterion by way ***utilizing the apparatus to record sounds, or characteristics sounds, which are to be correlated.***”
- [0022]: “According to one aspect of the invention, ***the user can program the headset to recognize specific sounds within their specific environment.*** The specific sounds to be recognized (correlated) with the external environment may be contained within the headset as a prerecorded sound, or as ***a sound that is recorded by the user from the target environment.*** The repertoire of triggering sounds can therefore be modified according to the conditions under which the wearer wants to be alerted, or interrupted.”
- [0035]: “Another object of the invention is to provide the capability to ***record sounds and patterns of sound recognition*** for which a headset mode change is to take place upon detection.”
- [0053]: “Microphones 58 a, 58 b, or similar audio conversion devices, convert sounds within the external acoustical environment to electrical signals which pass through the conditioning circuitry 60 a, 60 b into analog-to-digital converter (ADC) inputs of a combined digital signal processor and microcontroller 62 which contains an internal program store (ROM), ***random access memory (RAM), and has the ability to store sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated.***...The DSP/microcontroller 62 is capable of processing the audio signals from the microphones 58 a, 58 b and correlating these signals with a set of audio selection criterion which are stored in the ***sound characterization memory 63.***”

- [0051]: “If a command entry is detected, such as the pressing of a button, then a command processing routine is entered, **block 44, which interfaces with the user.**”
- Fig. 2

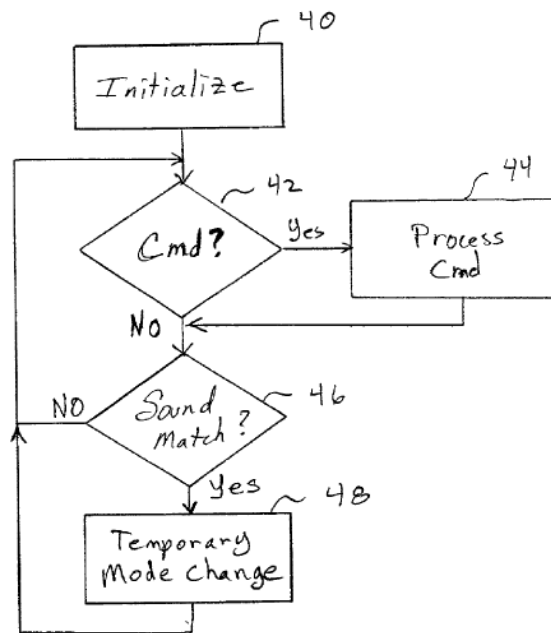


FIG. 2

- [0058]: “Referring now to FIG. 4 a **user interface 80** is shown for an environmentally responsive headset according to an aspect of the present invention. The interface **allows the user to** program the desired functionality of the headsets, such as operation, modes changes, as well as the selection of preprogrammed sounds, and both the **recording and downloading of sounds and sound characterization information for correlation with the external environment...** A **program button 88** allows for the activation of sound recording, or characterization.”
- Fig. 4

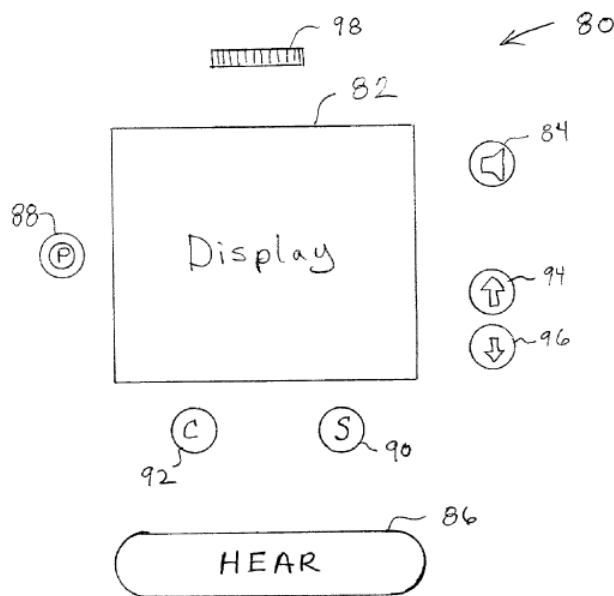


FIG. 4

- [0059]: “The mode of selective acoustical isolation to be provided by the headset apparatus as well as ***the sounds can preferably be programmed...The user may also record sounds that are specific to their environment which are to be used to modify the isolation provided by the headset.*** Prior to ***programming a sound that the user wants to be alerted to...By selecting the type, the apparatus can store the proper sound characteristics to provide adequate correlation....***”

100. See my discussion for claims [1] and [2] above (§IX.A).

101. Rast further discloses that the earpiece includes “sound characterization memory 63” in the form of “random access memory (RAM) [that] stores sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated.” Rast, [0053]. The user presses “program button 88” which “allows for the activation of sound recording, or characterization” Rast, [0058]. Thereby, the device learns the sound characterizations (in the form of

“sound waveforms, sound characteristics, and sound profiles”). Rast, [0053]. This is consistent with how the ’044 describes learning. For example, the ’044 describes a “learning mode” in which the user can “initiate a recording of ambient sounds in the environment....[and u]pon generating a sound signature for the new target sound, it can be stored in the user defined database 504.” ’044 patent, 7:20-26. And Rast describes the sounds as “stored sounds,” further demonstrating that the memory stores such sound. E.g., Rast [0016].

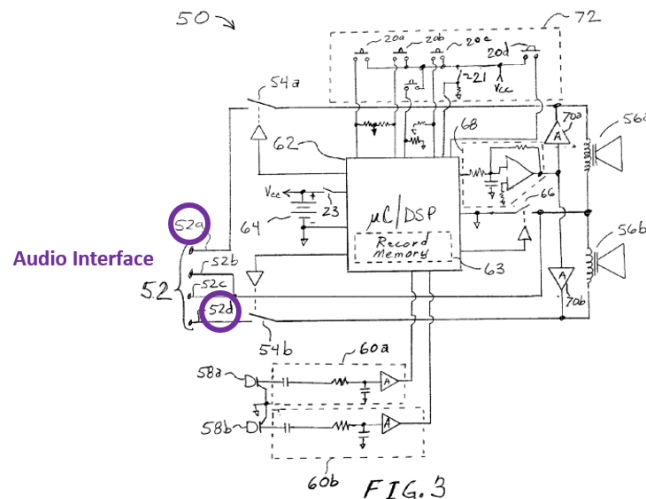
i. Element [5.a]

102. **Element [5.a] recites “The acoustic device of claim 1, where the acoustic device is an earpiece, further comprising an audio interface operatively coupled to the processor configured to receive audio content from a media player or cell phone,”** See my discussion of claim [1] above (§IX.A). The added limitation of claim [5.a] is met by Rast.

103. Rast further discloses the acoustic device is an earpiece (see [2]), further comprising an audio interface (e.g., “[a]udio source connections 52 a-52 d”) operatively coupled to the processor (e.g., “DSP” controls “switches 54 a, 54 b,” through which “an analog audio channel” from “audio source connections 52 a-52 d” are “routed”; “DSP...power may be derived from...bias voltage or signal voltage as provided on the input audio source connections 52 a-52 d”; “audio input is routed through the DSP/microcontroller on the way to the speakers”; sounds) configured to

receive audio content (e.g., receiving “musical audio”) from a media player (e.g., from “audio source,” “musical audio source”; “audio player device”).

- Fig. 3 (annotated)



- [0019]: ***“Preferably, the unit is configured so that it can be programmed with an audio stream from a source, such as the telephone, or audio from the Internet...”***
- [0025]: ***“The present invention may also be utilized within headsets which receive an audio channel, such as music, or communications, which is converted to sound within the earpieces to which the wearer is listening.”***
- [0051]: ***“FIG. 2 is a flowchart of basic sound discrimination within the headsets of the present invention...If a command entry is detected, such as the pressing of a button, then a command processing routine is entered, block 44, which interfaces with the user...Additionally, the audio source being listened to may be either attenuated...”***
- [0053]: ***“FIG. 3 is a simplified schematic of the electronics 50 for the environmentally responsive headphones according to an embodiment of the present invention. Audio source connections 52 a-52 d provide an analog audio channel which is routed through a pair of switches 54 a, 54 b to conventional audio conversion devices 56 a, 56 b, such as speakers.***

When listening to the audio source, the switches 54 a, 54 b are in the closed position...Power for the DSP/microcontroller is exemplified as battery power 64, although the power may be derived from a variety of sources, such as the bias voltage or signal voltage as provided on the ***input audio source connections 52 a-52 d.***”

- [0055]: “While ***listening to an audio source, such as music***, the wearer may press button 20 d and elect to listen to the environment; pressing of button 20 d again would return the headset to listening to the ***musical audio source.***”
- [0059]: “The user can also preferably select what is to be done with the ***audio source being received***...The headset may be utilized for listening to ***music*** or for communication, wherein the headset is receiving signals that are being converted to audio by the audio conversion devices within the earpieces... The user may then select to record a given sound, or to perform an audio “download” of sounds from an ***audio player device.***”
- [0063]: “An analog headset embodiment has been shown and described, however, it will be recognized that digital headsets, which may provide class D output to the audio conversion elements, such as speakers, are anticipated within the invention. Specifically ***utilizing the invention with a digital input source simplifies the system as audio input is routed through the DSP/microcontroller on the way to the speakers***, and power may be derived easily from the digital signal or a bias level embedded within that signal, as well as from battery sources.”

104. See my discussion for claims [1] and [2] above (§IX.A).

105. Rast further discloses the earpiece comprises “[a]udio source connections 52 a-52 d” that provide a means by which the earpiece receives analog audio from a “musical audio source”—a media player—and sends the audio to the speakers, such that the wearer “listen[s] to the musical audio source.” See Rast, [0053], [0055]. Further, given that Rast describes the “listening to an audio source, such as music,” a person of ordinary skill in the art would have understood (and

found it obvious) that the musical audio source is a “media player” because the audio from the musical audio source is already in a form for playback on the speaker. Rast, [0055]. A person of ordinary skill in the art would have understood (and found it obvious) that music is a type of media. As another example, Rast discloses its device is connected to an “audio player device”—a media player—from which a user may select to perform “an audio ‘download’ of sounds.” Rast, [0059].

106. Further, as explained in [5.b] below, the DSP is operatively coupled to the audio interface because the DSP controls the volume of audio received via audio source connections 52 a-52 d. This further demonstrates that Rast discloses a media player, for which the DSP controls the volume of media output. Further, **Rast** discloses, when digital audio is used, “audio input is routed through the DSP/microcontroller on the way to the speakers.” Rast, [0063].

j. Element [5.b]

107. **Element [5.b] recites “wherein the processor selectively adjusts a volume of the audio content delivered to the ear canal when the target sound is detected.”** This element is met by Rast.

108. Rast discloses that the processor (*e.g.*, “DSP”) selectively adjusts a volume of the audio content delivered to the ear canal (*e.g.*, “audio volume of programmed sounds being output from the audio transducers is preferably attenuated”; “attenuating the audio amplitude of the programmed sound source”

played by the speakers (see [1.b])) when the target sound is detected (e.g., when “sound detected...matches the correlation sound”). See, e.g., Rast:

- [0060]: “FIG. 5 is a flowchart of the method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics and changing the mode of the headset accordingly. ***The flowchart exemplifies a processor main loop...*** A determination is made at block 116 as to whether the ***sound detected in the environment matches the correlation sound, or characteristics...*** ***If the sound is validated then block 118 is reached which sets the mode of the headset according to the configuration as set by the user. For example, if the correlation sound was set for a sound that the user wished to be alerted to then the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound. The audio volume of programmed sounds being output from the audio transducers is preferably attenuated*** to a given decibel level in relation to the sound detected, so that the wearer can easily hear the external sounds.... It will be appreciated that ***the preceding flowchart is a simplified description of the sound discrimination program*** that will allow one of ordinary skill in programming sound discrimination systems to code the program.”
- [0059]: “These ***sound characterizations*** are configured to provide matching with a variety of devices that need not produce the exact same sound ... ***Prior to programming a sound that the user wants to be alerted to, the user may enter information about how to process the sound. For example, the user can select how the headset is to respond to the given sound, for instance should hearthrough be activated so that they may listen to the external environment, should the detected sound be repeated (echo mode), should an alert be generated...*** The user can also preferably select what is to be done with the audio source being received. The headset may be utilized for listening to music or for communication, wherein the headset is receiving signals that are being converted to audio by the audio conversion devices within the earpieces. The user can select if these signals are to be attenuated in response to the detection of a hearthrough match, for example, none (no attenuation), block (completely block), -10 dB (attenuate by 10 dB), -20 dB (attenuate by 20 dB), -40 dB (attenuate by 40 dB). The duration of hearthrough can preferably be selected by the user, such as a

duration in seconds. ***The relative volume to which hearthrough is to be configured is preferable set by the user.***”

- [0011]: “The manual change in acoustic isolation may be coupled with ***attenuation of incoming audio signals, such as music programming, or communications, so that the ability of the user to hear the external environment is improved.*** Attenuation of incoming audio signal preferably exceeds twenty decibels (20 dB).”
- [0012]: “Automatic changes in acoustic isolation may be similarly coupled with ***attenuation of the incoming audio signals***, as well as the generation of alerts, matching sound “echos”, and so forth to further enhance user cognition of the external environment.”
- [0051]: “Additionally, ***the audio source being listened to may be either attenuated, or disconnected***, such that the amplified external sounds from the microphones may be more clearly heard by the wearer.”
- [0058]: “A “hearthrough” button 86, provides for a manual activation of the hearthrough function wherein, external sounds picked up by the microphones are amplified and played within the earcups, preferably in concert with ***attenuating the audio amplitude of the programmed sound source.***”
- [0014]: “The mode change of the apparatus generally comprises altering the level of acoustical isolation accorded the wearer by the headset. ***The mode of the headset may be changed by any of a number of mechanisms, for example***, by temporarily routing external sound to earpieces (a “hearthrough mode”), replaying the detected sound (temporally displaced sound—an “echo back”), generating an alert signal (“user alert tone”), ***attenuating the sound*** by augmenting active noise cancelation with selective sound acoustical isolation (“block matched sound”), ***amplifying the ambient sounds after extracting spurious noises***, accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants), combinations of the preceding modes, and so forth. Furthermore, if the headset is configured for listening to a programmed sound, such as that being received over a wired connection, or wireless receiver, then the ***audio amplitude of the programmed audio source may be temporarily attenuated, or disconnected***, such that the external sound may be more easily discerned by the wearer.”

- [0022]: “The preferred method of changing the mode comprises **changing the level of acoustical isolation** provided by the headset to the wearer. For example, one preferred method delivers environmental audio to the wearer during a predetermined interval by amplifying the signal from the microphone and playing it over the speakers within the earpieces. **Preferably, any incoming audio source, such as music over a wired or wireless connection, is attenuated or disconnected during the period of time the environmental audio is being played.** The specific sounds to be recognized (correlated) with the external environment may be contained within the headset as a prerecorded sound, or as a sound that is recorded by the user from the target environment. The repertoire of triggering sounds can therefore be modified according to the conditions under which the wearer wants to be alerted, or **interrupted.**”

- Fig. 5

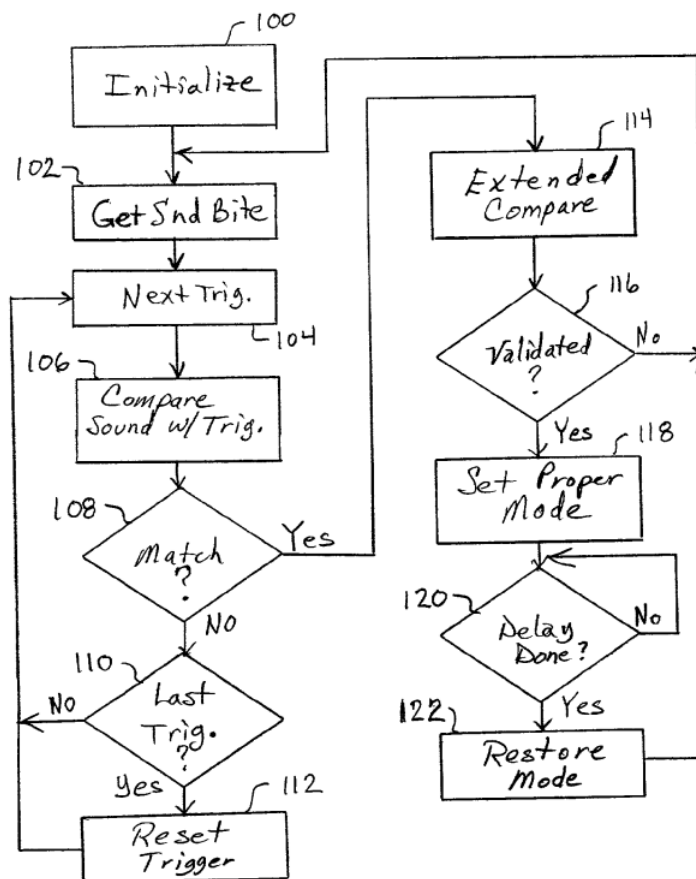


FIG. 5

109. See my discussion of claim [1] and element [5.a] above (§IX.A).

110. Rast further discloses that volume of audio playing in the speakers (see element [1.b]) is adjusted based on the headset's configuration, when "the sound detected in the environment matches the correlation sound" during the "processor main loop." Rast, [0060]; see Fig. 5. For example, "if the correlation sound was set for a sound that the user wished to be alerted to" then the "audio volume of programmed sounds being output from the audio transducers is preferably attenuated...so that the wearer can easily hear the external sounds." Rast, [0060]. A person of ordinary skill in the art would have understood (or at least found it obvious) that Rast's DSP controls the volume of the audio content. This is because as described in element [1.c.1], the DSP is coupled to the speakers. Further, the "processor main loop" illustrated in Figure 5 exemplifies functionality of the DSP, where at block 118 when a sound is matched, the mode of the headset is carried out according to the user configuration, including volume change. Rast, Fig. 5; Rast, [0060] (describing the processor main loop and block 118); see also Rast, [0059] (describing the "relative volume to which hearthrough is to be configured is preferably set by the user.") Further, for example, Rast discloses the algorithm shown in Fig. 5—including the attenuation of audio from the audio device—is part of a "sound discrimination program" performed in a "processor main loop." Rast, [0060]. As such, the attenuation, a volume change, is performed by a processor loop.

Further, as described in Rast, the “signal processing unit [(DSP)] compares the microphone signals against a set of stored selection criterion and upon finding a match alters the amount of acoustical isolation provided by the headset.” Rast, [0017]. Alteration of the amount of acoustic isolation, as exemplified by Rast and described above, includes the volume change. As such, the DSP controls the volume of the audio content.

k. Element [6.pre]

111. Element [6.pre] recites “**A method for personalized listening, the method comprising.**” This element is met by Rast.

112. Rast discloses a method for personalized listening (*e.g.*, “method...by which the acoustic isolation of the user is modified...based on user input”). *See, e.g.*, Rast:

- [0010]: “An apparatus and method are described by which the ***acoustic isolation of the user is modified either manually based on user input***, or automatically based on the sounds registered within the external environment and correlated to a set of stored sound characteristics.”
- [0017]: “One aspect of the present invention is a “hearthrough” mode which routes the acoustics of the environment, ***preferably at a user selected amplification level***, for some period of time to the headset wearer when a hearthrough trigger occurs.”
- [0018]: “Many of the sounds for which the user desires reduced acoustic isolation from the environment will be ***unique to their particular environment***, for example their spoken name, the sound of the doorbell, and so forth.”
- [0022]: “[*T*]***he user can program the headset to recognize specific sounds within their specific environment.***”

- [0028]: “Another object of the invention is to provide an *individual with the ability to selectively control the acoustical isolation between themselves and their acoustic environment.*”
- [0032]: “Another object of the invention is to *allow user programming of environmental acoustics against which external sounds are to be compared to trigger changes in the isolation provided by the acoustic barrier.*”

113. **Rast** discloses a method for isolating sounds “based on user input.”

[0010]. Rast further describes an object of its invention is to “provide an individual with the ability to selectively control the acoustical isolation between themselves and their acoustic environment.” Rast, [0028].

l. Element [6.a]

114. **Claim [6.a]** recites “**capturing ambient sound with an Ambient Sound Microphone (ASM).**” This element is met by Rast. See my discussion of element [1.a], above (§IX.A).

115. This limitation is substantively identical to element [1.a], which recites “an Ambient Sound Microphone (ASM) configured to capture ambient sound.”

116. Thus, this limitation is disclosed by Rast for the same reasons element [1.a] is disclosed by Rast.

m. Element [6.b]

117. **Element [6.b]** recites “**monitoring the ambient sound for a target sound by generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the**

target sound.” This element is met by Rast. See my discussion of element [1.c.2], above (§IX.A).

118. This limitation is substantively identical to element [1.c.2], which recites “where the processor monitors the ambient sound for a target sound and adjusts the audio delivered to the ear canal based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.” (Note: element [6.b] lacks the limitation of element [1.c.2] “by generating a sound signature from the ambient sound.”)

119. Thus, this limitation is disclosed by Rast for similar reasons element [1.c.2] is disclosed by Rast.

n. Element [6.c]

120. **Claim [6.c] recites “adjusting a delivery of audio by an Ear Canal Receiver (ECR) in an earpiece to an ear canal based on the target sound.”** This element is met by Rast. See my discussion of elements [1.b], [1.c.2] and claim [2] above (§IX.A).

121. This limitation is disclosed by Rast for the same reasons elements [1.b], [1.c.2] and claim [2] are disclosed by Rast. This limitation combines the prior limitations identified. Element [1.b] recites “at least one Ear Canal Receiver (ECR) configured to deliver audio to an ear canal,” element [1.c.2] recites “adjusts the audio

delivered to the ear canal based on the target sound” and claim [2] recites “where the acoustic device is an earpiece.” As such, Rast discloses the limitations of element [6.c].

o. Claim [7]

122. **Claim [7] recites “The method of claim 6, further comprising: passing the target sound to the ECR for delivery to the ear canal.”** See my discussion of claim [6], above (§IX.A). The added limitation of claim [7] is met by Rast.

123. Rast further discloses passing the target sound to the ECR for delivery to the ear canal (e.g., when “the sound being received by the microphone” matches the “correlation sound,” the “sound being received by the microphone” is “output as audio by the audio transducers in the earpiece so that the user can hear the sound.”) *See, e.g.*, Rast:

- [0060]: “[*I*]f the correlation sound was set for a sound that the user wished to be alerted to then the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound.”
- [0014]: “The mode change of the apparatus generally comprises altering the level of acoustical isolation accorded the wearer by the headset. The mode of the headset may be changed by any of a number of mechanisms, for example, by temporarily routing external sound to earpieces (a “hearthrough mode”), **replaying the detected sound (temporally displaced sound—an “echo back”)**, generating an alert signal (“user alert tone”), attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation (“block matched sound”), amplifying the ambient sounds after extracting spurious noises,

accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants), combinations of the preceding modes, and so forth. Furthermore, if the headset is configured for listening to a programmed sound, such as that being received over a wired connection, or wireless receiver, then the audio amplitude of the programmed audio source may be temporarily attenuated, or disconnected, ***such that the external sound may be more easily discerned by the wearer.***”

- [0023]: “In the case of hearing protection headsets (providing noise abatement) implemented according to the present invention; the external audio is attenuated by the earpieces, and optionally a noise canceling circuit within the headset, until the sound being received by the signal processing unit correlates with one or more of the stored selection criterion, ***at which time the mode of the headphone changes so that the wearer is made aware of the event associated with the stored selection criterion.*** One preferred method of changing the mode of the headphone is to ***temporarily couple the sound received from the microphone to the audio transducers, or speakers, within the headset earpieces.*** The ***wearer, therefore, is alerted*** and can clearly discern the danger themselves. The sound can be augmented, preferably depending on the duration, and nature of the sound being recognized, ***by playing back a recorded version of the precise received segment of received sound (an “echo”) which was correlated with the stored selection criterion***”
- [0024]: “Once a correlation occurs of a sound selected for hearthrough, the wearer is preferably provided with the environmental sounds for a period of time, such as three to five seconds (3S -5S), so that they may properly analyze the external environment to assess the situation. The wearer may at this time press a ***“hearthrough”*** button on the headset to continue listening to the environment. ***It will be appreciated that the “hearthrough” button would provide a similar result as removing the headset,*** however, it may be performed far more quickly and easily than taking off, holding the headsets, and then subsequently restoring them upon the ears.”
- [0054]: “To aid in understanding the correlation of external sounds with sounds stored within the apparatus, an example follows for a stored spoken sound of “hello”. Speech recognition characteristics relating to the spoken sound of the word “Hello!” are stored within the headset apparatus and the characteristics are compared with the sounds registered

by the microphones and fed to the signal processing unit. *Thereafter, if the spoken word “Hello” is registered by the microphone from the external environment, the headphone preferably switches to a hearthrough mode to communicate environmental sounds to the wearer...*”

124. See my discussion of elements [1.b], [1.c.2] and claim [6] above (§IX.A).

125. Rast further discloses when the environmental sound received by the microphone matches a correlation sound, “the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece.” Rast, [0014], [0060].

126. Rast further discloses a “hearthrough mode” wherein the target sound is passed to the user through the speakers, as Rast describes it “provide[s] a similar result as removing the headset.” Rast, [0024].

p. Claim [8]

127. **Claim [8] recites “The method of claim 6, further comprising: amplifying the target sound for delivery to the ear canal.”** See my discussion of claim [6], above (§IX.A). The added limitation of claim [8] is met by Rast.

128. Rast further discloses amplifying the target sound for delivery to the ear canal (*e.g.*, “[i]f the sound is validated,” “the sound being received by the microphone may be amplified and output as audio by the audio transducers”). See, *e.g.*, Rast:

- [0060]: “**A determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound**, or characteristics. If the sound is found not to match the correlation sound, then execution returns to processing the sound bite with other correlation sounds, as shown at block 102. **If the sound is validated** then block 118 is reached which sets the mode of the headset according to the configuration as set by the user. For example, if the correlation sound was set for a sound that the user wished to be alerted to then **the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound.**”
- [0022]: “For example, one preferred method delivers environmental audio to the wearer during a predetermined interval by **amplifying the signal from the microphone and playing it over the speakers within the earpieces.**”
- [0014]: “The mode of the headset may be changed by any of a number of mechanisms, for example, by...**amplifying the ambient sounds after extracting spurious noises...**”

129. See my discussion of claims [6], [7] and [19] above (§IX.A).

130. Rast further discloses, when the sound “received by the microphone” “matches the correlation sound,” “the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound.” Rast, [0060].

q. Claim [9]

131. Claim [9] recites “**The method of claim 6, further comprising: attenuating the target sound for delivery to the ear canal.**” See my discussion of claim [6], above (§IX.A). The added limitation of claim [9] is met by Rast.

132. Rast further discloses attenuating the target sound for delivery to the ear canal (e.g., “apply[ing] the attenuation to the given sound”). See, e.g, Rast:

- [0059]: “The mode of selective acoustical isolation to be provided by the headset apparatus as well as the sounds can preferably be programmed...**Similarly, default characteristics for how to treat sounds that are to be blocked. It will be appreciated that sounds to be blocked may be ongoing “events”, wherein the selective acoustical isolation device continues to apply the attenuation to the given sound...The user preferably can select which of these they wish to trigger a reduction in acoustical isolation. Similarly the user can select from preprogrammed sounds they wish to block, such as a car burglar alarm.**”
- [0014]: “The environmentally responsive headsets exemplified in the present invention are capable of changing isolation mode in response to a manual trigger, or automatically in response to matching a sound from the external environment with a stored element of sound characterization information which is programmed within the apparatus...The mode of the headset may be changed by any of a number of mechanisms, for example, by...**attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation** (“block matched sound”)...”
- [0049]: “FIG. 1. is a headset 10 according to an embodiment of the present invention **whose audio attenuation of external audio sounds** can be selectively changed in response to the correlation of those external sounds with sounds and sound characterizations stored within the headset apparatus.”

133. See my discussion of elements [1.b], [1.c.2] and [5.a] above (§IX.A).

134. Rast further discloses attenuating a correlation sound or sound characterization the user does not want to hear, like a “car burglar alarm.” Rast, [0014], [0059].

r. Claim [10]

135. Claim [10] recites “The method of claim 6, further comprising: **generating an audible message for delivery to the ear canal.**” See my discussion of claim [6], above (§IX.A). The added limitation of claim [10] is met by Rast.

136. Rast further discloses generating an audible message for delivery to the ear canal (*e.g.*, “the voiced phrase is played over the earpieces to alert the wearer that the environmentally responsive functionality is active”; “the wearer may be alerted with...playback of...the correlation sound”). *See, e.g.* Rast:

- [0023]: “A number of alternatives may be similarly produced, such as ***playing the recorded sound associated with the stored selection criterion***, or the playing of audio alert signals by themselves or in combination with the aforementioned sounds.”
- [0051]: “It will be appreciated, however, that the wearer may alternatively, or additionally be alerted to the correlated sound in a number of ways, such as alerts, and ***playing*** of the correlated sound, or ***the recorded correlation sound.***”
- [0055]: “In addition, an enable switch 25 as shown in FIG. 1, may be built into one or more of the earpieces so that ***when the headsets are placed on the wearers head the headset correlation functions will activate***, providing the power switch is set to on, ***at which time preferably a beep, tone sequence, or the voiced phrase is played over the earpieces to alert the wearer that the environmentally responsive functionality is active.***”
- [0059]: “The headset preferably is received by the user with a wide variety of pre-programmed sound characterizations, such as a telephone ring, smoke alarm siren, ***the spoken word “Hello”***, and so forth. ...”
- [0060]: “For example, if the correlation sound was set for a sound that the user wished to be ***alerted*** to then the sound being received by the microphone may be amplified and output as audio by the audio transducers in the earpiece so that the user can hear the sound...***In***

addition, the wearer may be alerted with additional warning tones, or a playback of the sound taken from the environment or the correlation sound. If the mode change is a temporary one, such as to allow the wearer to hear the external environment for a period of time, then a delay is checked for completion at block 120.”

- [0012]: “Automatic changes in acoustic isolation may be similarly coupled with attenuation of the incoming audio signals, as well as the ***generation of alerts***, matching sound “echos”, and so forth to further enhance user cognition of the external environment.”
- [0017]: “The signal processing unit compares the microphone signals against a set of stored selection criterion and upon finding a match alters the amount of acoustical isolation provided by the headset, and/or ***provides an alerting audio signal to the wearer.***”
- [0061]: “Perhaps the most common use may be for those persons wearing headphones to listen to music at work or at home, wherein they will be provided with a studio quality sound experience without the need of worrying about ***missing telephone calls, alarms, doorbells***, or the ***utterances of another individual such as a spouse***. Furthermore, the wearer can listen to the external environment, as triggered by sounds and speech, or as triggered by activating a hearthrough option, such as by pressing a button.”
- [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”)...***”

137. See my discussion of elements [1.b] and [1.c.2] above (§IX.A).

138. Rast further discloses, when the headset is put on the head of the user, a “voiced phrase” plays over the speakers to inform the user the headset is activated. Rast, [0055]. Rast further discloses that in addition to attenuating the audio being played when the environmental sounds match the sound characterization or

“correlation sound,” the device may also playback the correlation sound to alert the user. Rast, [0060]. For example, in the situation where the environmental sound is an audible message, like “Hello” ([0059]), then the device plays back the message “Hello” in the speakers (see element [1.b]). Rast further discloses generating “warning tones” and “alerts” to users, which are audible alarms, consistent with the ’044 patent. Rast, [0012], [0014], [0017], [0060]; ’044 patent, 5:65-6:9 (describing that the “audible alarm” may be an “ambulance,” or “fire engine.”)

s. Claim [11]

139. **Claim [11] recites “The method of claim 6, further comprising: mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound.”** See my discussion of claim [6], above (§IX.A). The added limitation of claim [11] is met by Rast.

140. Rast further discloses mixing (*e.g.*, “mix[ing]”) the target sound (*e.g.*, sound represented by “sound characterization”; “sound characterization data”; “correlation sound”; “stored sound,” see [1.c.2], including “voices, troop movements,” “artillery and gunfire”) with audio content delivered to the earpiece (*e.g.*, “incoming communication”) in accordance with a priority of the target sound (*e.g.*, “controlled attenuation, selective blocking”). See, *e.g.*, Rast:

- [0061]: ***“The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while sounds such as voices, troop movements and so forth are***

amplified to provide a hearing advantage to the soldier...The headset preferably mixes the incoming audio with the ambient sounds so that the radio communication function is realized without the wearer losing audio contact with the environment. Upon detection of selected sounds in the external environment the *incoming communication is blocked or attenuated and the external sounds are amplified* to improve the safety of the infantryman...A similar utilization exist for a combat pilot, wherein the headset, typically with the earpieces incorporated into the flight helmet, provides for remote communication, *the accentuation of critical audio, and the blocking of select noises, such as the noise of the particular engine*. A second example would provide coaches, such as football, with a communication headset that allows them to *still hear and communicate with players, while blocking crowd noise*. Examples of use abound for those in high ambient noise environments, and those wearing communication headsets, such as pilots. In addition, it will be appreciated that disc-jockeys may find that headsets according to the invention provide the benefits of a *studio quality sound reproduction headset while not preventing them from hearing necessary studio sounds, such as telephones, alerts, co-workers*, and so forth. Perhaps the most common use may be for those persons wearing headphones to *listen to music at work or at home, wherein they will be provided with a studio quality sound experience without the need of worrying about missing telephone calls, alarms, doorbells, or the utterances of another individual such as a spouse.*”

- [0014]: “The mode change of the apparatus generally comprises *altering the level of acoustical isolation* accorded the wearer by the headset. The mode of the headset may be changed by any of a number of mechanisms, for example, by temporarily routing external sound to earpieces (a “hearthrough mode”), replaying the detected sound (temporally displaced sound—an “echo back”), generating an alert signal (“user alert tone”), *attenuating the sound by augmenting active noise cancelation with selective sound acoustical isolation (“block matched sound”), amplifying the ambient sounds after extracting spurious noises, accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants), combinations of the preceding modes*, and so forth.”
- [0016]: “The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external

acoustic events (external sounds) may be correlated. For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”, however, it is to be recognized that these “stored sounds” may comprise any combination of sound representations, sound characterizations, sound related algorithms, sound correlation programming, sound scripting, sound signaturing, along with additional methods of storing information about audio waveforms.”

- [0018]: “For example, a single complex selection criterion of “siren” may comprise an algorithm that is capable of discerning a wide variety of common sirens in use.”
- [0022]: “According to one aspect of the invention, the user can program the headset to recognize specific sounds within their specific environment. When the signal processing unit within, or attached to the headset, recognizes the environmental audio as matching the criterion, then the mode of the headset is changed in response. The preferred method of changing the mode comprises ***changing the level of acoustical isolation*** provided by the headset to the wearer. For example, one preferred method delivers environmental audio to the wearer during a predetermined interval by ***amplifying the signal from the microphone and playing it over the speakers within the earpieces***. Preferably, any incoming audio source, such as music over a wired or wireless connection, is ***attenuated*** or disconnected during the period of time the environmental audio is being played.”
- [0023]: “***A number of alternatives may be similarly produced, such as playing the recorded sound associated with the stored selection criterion, or the playing of audio alert signals by themselves or in combination with the aforementioned sounds.***”

141. Rast discloses mixing the ambient sound with an “incoming communication” when a sound correlation occurs based on the desirability of the ambient sound. Rast, [0061]. The user sets preferences (prioritization) of target sounds and undesirable target sounds. Rast, [0061]. Target sounds that are desired, like “voices, [or] troop movements,” are amplified or “accentuated” (i.e.,

prioritized), while undesirable target sounds, like “artillery and gunfire,” are attenuated and/or blocked, before mixing the ambient sound with the incoming communication. Rast, [0061].

t. Claim [13]

142. **Claim [13] recites “The method of claim 6, further comprising: detecting and reporting from the sound signature a spoken utterance in the ambient sound associated with verbal warnings.”** See my discussion of claim [6], above (§IX.A). The added limitation of claim [13] is met by Rast.

143. Rast further discloses detecting (*e.g.*, “detecting”) and reporting (*e.g.*, “the wearer may be alerted with additional warning tones, or a playback of the sound taken from the environment or the correlation sound”) from the sound signature a spoken utterance in the ambient sound (*e.g.*, “sound bites” from “sound being recorded from the microphones” “compared” to “sound characterizations,” *see* [3]) associated with verbal warnings (*e.g.*, “spoken phrases (i.e. ‘Hey!’,...‘Help’),” *see* [3]). *See, e.g.* Rast:

- [0021]: “The selection criterion provides the ability for the signal processing unit to discern a variety of environmental acoustics, which by way of example and not of limitation comprise: alarms, **screams**, horns, screeching brakes, phones ringing, **spoken phrases (i.e. “Hey!”, “Hello”, “Wearer’s Name”, “Help”), along with additional acoustic patterns.**”
- [0053]: “**When the microphone input correlates with the stored parameters**, such as **detecting** the word “**Hello!**”, then the DSP/microcontroller 62 within the present embodiment preferably switches off the input sources....”

- [0059]: “The headset preferably is received by the user with a wide variety of pre-programmed sound characterizations, such as a telephone ring, smoke alarm siren, the spoken word **“Hello”**, and so forth.”
- [0060]: “In addition, the wearer may be **alerted with additional warning tones**, or a **playback of the sound** taken from the environment or the correlation sound.”
- [0054]: “To aid in understanding the correlation of external sounds with sounds stored within the apparatus, an example follows for a stored spoken sound of **“hello”**. Speech recognition characteristics relating to the spoken sound of the word **“Hello!”** are stored within the headset apparatus and the characteristics are compared with the sounds registered by the microphones and fed to the signal processing unit. Thereafter, if the spoken word **“Hello”** is registered by the microphone from the external environment, the headphone preferably switches to a hearthrough mode to communicate environmental sounds to the wearer. In addition, as it will be appreciated that the spoken word “hello” was blocked during correlation, the sound **“hello”** received from the environment and just correlated may be played, in essence as an echo upon which an additional alert tone may be superimposed to alert the wearer that a hearthrough correlation has occurred and to indicate that the replayed sound has been reproduced (thereby suffering from a temporal shift) so that the user will not be confused if the echo of “hello” is heard in concert with continued verbiage from the person who uttered the **“hello”**.”

144. See my discussion of elements [1.a], [1.c.2], and claims [3], [10], [12] above (§IX.A).

145. As I explained in claim [3], Rast discloses that sound characterizations (or “sound correlations”) can be the word “Hey!,” or “Help,” among other verbal audio and spoken phrases. Rast, [0021], [0053]. And as I described in element [1.c.2], when the environmental audio matches a sound correlation, like “Help,” the headset adjusts audio based on the user configuration or mode. One such

configuration includes the user being “alert[ed] with additional warning tones, or a playback of the sound taken from the environment or the correlation sound.” Rast, [0060]. When the sound correlation is a verbal warning, like “Hey!” and the headset configuration is set to playback the sound correlation, then the device detects “Hey!” in the ambient noise and reports the “Hey!” to the user through the speakers.

u. Claim [14]

146. Claim [14] recites “**The method of claim 6, further comprising: identifying the target sound from the sound signature and transmitting a warning notification to other devices.**” See my discussion of claim [6], above (§IX.A). The added limitation of claim [14] is met by Rast.

147. Rast further discloses identifying the target sound from the sound signature (*e.g.* a “determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics,” see [1.c.2]) and transmitting a warning notification to other devices (*e.g.*, “a radio signal...forwarded to the squad leader”). See, *e.g.*, Rast:

- [0060] (“***A determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics.***”)
- [0061] (“One somewhat unique example involves the configuration of the headphones for use by infantrymen on the battlefield. The unit can be integrated into a helmet, or worn as a headset. The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while sounds such as voices, troop

movements and so forth are amplified to provide a hearing advantage to the soldier. ***The headset itself would preferably be configured with a radio so that each infantryman may be retained in communication contact, such as with the squad leader.*** The headset preferably mixes the incoming audio with the ambient sounds so that the radio communication function is realized without the wearer losing audio contact with the environment. ***Upon detection of selected sounds in the external environment the incoming communication is blocked or attenuated and the external sounds are amplified to improve the safety of the infantryman. In addition, at the time of sound correlation, a radio signal may be forwarded to the squad leader, or otherwise utilized to signal the external event.***)

148. See my discussion of elements [1.c.2] and [6.b] above (§IX.A).

149. Rast further discloses the device determines whether the environmental sound matches the correlation sound or sound characteristic. Rast, [0060]. Rast further discloses one application for its invention is “headphones for use by infantrymen on the battlefield,” in which, when “artillery and gunfire” are detected, “at the time of sound correlation, a radio signal may be forwarded to the squad leader.” Rast, [0061]. The radio signal transmitted to the squad leader’s radio device is a warning notification to another device.

150. To the extent that it is determined that multiple “other devices” are required, it would have been obvious to forward the “radio signal” to multiple other members of a soldier’s squad. This has practical applications. For example, a person of ordinary skill in the art would have been motivated to cause the radio signal to be sent to multiple squad members to ensure that, even if the squad leader is unavailable

and/or the information is immediately relevant to other squad members, other squad members would be able to respond to the artillery and gunfire as appropriate.

v. Claim [15]

151. **Claim [15]** recites “**The method of claim 6, wherein the target sound is at least one among an alarm, a horn, a voice, and a noise.**” See my discussion of claim [6], above (§IX.A). The added limitation of claim [15] is met by Rast.

152. Rast further discloses wherein the target sound (*see* [1.c.2]) is at least one among an alarm, a horn, a voice, and a noise (*e.g.*, “alarm”; *see* [2]).

153. See my discussion of elements [1.c.2] and claim [2] above (§IX.A). The limitation added of claim [15] is substantively equivalent to claim [2], which recites, “wherein the target sound is at least one among an alarm, a horn, and a noise.”

w. Element [16.pre]

154. **Element [16.pre]** recites “**A method for sound signature detection, the method comprising.**” To the extent the preamble is limiting, this limitation is met by Rast.

155. Rast discloses a method for sound signature detection (*e.g.*, “method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics”).

- [0060] (“FIG. 5 is a flowchart of the ***method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics*** and changing the mode of the headset accordingly.”)
- [0058] (“The interface allows the ***user to program the desired functionality of the headsets***, such as operation, modes changes, as well as the selection of preprogrammed sounds, and ***both the recording and downloading of sounds and sound characterization information for correlation with the external environment.***”

156. Rast’s device performs a method of detecting “programmed...sounds” (sound signatures) from “external sounds.”

x. Element [16.a]

157. **Element [16.a] recites “capturing ambient sound with an Ambient Sound Microphone (ASM).”** This element is met by Rast. See my discussion of elements [1.a] and [16.pre] above (§IX.A).

158. This element is substantively identical to element [1.a], which recites “an Ambient Sound Microphone (ASM) configured to capture ambient sound.”

y. Element [16.b]

159. **Element [16.b] recites “receiving a directive to learn a sound signature within the ambient sound.”** This element is met by Rast.

160. **Rast discloses receiving a directive to learn a sound signature within the ambient sound** (*e.g.*, the user uses “program button 88” on “user interface 80” to initiate “recording...of sounds and sound characterization information for correlation with the external environment” which become “stored sound” in the “memory”). See, *e.g.*, Rast:

- [0016]: “***The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external acoustic events (external sounds) may be correlated.*** For simplicity, this retained information is often referred to as “*stored sounds*”, or “*sound characterizations*”....”
- [0053]: “Microphones 58 a, 58 b, or similar audio conversion devices, convert sounds within the external acoustical environment to electrical signals which pass through the conditioning circuitry 60 a, 60 b into analog-to-digital converter (ADC) inputs of a combined digital signal processor and microcontroller 62 which contains an internal program store (ROM), ***random access memory (RAM), and has the ability to store sound waveforms, sound characteristics, and sound profiles 63, to which the external sounds are to be correlated...***The DSP/microcontroller 62 is capable of processing the audio signals from the microphones 58 a, 58 b and correlating these signals with a set of audio selection criterion which are stored in the ***sound characterization memory 63.***”
- [0058]: “Referring now to FIG. 4 ***a user interface 80*** is shown for an environmentally responsive headset according to an aspect of the present invention. The interface ***allows the user to*** program the desired functionality of the headsets, such as operation, modes changes, as well as the selection of preprogrammed sounds, and both the ***recording and downloading of sounds and sound characterization information for correlation with the external environment...*** A program button 88 allows for the activation of sound recording, or characterization.”

161. See my discussion of claim [4] and element [16.pre] above (§IX.A).

162. Rast further discloses the user depresses program button 88 to initiate recording of ambient sound to store. Rast, [0016], [0053], [0058]. As described in [4], the resulting stored sound characteristics (sound signatures) are saved in memory, and thus learned, consistent with how the '044 patent describes learning. See, e.g., '044 patent, 7:20-26, discussed in claim [4].

z. Element [16.c]

163. Element [16.c] recites “where a voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.” This element is met by Rast.

164. Rast discloses where a voice command or an indication from a user is received (e.g., “user to program” by “pressing of a button”) and is used to initiate the steps of capturing and learning (e.g., to initiate “the recording...of sounds and sound characterization information for correlation with the external environment.”)

See, e.g., Rast:

- [0051]: “A check is made to detect **command entry at block 42, because the user at any time may elect to program a new sound** or alter the mode characteristics of the headset. If a command entry is detected, **such as the pressing of a button**, then a command processing routine is entered, block 44, which interfaces with the user.”
- [0058]: “Referring now to FIG. 4 a user interface 80 is shown for an environmentally responsive headset according to an aspect of the present invention. **The interface allows the user to program the desired functionality of the headsets, such as** operation, modes changes, as well as the selection of preprogrammed sounds, and **both the recording and downloading of sounds and sound characterization information for correlation with the external environment....A program button 88 allows for the activation of sound recording, or characterization.** It should be recognized that **the duration of sound recorded is selected by the user in response to the retention of the program button**, such that the user can select the amount of sound which is to be correlated prior to the occurrence of a mode change by the headset. Program button 88 is shown recessed to prevent inadvertent programming.”

165. See my discussion for elements [16.pre] and [16.b] above (§IX.A).

166. As I explained in element [16.b], Rast discloses the user depresses program button 88 to initiate recording sound to store, thereby learning the sound signature. The user depressing program button 88 is an indication from the user that initiates recording and learning, consistent with the '044 patent's "learn mode." See, e.g., '044 patent, 7:11-22.

aa. Claim [17]

167. Claim [17] recites **"The method of claim 16, further comprising: saving the sound signature locally on an earpiece or remotely to a server."** See my discussion of claim [16], above (§IX.A). The added limitation of claim [17] is met by Rast.

168. Rast further discloses saving the sound signature locally on an earpiece (e.g., "[s]peech recognition characteristics" "are stored within the headset apparatus," which may be an "earpiece[]"). See, e.g., Rast:

- [0051]: "To speed discernment, a small segment of audio is compared with all the ***stored trigger conditions***, and upon meeting a first phase of correlation, the discernment process continues to check further portions of the sound against the ***stored sounds and characteristics***."
- [0053]: "The DSP/microcontroller 62 is capable of processing the audio signals from the microphones 58 a, 58 b and correlating these signals with a set of audio selection criterion which are ***stored in the sound characterization memory 63***."
- [0054]: "***Speech recognition characteristics relating to the spoken sound of the word "Hello!" are stored within the headset apparatus*** and the characteristics are compared with the sounds registered by the microphones and fed to the signal processing unit."

- [0016]: “The apparatus of the present invention is capable of retaining information about acoustic events (**stored sounds**) to which external acoustic events (external sounds) may be correlated. **For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”...**”
- [0019]: “Preferably, the unit is configured so that it can be programmed with an audio stream from a source, such as the telephone, or audio from the Internet. When set in a program download mode, the audio source is started and the unit detects, interprets the incoming data, and converts the data to a programmatic entity which provides the desired sound discernment. A simple, albeit bandwidth inefficient, method of tonal code communication may be found in the use of DTMF, wherein sets of dual tones comprise the digits 0-9. It will be recognized that the speed at which digital data may be communicated as audio for an acoustically coupled unit is generally limited to approximately 2400-9600 baud. Aside from actual sound snippets, however, the data being loaded can comprise scripts and other sound related descriptions that can be encoded in less space than individual instructions for a DSP chip. To provide convenience to the user, a library of sounds may be provided, such as over the Internet, or from a CD, wherein the user can select prerecorded sound patterns, sound characterizations, and algorithms to suit their needs. It should, however, be appreciated that the simpler preprogrammed functions of the device may be adequate for a large percentage of intended applications. Alternatively, the apparatus of the present invention may be configured with a conventional jack or similar mechanism through which programming may be received.
- [0049]: “FIG. 1. is a headset 10 according to an embodiment of the present invention whose audio attenuation of external audio sounds can be selectively changed in response to the correlation of those external sounds with sounds and sound characterizations **stored within the headset apparatus**. The headsets are exemplified for listening to an audio source over a wired connection, however, it will be appreciated that the present invention is applicable to wireless headphones, communication headsets, noise abatement headsets, headsets incorporated into headgear such as helmets, single ear headsets, earbud style devices, and so forth. The headset 10 is shown comprising an earpiece positioning member 12, earpieces 14 a, 14 b, with cushioned inner faces 16, an inner cup portion 18 which retains an audio annunciator directed toward, and acoustically

coupled to, the ear canal. An exterior surface 19 comprises control buttons 20 a-20 d, programming switch 21, audio collection grooves 22 a, 22 b, power switch 23, microphone 24, and hidden enablement switch 25.”

169. See my discussion for claims [2] and [4] above (§IX.A).

170. Rast further discloses that sound characteristics are “stored” within the headset in the “sound characterization memory 63.” Rast, [0016], [0053]. As explained in [2], Rast discloses its headset implemented as an earpiece.

bb. Claim [18.pre]

171. **Claim [18.pre] recites “A method for personalized listening, the method comprising.”** See my discussion of element [6.pre], above (§IX.A). This element is disclosed by Rast.

172. This element is identical to element [6.pre].

cc. Element [18.a]

173. **Element [18.a] recites “capturing ambient sound via an earpiece that is at least partially occluded in an ear canal.”** This element is met by Rast.

174. Rast discloses capturing ambient sound (*see* [1.a], [6.a], [16.a]) via an earpiece that is at least partially occluded in an ear canal (*e.g.*, “earbud style devices,” *see* [1.b]). *See, e.g.*, Rast:

- [0013]: “It will be appreciated, therefore, that the apparatus referred to as a headset may be configured in various mechanical configurations, such as...*earbuds*...”

- [0014]: “Sounds from the external environment are preferably registered by a microphones, with one *microphone attached, or integrated, into the exterior of each earpiece.*”
- [0049]: “The headsets are exemplified for listening to an audio source over a wired connection, however, it will be appreciated that the present invention is applicable to...*earbud style devices*...”
- [0053]: “*Microphones 58 a, 58 b, or similar audio conversion devices, convert sounds within the external acoustical environment to electrical signals...*”

175. See my discussion for elements [1.a] and [1.b] above (§IX.A).

176. Rast further discloses microphones that are a part of the earpiece. Rast, [0014], [0053]. Rast further discloses “earbud style devices,” which are earpieces, and at least partially in the ear canal. Rast, [0049].

dd. Element [18.b]

177. Element [18.b] recites “**detecting a sound signature within the ambient sound that is associated with a target sound.**” This element is met by Rast.

178. Rast discloses detecting a sound signature within the ambient sound (e.g., “the sound bite is compared at block 106” to “stored audio parameters” or “stored sounds” or “correlation sound” or “sound characteristics”) that is associated with a target sound (e.g., “determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics”).

179. See my discussion for element [1.c.2] above (§IX.A).

180. This limitation is substantively identical to element [1.c.2], which recites “where the processor monitors the ambient sound for a target sound and adjusts the audio delivered to the ear canal based on the target sound, the processor generating a sound signature from the ambient sound and comparing the sound signature to a plurality of learned signature models to detect the target sound.”

ee. Element [18.c]

181. **Element [18.c] recites “mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound and a personalized hearing level (PHL).”** See my discussion of claim [11], above (§IX.A). This limitation adds to claim [11]’s limitations by requiring mixing of target sound with audio content in accordance with a “personalized hearing level (PHL).” Element [18.c] is met by Rast.

182. Rast discloses mixing the target sound with audio content delivered to the earpiece in accordance with a priority of the target sound (*e.g.*, “mix[ing] the incoming audio with the ambient sounds” based on “preferabl[e] configur[ation] for controlled attenuation [and] selective blocking,” *see* [11]) and a personalized hearing level (PHL) (*e.g.*, “a volume control to allow the wearer to set the audio level played over the earpieces”). *See, e.g.*, Rast:

- [0056]: “It will be appreciated that audio headphones are often equipped with ***a volume control to allow the wearer to set the audio level played over the earpieces.***”

- [0059]: “The user can select if these signals are to be attenuated in response to the detection of a hearthrough match, for example, none (no attenuation), block (completely block), −10 dB (attenuate by 10 dB), −20 dB (attenuate by 20 dB), −40 dB (attenuate by 40 dB)...***The relative volume to which hearthrough is to be configured is preferable set by the user.***”
- [0061]: “The headset preferably ***mixes the incoming audio with the ambient sounds*** so that the radio communication function is realized without the wearer losing audio contact with the environment.”

183. See my discussion for elements [5.b], [6.c] and claim [11] above (§IX.A).

184. As I described in claim [11], Rast discloses “mix[ing] the incoming audio with the ambient sounds” based on “preferabl[e] configur[ation] for controlled attenuation [and] selective blocking.” Rast, [0061]. Rast further discloses the “volume” of audio provided to the user—in this case, the mixed audio—is set by the user. Rast, [0056]. This is consistent with the ’044’s disclosure, which states “[t]he PHL establishes comfortable and uncomfortable levels of hearing,” and “can also be used to properly mix the volumes of the different sounds.” ’044, 5:42-50. Therefore, since Rast’s device mixes audio both based on the preferences of the user and the volume chosen by the user, this element is met by Rast.

ff. Element [18.d]

185. **Element [18.d] recites “where learned models are retrieved from a database, the sound signature is compared to the learned models, and the target**

sound is identified from the learned models in view of the comparison.” This element is met by Rast.

186. Rast discloses where learned models are retrieved from a database (*e.g.*, “a library of sounds may be provided, such as over the Internet, or from a CD, wherein the user can select prerecorded sound patterns, sound characterizations, and algorithms”), the sound signature is compared to the learned models, (*e.g.*, “signal processing circuit” “compares the microphone signals against a set of stored selection criterion”) and the target sound is identified from the learned models in view of the comparison (*e.g.*, “match” of “sound characteristics programmed.”).

See, e.g., Rast:

- [0019]: “Preferably, ***the unit is configured so that it can be programmed with an audio stream from a source***, such as the telephone, or audio from the Internet...***To provide convenience to the user, a library of sounds may be provided, such as over the Internet, or from a CD, wherein the user can select prerecorded sound patterns, sound characterizations, and algorithms to suit their needs.***”
- [0017]: “The signal processing unit ***compares the microphone signals against a set of stored selection criterion*** and upon finding a ***match*** alters the amount of acoustical isolation provided by the headset, and/or provides an alerting audio signal to the wearer.”
- [0051]: “Segments of sounds from the external environment are continually ***compared*** with the sounds and ***sound characteristics programmed*** in the sound correlation unit of the headset.”
- [0059]: “The user may then select to record a given sound, or to perform an audio ***“download” of sounds from an audio player device.***”

187. See my discussion for elements [1.c.2], [6.b], [18.b] and claim [17] above (§IX.A).

188. Rast discloses sound bites (sound signatures) of environmental sounds are compared to a set of stored selection criterion of sound characteristics (learned model) to find a match. Rast, [0051]. Rast further discloses downloading the stored sounds (i.e. learned models) from a “library”—which is a database—on the Internet or a CD. Rast, [0019]. “Preprogrammed sounds” may be “downloaded into the unit.” Rast, [0018]. Further, Rast describes a “program download mode” where the device can download such preprogrammed sounds. Rast, [0018].

B. GROUND 2: RAST IN VIEW OF VICTORIAN (Claims 2-6, 11, 17-18 and 21)

189. Claims 2-5 require “where the acoustic device is an earpiece.” As discussed for claims 2-5 in Section IX.A, Rast discloses or at least renders obvious that the acoustic device is an earpiece (e.g., “earpiece[]”). To the extent further disclosure is required, for example an argument that the entire acoustic device must be an earpiece, Victorian discloses an acoustic device (e.g., “audio communication system[]”) is an earpiece (e.g., “single ear-level device.”). *See, e.g.*, Victorian:

- Fig. 1

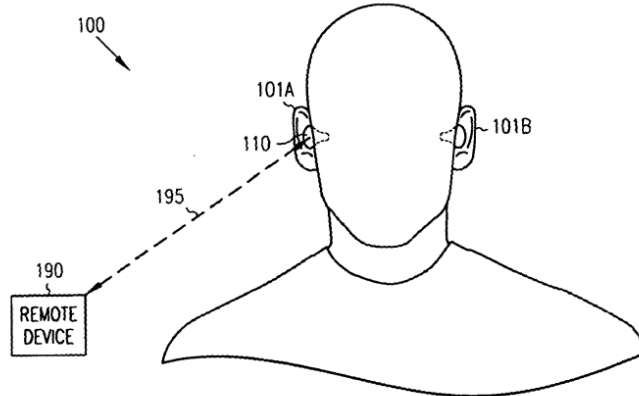


FIG. 1

- [0005]: “In another example, a full duplex system includes a **single headphone** to transmit and receiving voice, again with a VOX circuit in a control unit wired to the headphone.”
- [0008]: “Ear level full duplex **audio communication systems** each include **one** or two **ear attachments devices**, such as **in-the-ear (ITE)** or **behind-the-ear (BTE)** devices, that wirelessly communicates to a remote device.”
- [0013]: “This ear-level communication system uses a **single ear-level device**.”
- [0015]: “It should be noted that references to "an", "one", or "various" embodiments in this disclosure are not necessarily to the same embodiment, and such references contemplate more than one embodiment.”
- [0022]: “FIGS. 3A, 3B, and 3C are illustrations of...**one of ear- level devices 210A and 210B**...It is to be understood that each of ear- level devices 110, 210A, and 210B can be either a device configured **for right ear attachment or a device configured for left ear attachment**. Generally, a device configured for right ear attachment has an exterior configuration that is substantially symmetrical to that of a device of the same type configured for left ear attachment. In system 100, a single ear-level device 110 is a device configured either for right ear attachment or for left ear attachment.”

- [0029]: “FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the **single ear-level device 110**. The circuit is housed in an earmold configured for use as one of the device types illustrated as ear-level devices 310A, 310B, or 310C.”
- [0030]: “Ear-level device 110 includes an external microphone 420, an **ear canal microphone 440.... Ear canal microphone 440 detects sound from the ear canal**. The sound includes primarily speech of the person wearing ear-level device 110.”

190. Victorian is directed to an “ear-level” “audio communication system[]” used as a “single ear-level device.” Victorian, [0008], [0013], [0015]; *see also* [0005], [0022]. The single ear-level device is in the form of an “earmold” that “is fully occluded to maximize audio efficiency.” Victorian, [0029], Fig. 1. The ear-level device 110 includes “ear canal microphone 440 [that] detects sound from the ear canal. Victorian, [0030].

191. Rast and Victorian are analogous art. Rast and Victorian are in the same field as the '044 patent, which includes acoustic sensory devices. Rast and Victorian are reasonably pertinent to alleged problems addressed by the '044 of “enhancing the user’s ability to hear target sounds in their environment without compromising his hearing.” ’044, 1:45-48; *e.g.*, Rast, [0012] (“further enhance user cognition of the external environment”); Victorian, [0054] (“[D]etecting a sound from the environment and transmit it to the ear canal.”) Both Rast and Victorian deal with bettering a user’s ability to interact with their audio environment. For example, Victorian’s device “allows a hearing impaired individual to communicate”

with their environment, which is pertinent to the problem addressed by the '044 which looks to enhance a user's awareness of their environment. '044, 1:40-42; Victorian, [0009]. Likewise, Rast emphasizes the importance of “controlling acoustic isolation, so that important events are not missed.” Rast, [0009].

192. A person of ordinary skill in the art would have been motivated to apply Victorian's teachings of a single ear-level acoustic device to Rast's acoustic device for a more discrete, “not easily visible” system. Victorian, [0007]. Rast encourages the application of its invention to a one-ear device. *See, e.g.*, Rast [0049] (“[T]he present invention is applicable to ...single ear headsets....”) And Victorian discloses implementing acoustic devices both as a single ear-level device or a pair of ear-level devices (Victorian, [0063]), further confirming a person of ordinary skill in the art would have found it straightforward to implement devices using a pair of earpieces using a single earpiece. A person of ordinary skill in the art would have found it straightforward to implement Rast's audio device with a microphone, ear canal receiver and processor in a single earpiece, as Victorian teaches all of these features are included in a single earpiece. *See* Victorian, Fig. 4A. And again, Rast describes its invention as applicable to a single ear device. Rast, [0049]. Thus, a person of ordinary skill in the art would have found it routine, straightforward and advantageous to apply Victorian's teachings of a one-sided ear device to Rast's teachings of an acoustic device, and would have known such a combination (yielding

the claimed limitations) would predictably work and provide the expected functionality.

193. Note the distinction between claims 2-5, which recite “where the acoustic device is an earpiece,” and claims 6, 11, 17-18 and 21, which recite method or device comprising an “earpiece.” Regardless, to the extent further disclosure is required for “earpiece” in claims 6, 11, 17-18 and 21, it would have been obvious to apply Victorian’s teachings of an earpiece for those limitations for the same reason articulated above.

C. GROUND 3: RAST IN VIEW OF GREEN (Claim 12)

194. **Claim [12] recites “The method of claim 6, further comprising: detecting and reporting from the sound signature a direction or a speed of a sound source generating the target sound.”** See my discussion of the elements of claim 6 above (§IX.A). The added limitation of claim [12] is met by Rast in view of Green.

195. Green further discloses detecting (e.g., “detect[ing]”) and reporting (e.g., “communicat[ing]”) from the sound signature (e.g., “acoustic signatures”) a direction of a sound source generating the target sound (e.g., “directional information about the origin of an acoustic event” such as “vehicle movement or artillery fire”), as recited in Claim 12. *See, e.g.,* Green:

- [0023] (“The sensor module may also comprise one or more ***orientation***/attitude ***monitors to determine the orientation of the sensor***”)

- module.** The orientation of the sensor module is useful information that the processor can use to determine the data reference for that particular node in the network. Preferably a single orientation sensor is used and is capable of determining magnetic north, so that ***the directionality of the sensors can be determined***, and also the orientation of the module with respect to vertical.”)
- [0024] (“In one embodiment the at least one sensor comprises an acoustic sensor such as a microphone. Acoustic sensing can reveal very useful information about an environment such as a battlefield. For instance the sound of vehicle movement can give information about enemy deployment. Different events such as vehicle movement or artillery fire can give very distinctive acoustic signatures which can be detected and processed by the sensor modules. Preferably the sensor module has a plurality of acoustic sensors which, when the sensor is deployed, form a spaced array. Using an array of sensors information regarding the direction of origin of an acoustic wavefront can be gathered as the acoustic wave will pass the different sensors at different times. The processor can use this information to determine the location of an event. The sensor modules themselves of course form a spaced array and so can be used to give directional information about the origin of an acoustic event.”)
 - [0034] (“The orientation sensor includes a magnetic sensor for locating magnetic north, in effect a compass, ***so that the direction to event can be established***. The orientation sensor also determines orientation with respect to the vertical.”)
 - [0045] (“The microphones 8 are arranged in a spaced array so as to ***detect acoustic signals coming from any direction***. Further from the time interval in the signal reaching different microphones ***the direction of the source of the signal can be detected***. When this information is combined with similar reports from other modules in the network ***the item causing the sound can be located***. The acoustic signature can be analysed looking at frequency, waveform, duration, intensity etc. to try to classify the event. For instance the sound of an artillery unit firing a shell will be relatively short and loud whereas movement of a nearby tracked vehicle will tend to be more prolonged, quieter and rhythmic.”)
 - [0046] (“The signal processing may be performed by the module that ***detects*** the event to ***determine*** the type of event and its ***direction***.”)

196. Green discloses “sensor modules” that “detect[]” events, such as “vehicle movement or artillery fire” based on their “acoustic signatures.” Green, [0024]. From these detected sounds, the sensor module determines the direction of an acoustic event, or “direction of the source of the signal can be detected.” Green, [0024], [0045]. The information “communicated to the module acting in control mode” by the sensor module includes “location information,” including the “direction.” Green, [0046].

197. Green describes an “acoustic sensor[y]” apparatus used to determine the location of an event. Green, [0024]. “[S]ensory modules” with acoustic sensors detect and process the acoustic signatures of sound signals. Green, [0024]. The sensors provide “information regarding the direction of origin of an acoustic wavefront” which is “gathered as the acoustic wave will pass the different sensors at different times.” Green, [0024]. “The processor can use this information to determine the location of an event.” Green, [0024].

198. Rast and Green are analogous art. Rast and Green are in the same field as the ’044 patent, which includes acoustic sensory devices. And Rast and Green are reasonably pertinent to alleged problems addressed by the ’044 of “enhancing” the user’s awareness to their environment. *E.g.*, ’044, 1:40-42; Rast, [0023] (“[T]he wearer is made aware of the event”); Green, [0002] (“[D]esire to be able to identify and locate certain events in an environment.”). For example, both Green and Rast

describe applications of their inventions on the battlefield, which is pertinent to the '044 patent which describes setting a target sound of a “gunshot.” '044, 4:10-16, 11:47-51. Green describes “locat[ing] events on a battlefield,” while Rast describes use of its invention “by infantrymen on the battlefield.” Green, [0002], Rast, [0061].

199. Further, a person of ordinary skill in the art would have been motivated to apply Green’s teachings of acoustic location sensory modules to Rast’s acoustic device because, as Green describes, the ability to “identify and locate certain events in an environment” is beneficial, for example, to reconnaissance teams. Green, [0002]-[0003]. By identifying the location of an event, “risk” to a reconnaissance team may be decreased. Green, [0003]. For example, identification of the location of a threatening acoustic event, like artillery fire, would be beneficial to a team on the battlefield in an effort to decrease casualties. A person of ordinary skill in the art would have found it straightforward to apply Green’s teachings of identifying direction of a sound source to Rast’s acoustic device, as this was an application that was well known at the time, as evidenced by textbook descriptions. *See, e.g.*, Ex, 1012 (Spoken language Processing, Huang, 2001 (“SLP”)), Page 501 (explain how to determine direction of arrival with microphones). Further, Rast highlights the importance of such directional information of sound signals through its hearthrough mode. Rast, [0040] (“[P]roviding directional sound, similar to that of unaided ears, while in hearthrough mode.”); *see also* Rast, [0050] (describing the “user can discern

directional characteristics of sound when the headsets are configured to listen to the external environment.”) As such, a person of ordinary skill in the art would have been motivated to further enhance directional sound information gathering using Green’s invention.

200. Thus, a person of ordinary skill in the art would have found it routine, straightforward and advantageous to apply Green’s teachings of direction sensing acoustic modules to Rast’s teachings of an acoustic device, and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

D. GROUND 4: RAST IN VIEW OF BETTS (Claim 14)

201. See §IX.A.2 discussing claim 6.

202. **Claim [14] recites “The method of claim 6, further comprising: identifying the target sound from the sound signature and transmitting a warning notification to other devices..”** See my discussion of the elements of claim 6 above (§IX.A). The added limitation of claim [14] is met by Rast in view of Betts.

203. Claim 14 further requires “identifying the target sound from the sound signature and transmitting a warning notification to other devices.” As explained above in §IX.A, Rast discloses and at least renders obvious this limitation. To the extent it is argued additional disclosure of notification to multiple other devices is

required, Betts discloses transmitting a warning notification to other devices (e.g., “generating a notification signal in the one or more PEDs indicating detection of the event”). See, e.g., Betts:

- [Abstract] (“Disclosed are personal electronic devices (PEDs) having a sensory enhancement (SE) system for **monitoring environmental conditions** and **detecting environmental events**, for example but not limited to, changes in acoustic, thermal, optical, electromagnetic, chemical, dynamic, wireless, atmospheric, or biometric conditions. **The detection of such events can be used to invoke a notification, an alert, a corrective action**, or some other action, depending upon the implementation to the PED user or another party.”)
- [0008] (“Such an embodiment of the distributed system, among others that are described herein, can be summarized as follows: a plurality of PEDs; means for communicating among the plurality of PEDs a selection of a reference signature corresponding to an event to be detected; means for permitting one or more of the PEDs to measure a characteristic of an environment with a transducer associated therewith; means for detecting the event in one or more of the PEDs; and means for generating a notification signal in the one or more PEDs indicating detection of the event. Furthermore, although not necessary for implementation, in the preferred embodiment, the PEDs further include a means for permitting the users to define whether or not their respective PEDs will cooperate and exchange information with others.”)
- [0021] (“The system 100 further comprises a detection engine 215 that stores the one or more reference signatures that are used to identify environmental events, that correlates sensed environmental signals with the reference signatures, and that detects occurrences of the environmental events. The detection engine 215 can be implemented in hardware, software, or a combination thereof, but is preferably implemented in software executed by a computer based architecture. When designed via software, it can be stored and transported in a computer readable medium.”)
- [0035] (“In some embodiments of the SE system 100, the SE system 100 may be designed to **communicate a notification to a remote**

communications device in order to advise a remote party of detection of an event. Examples include a text message, an email, a voice message, etc.”)

- [0044] (“The acoustic SE system 100 recognizes a much broader set of signals beyond the speech recognized as dial commands. The acoustic SE system 100 stores additional reference signals for recognition. These additional reference signals can be recorded directly by the SE system 100. Or, preferentially, these signals can be obtained as files downloaded from a central repository. Examples include a set of bird songs or a set of registered emergency signals.”)
- [0103] (“Beneficially, all of the cooperating PEDs can be notified of the detected event by wireless signals communicated from the detecting PED or PEDs.”)

204. Betts discloses communicating and generating a notification signal to a plurality of personal electronic devices (“PEDS”). Betts, [0008]; *see also* [Abstract], [0021], [0035], [0044], [0103]. Betts describes a “sensory enhancement (SE) system” for detecting environmental events and “generating a notification signal in [] one or more [] PEDS indicating detection of the event.” Betts, Abstract, [0008]. Betts describes a “means for communicating among the plurality of PEDs” that an event in the environment has been detected, and notifying the PEDs. Betts, [0008].

205. Rast and Betts are analogous art. Rast and Betts are in the same field as the ’044 patent, which includes environmental sensor devices. And Rast and Betts are reasonably pertinent to alleged problems addressed by the ’044 of “enhancing” the user’s awareness to their environment. *E.g.*, ’044, 1:40-42; Rast, [0023]; Betts, [0001] (“[S]ensory enhancement (SE) systems...for monitoring environmental

conditions and detecting environmental events, for...changes in acoustic...conditions.”). For example, Betts describes using its system for “sensory enhancement to the user,” which is pertinent to the problem addressed by the ’044 and Rast of enhancing the user’s awareness of their environment. ’044, 1:45-47; Rast, [0012], Betts, [0003].

206. A person of ordinary skill in the art would have been motivated to apply Betts’s teachings of generating and sending a notification to a plurality of personal electronic devices at the detection of an acoustic event to Rast’s acoustic device to further advance awareness of the environment to not only the user, but to others who may be affected by or affect the user’s environmental conditions. For example, Rast acknowledges the benefit of generating and sending a notification to another device, by, for example, signaling an external event to a squad leader. Rast, [0061]. Betts describes expanding the notification functionality by providing that notification to a plurality of PEDs, in order “to provide beneficial sensory enhancement” to the users through such notifications. Betts, [0003], [0008]. A notification to several devices or users will notify more individuals, and would be beneficial in scenarios as described by Rast, like a battlefield, where there are many infantrymen. *See, e.g.,* Rast, [0061].

207. Thus, a person of ordinary skill in the art would have found it routine, straightforward and advantageous to apply Betts’s teachings of notifications to a

plurality of devices to Rast's teachings of an acoustic device, and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

E. GROUND 5: RAST IN VIEW OF GIESBRECHT (Claim 19)

208. **Claim [19] recites “The method of claim 18, further comprising: enhancing auditory queues in the target sound relative to the audio content based on a spectrum of the ambient sound captured at an ambient sounds microphone (ASM).”** See my discussion of the elements of claim 18 above (§IX.A). The added limitation of claim [19] is met by Rast in view of Giesbrecht.

209. As I described in §IX.A, Rast discloses or at least renders obvious the method of claim 18. Rast further discloses enhancing auditory queues (e.g., “voices, [or] troop movements”) in the target sound relative to the audio content (e.g., “amplifying the signal from the microphone” and “any incoming audio source, such as music over a wired or wireless connection, is attenuated or disconnected”) based on the ambient sound (e.g., based on “sounds from the external environment”) captured at an ambient sounds microphone (ASM) (e.g., from “external microphones,” *see also* [1.a], [11]). *See, e.g.*, Rast:

- [0013] (“Typically, headsets (or headphones) are utilized for modifying the personal acoustic environment for a single wearer, such as in the case of listening to an audio program source, maintaining communication (such as between pilot and copilot), blocking external sounds, **enhancing external audio**, and so forth. It will be appreciated, therefore, that the apparatus referred to as a headset may be configured in various

mechanical configurations, such as mounted as earpieces within a helmet, single sided headset, earbuds, and so forth.”)

- [0014] (“**Sounds from the external environment are preferably registered by a microphones**, with one microphone attached, or integrated, into the exterior of each earpiece... The mode of the headset may be changed by any of a number of mechanisms, for example, by...accentuating spoken sounds (drop noise floor while amplifying speech highlights and formants)...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.”)
- [0022] (“For example, one preferred method delivers environmental audio to the wearer during a predetermined interval by **amplifying the signal from the microphone and playing it over the speakers within the earpieces**. Preferably, **any incoming audio source, such as music over a wired or wireless connection, is attenuated** or disconnected **during the period of time the environmental audio is being played**. The specific sounds to be recognized (correlated) with the external environment may be contained within the headset as a prerecorded sound, or as a sound that is recorded by the user from the target environment.”)
- [0059] (“The relative volume to which hearthrough is to be configured is preferable set by the user.”)
- [0061] (“The sound correlation unit being preferably configured for controlled attenuation, selective blocking, of artillery and gunfire to a predetermined decibel level so that hearing is not dulled or damaged, while **sounds such as voices, troop movements and so forth are amplified to provide a hearing advantage to the solder.**”)

210. Rast discloses “amplifying” sound characteristic such as “voices” or “troop movements,” and “attenuat[ing]” the incoming audio sound from the audio source based on the environmental sound captured by the external microphones. Rast, [0061].

211. Giesbrecht discloses enhancing auditory queues in the target sound relative to the audio content based on a spectrum of the ambient sound (e.g., “enhances the quality and intelligibility of a received speech signal by adjusting one or both of the average speech spectral shape of a received speech signal and the background noise spectral shape of the received signal”). See, e.g., Giesbrecht:

- [0005] (“A system and method for adaptively enhancing speech signals are provided. The system and method of the invention affectively ***normalize the spectrum of an input speech signal toward a target spectral shape***, or ideal frequency response. ***The target spectral shape may be selected based on the application for which the speech signal is intended***. For example, a desired spectral shape for a speech signal destined to be transmitted via a handsfree telephone in an automobile may be significantly different from the desired spectral shape of a speech signal which is to be input into a speech recognition system.”)
- [0010] (“Another embodiment enhances the frequency response of a speech signal by adjusting the spectral shape of the background noise of a received speech signal. This embodiment includes performing a frequency sub-band analysis on successive overlapping windowed buffers of a speech signal. A background noise estimate is generated based on the received signal. Next, a background noise spectral shape correction factor is calculated by subtracting the background noise estimate from a target background noise spectral shape. The background noise spectral shape correction factor is then added to a spectrum corresponding to one of the successive overlapping windowed buffers.”)
- [0011] (“Yet another embodiment enhances the quality and intelligibility of a received speech signal by adjusting one or both of the average speech spectral shape of a received speech signal and the background noise spectral shape of the received signal.”)
- [0061] (“The difference between background noise estimate 322 and the target background noise spectral shape represents the amount by which the background noise estimate must be adjusted in order to conform to the shape of the target background noise spectral shape. Like the

determined speech spectral shape correction factor 314, the determine background noise spectral correction factor 326 calculates a background noise spectral correction factor by subtracting the target speech spectral shape from the background noise estimate across all frequency bins of the compressed dB spectrum of the input signal. Also like the speech spectral shape correction factor, the background noise spectral shape correction factor may be added directly to the compressed dB spectrum of the input speech signal 302 in order to shape the frequency spectrum of the background noise included in the input speech signal 302.

However, in the embodiment depicted in the flow chart 300, both the speech spectral shape correction factor and the background noise spectral shape correction factor contribute to a final spectral shape correction factor. The final spectral shape correction factor is then added to the compressed db spectrum of the input speech signal 302.”)

- Fig. 12 (Flow Chart 300)

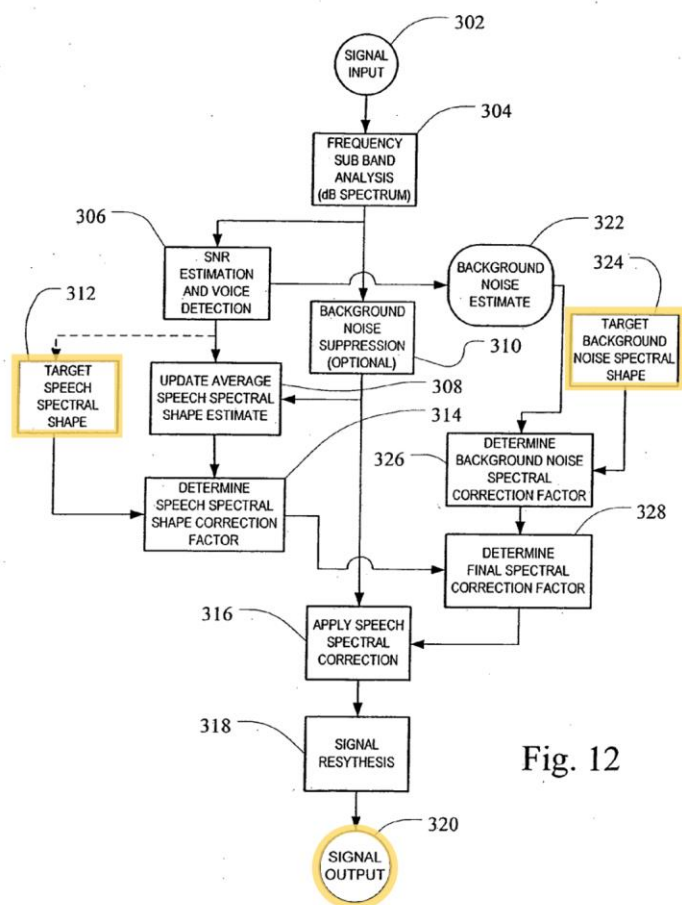


Fig. 12

212. Giesbrecht discloses enhancing speech signals based on their spectrum. Giesbrecht, [0011], [0061]. Giesbrecht effectively normalizes the spectrum of a speech signal to resemble the selected “target spectral shape” that represents the ideal frequency for the speech signal, based on the user’s choice of what type of speech signal to enhance. Giesbrecht, [0005], [0061].

213. A “spectral correction factor” is the difference between the spectrum of received speech signal and the desired signal. Giesbrecht, [0007]. Using the spectral correction factor, the spectrum of input speech signal is adjusted towards the desired spectral shape. Giesbrecht, [0061]. This results in the enhancement of certain speech signals relative to the spectrum of the received signal. Giesbrecht, [0005]-[0012]. The spectrum of background noise can, in addition or in the alternative to the speech signal, be normalized to a target spectral shape. Giesbrecht, [0061].

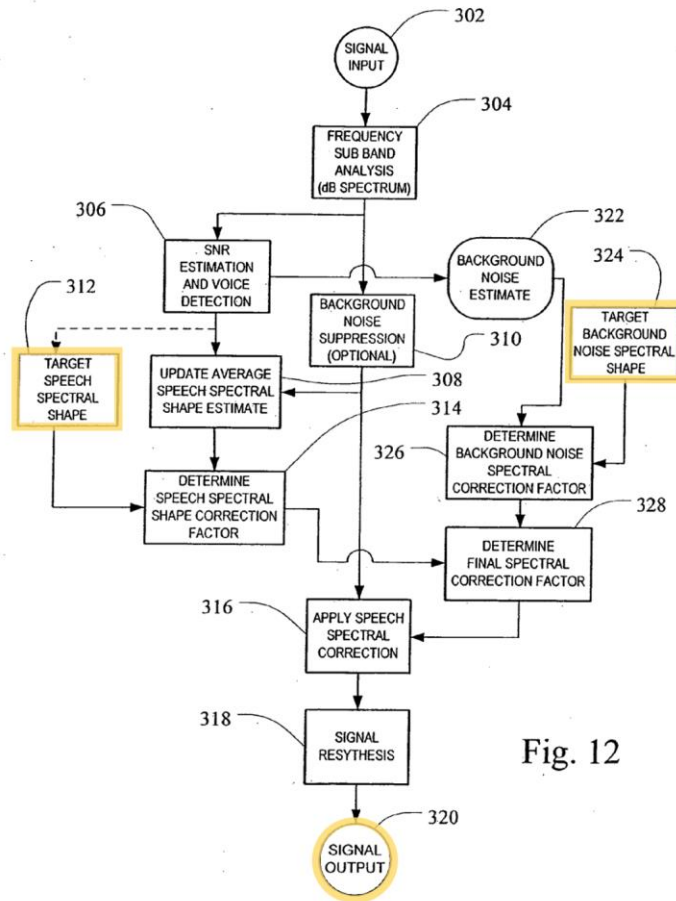


Fig. 12

Giesbrecht, Fig. 12

214. Rast and Giesbrecht are analogous art. Rast and Giesbrecht are in the same field as the '044 patent, which includes acoustic devices for the enhancement of user interaction with the environment. And Rast and Giesbrecht are reasonably pertinent to alleged problems addressed by the '044 of users in “compromising situation[s] since they may not be aware of...their environment.” E.g., '044, 1:40-42; Rast, [0012] (“[F]urther enhance user cognition of the external environment”); Giesbrecht, [0001], [0005]. For example, Giesbrecht describes “improv[ing] both sound quality and intelligibility” to enhance user cognition of their environment,

which is reasonably pertinent to the problem addressed by the '044 because the '044 also seeks to improve user awareness of environmental signals. '044, 1:45-47 (“A need therefore exists for enhancing the user's ability to hear target sounds in their environment without compromising his hearing”); Giesbrecht, [0002] (“Adjusting the spectral shape of a received speech signal can significantly improve the quality of the speech signal...”).

215. A person of ordinary skill in the art would have been motivated to apply Giesbrecht's teachings of enhancing particular speech signals relative to the spectrum of sound to Rast's acoustic device to further enhance the amplification of certain user chosen sound characteristics. As Giesbrecht describes, “adjusting the spectral shape of a received speech signal can significantly improve the quality of the speech signal” (Giesbrecht, [0002]), and thus a person of ordinary skill in the art would have been motivated to improve the quality of the signal in Rast's device, using Giesbrecht's spectral enhancements. Rast's invention goal is to improve the auditory experience of the user, and Giesbrecht provides an additional avenue towards this goal. A person of ordinary skill in the art would have understood that, when Giesbrecht's teachings of speech enhancement are applied to Rast's device, (1) the speech signal is obtained from the ambient sound (see claim [13]), and (2) the detected speech would be enhanced relative to the audio from audio source

because no corresponding enhancement is also made to the audio from the audio source (i.e., only the speech signal is enhanced).

216. Thus, a person of ordinary skill in the art would have found it routine, straightforward and advantageous to apply Giesbrecht’s teachings of speech signal enhancement relative to a spectrum to Rast’s teachings of amplifying sound characterizations, and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

F. GROUND 6: RAST IN VIEW OF VICTORIAN AND NAKAZAWA (Claims 20, 21)

1. Overview of Victorian

217. See my discussion in §IX.B. Further to that, Victorian’s ear-level device includes “**ear canal microphone 440**” that “detects sound from the ear canal,” and “[e]ar canal speaker 470” that plays sound to the ear canal. Victorian, [0030]; *see* Fig. 4A. Victorian’s ear-level device also includes processor 460, including “a speech recognition module.” Victorian, [0032]. Speech recognition is performed on audio from **ear canal microphone 440**. Victorian, [0052]-[0054]; *see* Fig. 9 (steps 910-918).

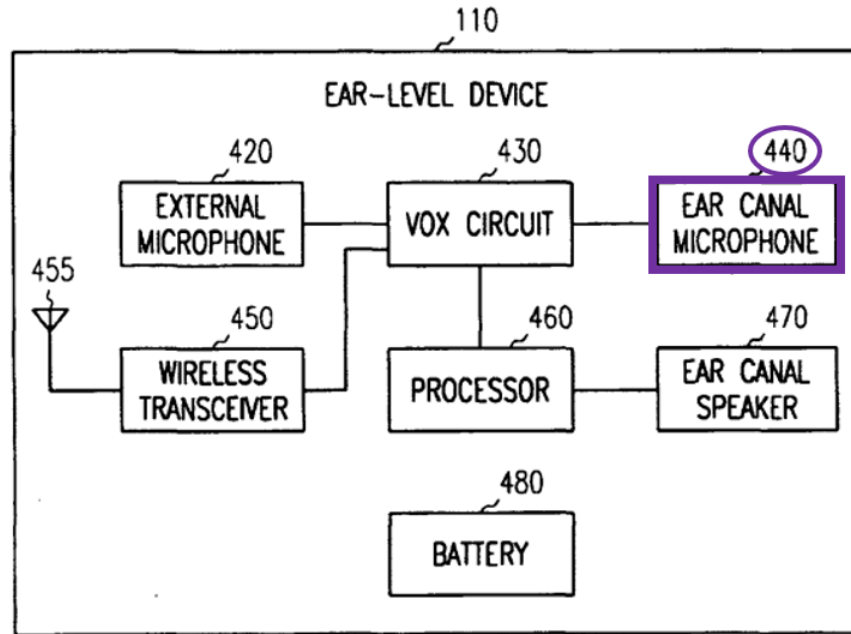


FIG. 4A

Victorian, Fig. 4A

2. Overview of Nakazawa

218. **Nakazawa** describes a wearable microphone-speaker device. Nakazawa, 1:7-12, 1:22-28. The microphone-speaker “output[s]...an audible sound” once a signal is received to do so. Nakazawa, 4:10-14. Nakazawa describes a single circuit element that operates as a speaker “for reception” when the press-to-talk switch 12 is left open, and operates as a microphone “for transmission,” where the press-to-talk switch 12 is closed. Nakazawa, 3:59-61, 4:15-21.

3. Motivation to Apply Victorian’s and Nakazawa’s Teachings to Rast

219. Rast, Victorian, and Nakazawa are analogous art. Rast, Victorian, and Nakazawa are in the same field as the ’044 patent, which includes acoustic devices.

And Rast, Victorian, and Nakazawa are reasonably pertinent to alleged problems addressed by the '044 of “facilitat[ing] portable applications” of acoustic devices. E.g., '044, 4:48-52; Rast, [0061] (“The unit can be...worn”); Victorian, [0006] (describing the problems of “[w]ired” devices as “cumbersome to users who [are] physically active.”); Nakazawa, 1:22-28 (“[C]arried on the person...”); see §IX.B.

220. A person of ordinary skill in the art would have been motivated to apply Victorian’s teachings of an ear canal microphone to Rast’s acoustic device because, as Victorian describes, “there is a need for an ear-level communication system that is not easily visible,” and an ear canal microphone is discrete. Victorian, [0007]. As Victorian describes, “minimal visibility is generally desirable” for “appearance.” Victorian, [0006]. Thus, Victorian’s teachings provide the benefit of a minimally visible and discrete device. Further, Victorian describes the benefit of an ear canal microphone in an acoustic device like Rast, for the purposes of “detect[ing] sound from the ear canal[,]...primarily speech of the person wearing ear-level device 110” for “communication,” which is an application of Rast’s device. Victorian, [0004], [0030]; Rast [0025].

221. A person of ordinary skill in the art would have been motivated to apply Nakazawa’s teachings of a microphone-speaker that plays audible sound to Rast’s acoustic device (as modified by Victorian) because Nakazawa would further advance both Rast and Victorian’s goals of being small, portable and wearable by

reducing necessary circuitry and electronics by not requiring a separate microphone and speaker. E.g., Rast, [0061]; Armstrong, [0005]; Nakazawa, 1:22-28; '044, 4:48-52. Enabling an audio device to have less components generally reduces the size of the device. Thus, instead of having a separate microphone and separate speaker, having one microphone-speaker would reduce the device components and size. A person of ordinary skill in the art would have understood that, when applying Victorian's teachings of an ear canal microphone to Rast, integrating Rast's speaker with the ear canal microphone would have reduced the size of the earpiece because there would be one microphone-speaker instead of a separate microphone and separate speaker. *See, e.g.*, Nakazawa, 1:18-22 (describing the device as "small-sized"). Further, integrating Rast's speaker with the ear canal microphone would reduce the size of the earpiece because a single transducer for both the detection and playing of audio would be used. Nakazawa, Fig. 4. Figure 4 depicts a single circuit element that is operating as a speaker when in receive mode and a microphone when in transmit mode. Nakazawa, 3:56-4:49 (describing a single circuit device).

222. Thus, a person of ordinary skill in the art would have found it routine, straightforward and advantageous to apply Victorian's teachings of an ear canal microphone to Rast's teachings of an acoustic device for noise cancellation, and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality. Further, a person of

ordinary skill in the art would have found it routine, straightforward and advantageous to apply Nakazawa’s teachings of a microphone-speaker to Rast’s teachings of an acoustic device, as modified by Victorian’s ear canal microphone, and would have known such a combination (yielding the claimed limitations) would predictably work and provide the expected functionality.

4. Invalidity of Claims 20 and 21 Over Rast in view of Victorian and Nakazawa

223. As I discuss below, Claims 20 and 21 are rendered obvious by Rast in view of Victorian and Nakazawa. In addition to my discussion in Sections I, IV, and V, the following discussion identifies example disclosures followed by bulleted excerpts that each disclose Claims 20 and 21.

a. Element [20.pre]

224. **Element [20.pre] recites “A sound detection device comprising.”**
See my discussion of element [16.pre] in §IX.A. This element is met by Rast.

225. To the extent the preamble is limiting, Rast discloses a sound detection device (*e.g.*, “method of correlating the external sounds and characteristics with programmed or recorded sounds or characteristics”).

226. As I discussed in element [16.pre], Rast’s device detects “programmed...sounds” (sound signatures) from “external sounds.”

b. Element [20.a]

227. Element [20.a] recites “an ambient sound microphone configured to measure an ambient sound.” See my discussion of element [1.a] in §IX.A.2. This element is met by Rast.

228. Rast discloses an ambient sound microphone (see [1.a]) configured to measure an ambient sound (e.g., “sounds from the external environment ... registered by the microphones” “compromis[ing] any combination of sound representations”)

- [0014]: “***Sounds from the external environment are preferably registered by a microphones***, with one microphone attached, or integrated, into the exterior of each earpiece. Each microphone converts the received external sound energy to an external sound signal.”
- [0016]: “The apparatus of the present invention is capable of retaining information about acoustic events (stored sounds) to which external acoustic events (external sounds) may be correlated. For simplicity, this retained information is often referred to as “stored sounds”, or “sound characterizations”, however, it is to be recognized that these “***stored sounds***” ***may comprise any combination of sound representations, sound characterizations, sound related algorithms, sound correlation programming, sound scripting, sound signaturing, along with additional methods of storing information about audio waveforms.***”
- [0056] (“It will be appreciated that audio headphones are often equipped with a volume control to allow the wearer to set the audio level played over the earpieces. The headset may additionally contain a ***sensitivity control that allows the user to set the sound level required in order to trigger the correlated response.*** In the example of the word “Hello!” the ***sensitivity adjustment would select how loud***, or alternatively how accurate, ***the sound would need to be in order to trigger the correlation response.***”)

229. As I discussed in element [1.a], Rast’s “[m]icrophones 58a, 58b...convert sounds within the external acoustical environment to electrical signals.” In order to convert the sound to an electrical signal, the microphone measures the sound, and outputs a corresponding electrical signal. Rast, [0053]. Rast’s microphones register the sounds from the external environment as “any combination of sound representations,” including several “methods of storing information about the audio waveforms.” Rast, [0016]. As such, Rast’s microphones measure the ambient sound.

230. Further, Rast discloses that the user may “set the sound level required in order to trigger the correlated response,” such that the device must measure the ambient sound in order to make the determination as to whether a match has occurred, and thus trigger the programmed response. Rast, [0056].

c. Element [20.b]

231. **Element [20.b] recites “an ear canal microphone.”** This element is met by Victorian.

232. Victorian discloses an ear canal microphone (*e.g.*, “ear canal microphone 440”).

- Fig. 4A

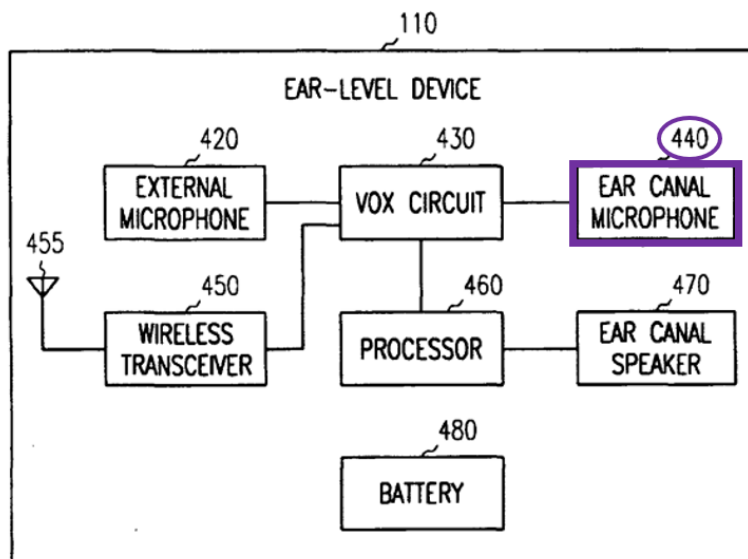


FIG. 4A

- [0030] (“Ear-level device 110 includes an external microphone 420, *an ear canal microphone 440.... Ear canal microphone 440 detects sound from the ear canal.* The sound includes primarily speech of the person wearing ear-level device 110...In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110.”)
- [0031] (“[S]ound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440...Because *ear canal microphone 440 is capable of detecting the sound delivered to the ear canal* from ear canal speaker 470...”)

233. Victorian discloses an “ear canal microphone 440 [that] detects sound from the ear canal.” Victorian, [0030]. As discussed in §IX.F, a person of ordinary skill in the art would have been motivated to apply Victorian’s teaching of an ear canal microphone to Rast’s audio device.

234. Further, Rast discloses that the user may “set the sound level required in order to trigger the correlated response,” such that the device must measure the ambient sound in order to make the determination as to whether a match has occurred, and thus trigger the programmed response. Rast, [0056].

d. Element [20.c]

235. **Element [20.c] recites “a processor configured to compare the ambient sound to at least one target sound signature, and where the processor identifies an onset of an identified target sound signature in the ambient sound.”** This element is met by Rast.

236. Rast discloses a processor (*see* [1.c.1] in §IX.A.2) configured to compare the ambient sound to at least one target sound signature (*see* [1.c.2] in §IX.A.2), and where the processor identifies an onset of an identified target sound signature in the ambient sound (*e.g.*, “[s]egments of sounds from the external environment are continually compared with the sounds and sound characteristics programmed in the sound correlation unit” and a “determination is made at block 116 as to whether the sound detected in the environment matches the correlation sound, or characteristics”).

- [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”)...”)

- [0051] (“Segments of sounds from the external environment are ***continually*** compared with the sounds and sound characteristics programmed in the sound correlation unit of the headset. To speed discernment, a small segment of audio is compared with all the stored trigger conditions, and ***upon meeting a first phase of correlation***, the discernment process continues to check further portions of the sound against the stored sounds and characteristics. ***If the external sound correlates with the programmed sound at block 46, then the mode of operation of the headphone is changed at block 48 to that which has been programmed by the user.***”)

237. See my discussion of claim [1] in §IX.A.2.

238. As I described in element [1.c.2], Rast discloses a DSP that “continually” compares sounds from the external environment to stored sound characteristics and identifies when a match is made. Rast, [0051]. At the first instance of a match, the mode of the headset is changed per user programming. Rast, [0051]. A person of ordinary skill in the art would have understood, and at minimum found obvious, that this is the “onset” of a sound match because the device supports a “hearthrough mode” in which the device passes through audio until the sound of interest ends. Rast, [0014]. In this hearthrough mode, the processor performs the “processor main loop” continually, including the comparison to the target sound and identification of a match, and continues the “hearthrough” operation until the sound of interest stops, and there is no longer a target sound match. See, e.g., Rast [0014]. Thus, the processor identifies the “onset” of the target sound in the ambient sound, as it must be continually monitoring the ambient sound in order to carry out hearthrough mode.

e. Element [20.d]

239. Element [20.d] recites “where the ear canal microphone is configured to emit an auditory warning when the processor identifies the onset.” This element is met by Rast, in view of Victorian and Nakazawa.

240. Rast discloses a speaker is configured to emit an auditory warning (see [3], [13], [14] in §IX.A.2) when the processor identifies the onset (see [20.c]; see also [1.c.2], [6.b], [18.d] in §IX.A.2).

- [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.*”)
- [0022] (“For example, one preferred method delivers environmental audio to the wearer during a predetermined interval by *amplifying the signal from the microphone and playing it over the speakers within the earpieces.*”)

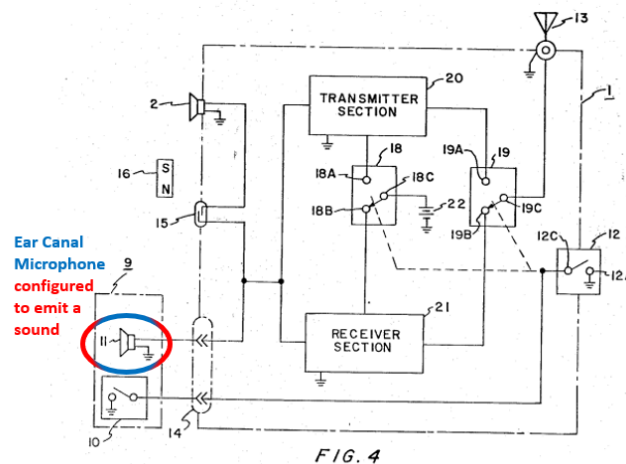
241. See my discussion of element [1.c.2] and claim [10] in §IX.A.2.

242. Rast further discloses its speaker provides alerts and warning tones when microphone signal matches the correlation sound, and these alert tones are played together with the hearthrough mode discussed in [20.c]. Rast, [0014], [0017], [0051]. As explained in [20.c], the match is determined at the onset of an identified target sound. Rast, [0051]; see also [1.c.2], [6.b], [18.d].

243. Victorian discloses an ear canal microphone (e.g., “ear canal microphone 440”). See my discussion of element [20.b] in §IX.A.2.

244. Nakazawa discloses a microphone configured to emit sound (e.g., “microphone-speaker” configured to “output...an audible sound”). See, e.g., Nakazawa:

- Fig. 4



- 1:47-53 (“[I]ts **internal microphone-speaker** can be efficiently connected and disconnected, as desired...”)
- 2:33-46 (“[T]he portable radio unit which includes an **internal microphone-speaker** 2 mounted inside of the grilled front face 3 of the radio unit and a nonlock type pushbutton switch 4 operative to connect and disconnect the internal microphone-speaker 2...Reference numeral 9 indicates a handset including an **external microphone-speaker 11**...”)
- 3:9-15 (“Such arrangement is particularly advantageous in that it enables **communication through the internal microphone-speaker 2** of the radio unit with no external microphone-speaker connected thereto even in cases where the radio unit 1 is carried on a person as at his waist with the aid of the carrying holder 5.”)
- 4:10-15 (“In the circuit state illustrated, when the radio signal is received at the antenna 13, it is demodulated at the receiver section 21 and is

outputted through the internal microphone-speaker 2 as an audible sound.”)

- 4:20-25 (“*[T]he audible sound signal inputted by way of the internal microphone-speaker 2* is modulated at the transmitter section 20...”)
- 4:28-59 (“Further, when the handset 9 is connected to the radio unit 1 by coupling plug 14, the *external microphone-speaker 11* is connected to the same junction as the internal microphone-speaker 2, that is, to the junction between the voice-signal output end of receiver section 21 and the voice-signal input end of transmitter section 20. At the same time, the external press-to-talk switch 10 on the handset 9 is connected, like the press-to-talk switch 12 on the radio unit 1, to the control terminal for controlling the switch circuits 18 and 19.”)

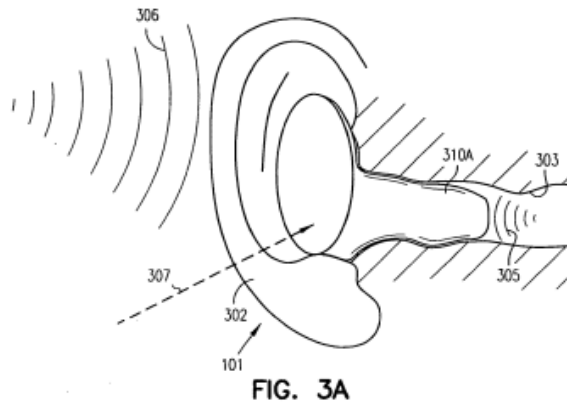
245. Nakazawa discloses a microphone-speaker that outputs an audible sound. Nakazawa, 4:10-15. As discussed in §IX.F.3, a person of ordinary skill in the art would have been motivated to apply Nakazawa’s teachings of a “microphone speaker” to Rast’s device (to which Victorian’s teachings of an ear canal microphone have been applied).

f. Claim [21]

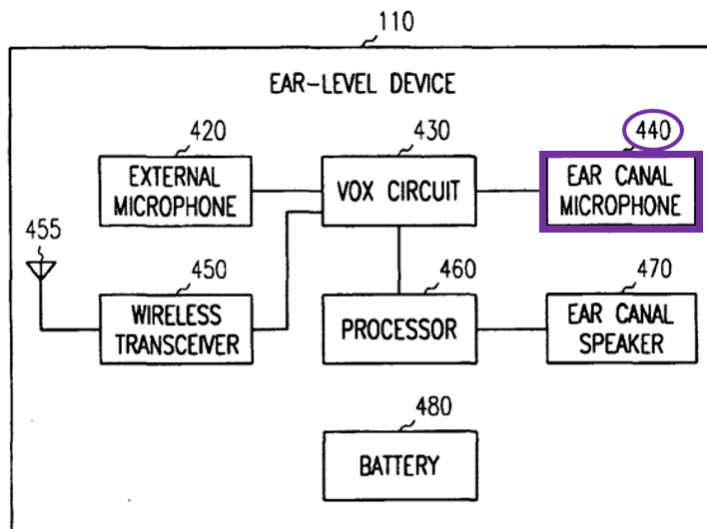
246. **Claim [21] recites “The sound detection device according to claim 20, where the ear canal microphone is operatively connected to an earpiece.”** See my discussion of the claim 20 above (§IX.F). The added limitation of claim [21] is met by Rast, in view of Victorian.

247. Victorian discloses the ear canal microphone (see element [20.b]) connected to an earpiece (e.g., “ear canal microphone 440” included in “ear-level device 110,” as seen in Figs. 3A, 3B, 3C, and 4A). See, e.g., Victorian:

- Fig. 3A



- Fig. 4A



- [0024]: “FIG. 3A is an illustration of an exemplary exterior configuration of *an in-the-ear (ITE) device 310A* used as *ear-level device 110 or one of ear-level devices 210A and 210B*. As shown in FIG. 3A, ITE device 310A is attached to an ear 101 with a pinna 302 and an ear canal 303. ITE device 310A includes a portion inserted into ear canal 303 and another portion fit into the cavity formed by pinna 302.”
- Fig. 3A

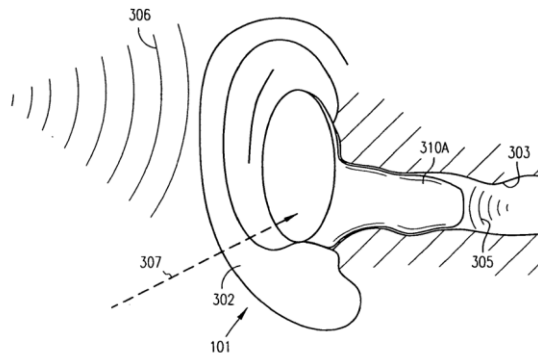


FIG. 3A

- [0025] (“FIG. 3B is an illustration of an exemplary exterior configuration of an *in-the-canal (ITC) device 310B* used as *ear-level device 110* or one of *ear- level devices 210A and 210B.*”)
- Fig. 3B

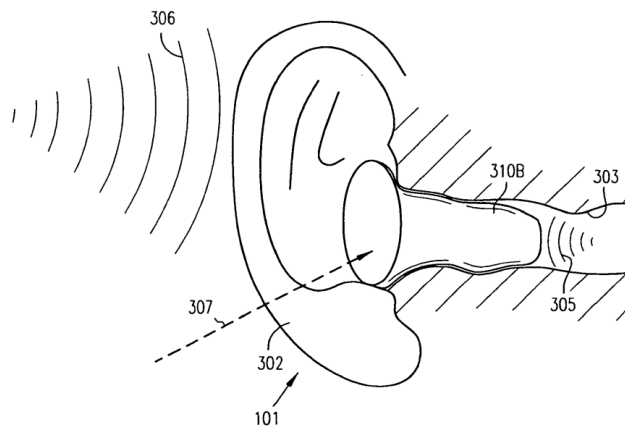


FIG. 3B

- [0026] (“FIG. 3C is an illustration of an exemplary exterior configuration of a *completely-in-the-canal (CIC) device 310C* used as *ear-level device 110* or one of *ear- level devices 210A and 210B.* As shown in FIG. 3C, CIC device 310C is attached to ear 101 with pinna 302 and ear canal 303.”)
- Fig. 3C

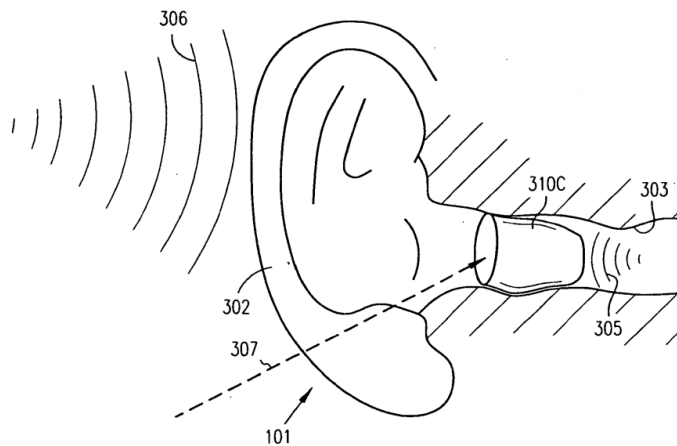


FIG. 3C

- [0029] (“**[E]ar-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device 110.** The circuit is housed in an earmold configured for use as one of the device types illustrated as ear-level devices 310A, 310B, or 310C.”)
- [0030] (“Ear-level device 110 includes...an ear canal microphone 440...”)

248. See my discussion of element [1.c.2] and claim [10] in §IX.A.2.

249. Victorian discloses that the “[e]ar-level device 110 includes...an ear canal microphone 440,” and illustrates that ear level device 110 in the form of an “in-the-ear” device 310A. Victorian, [0024], [0029], [0030]. As discussed in §IX.F, a person of ordinary skill in the art would have been motivated to apply Victorian’s teaching of an ear canal microphone to Rast’s audio device.

X. SECONDARY CONSIDERATIONS

250. I have seen no evidence in the ’044 patent’s prosecution history or elsewhere supporting any secondary considerations arguments, or evidence of nexus to any of the Challenged Claims. See generally ’044 File History. Indeed, per my

discussion of the prior art referenced herein, any purported solutions to problems or unexpected results in the '044 patent were already well known. To the extent Patent Owner asserts the existence of any secondary considerations in its responses, I reserve my right to address any such evidence.

XI. CONCLUSION

251. In summary, I have concluded that the Challenged Claims are rendered obvious over the prior art references as discussed above. Specifically, in my opinion, for at least the reasons above, the Challenged Claims of the '044 patent obvious.

252. To the extent it is argued that any further disclosure is required for an element in the Challenged Claims that I have identified as being disclosed, a person of ordinary skill in the art would certainly have found that element obvious to include based on the sample explicit disclosure or inherent disclosures and analysis I have identified above.

253. I reserve the right to supplement my opinions in the future to respond to any arguments that Patent Owner or its expert(s) may raise and to take into account new information as it becomes available to me.

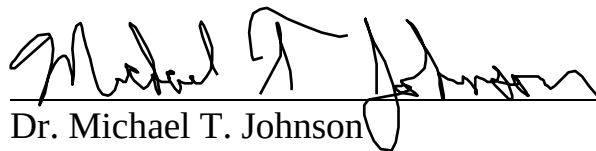
254. In signing this declaration, I recognize that the declaration will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I also recognize that I may be subject to cross-examination in the case and that cross-examination 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.

255. I reserve the right to supplement my opinions in the future to respond to any arguments that Patent Owner or its expert(s) may raise and to take into account new information as it becomes available to me.

256. I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

257. Executed this 1 day of September 2024, in Lexington KY.


Dr. Michael T. Johnson

APPENDIX A

Curriculum Vitae

Michael T. Johnson

Associate Dean for Undergraduate Education and Student Success
Professor, Electrical and Computer Engineering
University of Kentucky College of Engineering
453 F. Paul Anderson Tower, Lexington, KY 40506, Phone (859) 257-0717
Email: mike.johnson@uky.edu
Web page: <http://johnson.engineering.uky.edu/>

CURRICULUM VITAE EXECUTIVE SUMMARY

Publications: **52 journal papers, over 150 refereed publications/presentations**
 Citations ≈3600, h-index 31, i10-index 70 (Google scholar Jan 2024)

Grant funding: **6 major grants, over \$3.5 million in external funding (\$2.7m as PI)**
 Multi-institution, interdisciplinary, international collaborations

Students: **8 PhD graduates, 14 MS graduates**

Leadership: **Associate Dean for Undergrad Education and Student Success, UK COE (2023-)**
 President, Southeast ECE Department Heads Association (2021-22)
 SEC Academic Leadership Development Program, (2019-2020)
 Department Chair, UK ECE (2016-2023)
 Director, UK CoE Scholars in Engineering Leadership (2017-2020)
 Chair of Marquette Board of Graduate Studies (2012-2016)
 Director of Graduate Studies, Marquette EECE (2009-2012)
 Senior Visiting Professor, Tsinghua University (2008-2009, 2014-2015)
 Engineering Manager (1993-1996)

Honors: **Featured in “Groundbreaking Thinkers in Wisconsin” series 2007**
 Marquette COE Researcher of the Year 2006-2007
 Engineers and Scientists of Milwaukee Young Engineer of the Year 2006
 HKN EECE Teacher of the year 2005, 2014

RESEARCH SPECIALIZATION

- Speech Processing: speech recognition and enhancement, natural language processing
- Signal Processing: bioacoustics, microphone array processing, signal enhancement
- Pattern Recognition and Machine Learning: statistical estimation and classification

EDUCATION

Doctor of Philosophy, School of Electrical and Computer Engineering, August 2000
Purdue University, West Lafayette, IN GPA: 4.00/4.00
"Incorporating Prosodic Information and Language Structure into Speech Recognition Systems"
Advisor: Dr. Leah H. Jamieson

Master of Science in Electrical Engineering (concentration Digital Signal Processing), Dec. 1994
University of Texas at San Antonio, San Antonio, TX GPA: 4.00/4.00
"An Adaptive Texture Model Based on Markov Random Fields"
Advisor: Dr. Mita Desai

Bachelor of Science, Engineering with Electrical Concentration, April 1990
LeTourneau University, Longview, TX GPA: 3.60/4.00

Bachelor of Science, Computer Science and Engineering with minor in Mathematics, April 1989
LeTourneau University, Longview, TX GPA: 3.57/4.00

ACADEMIC WORK EXPERIENCE

8/2016 – present University of Kentucky, Lexington, KY

- *Associate Dean for Undergraduate Education, College of Engineering* (2023-present)
- *Department Chair, Electrical and Computer Engineering* (2016-2023)
- *Full Professor, Electrical and Computer Engineering* (2016 - present)

8/2000 – 8/2016 Marquette University, Milwaukee, WI

- *Full Professor, Electrical and Computer Engineering*: (2013 - 2016)
- *Director of Graduate Studies, Electrical and Computer Engineering*: (2009 - 2012)
- *Associate Professor with Tenure, Electrical and Computer Engineering* (2007 – p2009)
- *Assistant Professor, Electrical and Computer Engineering*: (2000 – 2007)

2008-2009 and 2014-2015 Tsinghua University, Beijing, China (two 1-year sabbaticals)

- *Senior Visiting Faculty, Electronic Engineering Dept.* While on sabbatical, taught courses and collaborated on research in speech processing with Prof. Liu Jia.

9/1996 – 8/2000 Purdue University, West Lafayette, IN

- *Graduate Research Assistant, Audiology Dept.*: (5/97 to 5/00) Research focused on studying the consistency of speech articulator movement in children and young adults.
- *Graduate Research Assistant, ECE Dept.*: (1/97 to 1/98) Part of an NSF-funded research grant working on interfacing speech recognition and language processing systems.
- *Teaching Assistant, ECE Dept.*: (9/96 to 12/96 and 1/98 to 5/98) Assisted with a course in Speech Processing and taught a laboratory course in Digital Circuits.

1/1993 - 9/1993 University of Texas at San Antonio, San Antonio, TX

- *Graduate Research Assistant, UTSA Image Processing Lab*: Did research in the field of biological image processing as part of masters thesis in mammogram image analysis.

INDUSTRIAL WORK EXPERIENCE

9/1993 - 7/1996 SNC Manufacturing, Oshkosh, WI

- *Engineering Manager, Telecommunications Division* (11/94 - 7/96): Responsible for all new and existing product lines, including design, project management, and personnel.
- *Senior Engineer* (9/93-11/94): Project lead for new products, primarily in the area of SNC's Lyte Lynx line of fiber-optic interfaces for high-voltage environments.

9/1991 - 1/1993 Datapoint Corporation , San Antonio, TX

- *Engineer II*: Hardware engineer on the ArcnetPlus networking and MINX videoconferencing projects. Lead engineer on the PC HUB project for ArcnetPlus.

5/1990 - 9/1991 Micronyx, Inc. / Micro Technology Services, Inc., Richardson, TX

- *Design Engineer*: Telecommunications and embedded microprocessors design, at both hardware and software levels, for Micronyx and its spin-off company MTSI.

5/1989 - 8/1989 GTE , Ft. Wayne, IN *Engineering Intern*: Designed circuits for Business Services, including a project restructuring an analog phone system into a digital carrier system.

6/1988 - 8/1988 NBS , Ft. Wayne, IN *Electrical Technician*: Gained experience with troubleshooting circuit boards, soldering, and designing and building test equipment.

PUBLICATIONS

Peer-reviewed Journal Publications

1. Rahim Soleymanpour, Mohammad Soleymanpour, Anthony J Brammer, Michael T Johnson, Insoo Kim, Speech Enhancement Algorithm Based on a Convolutional Neural Network Reconstruction of the Temporal Envelope of Speech in Noisy Environments, IEEE Access, Volume 11, 5328-5336, January 2023.
2. Julie N. Oswald, Amy M. Van Cise, Angela Dassow, Taffeta Elliott, Michael T. Johnson, Andrea Ravignani, Jeffrey Podos, A Collection of Best Practices for the Collection and Analysis of Bioacoustic Data, Applied Sciences, Volume 12, No. 23, November 2022.
3. Chao Li; Wang, Qiyue Wang, Wenhua Jiao, Michael T. Johnson, Yuming Zhang, Deep Learning-Based Detection of Penetration from Weld Pool Reflection Images, Volume 99 No. 2, Welding Journal, September 2020.
4. Qiyue Wang, Wenhua Jiao, Rui Yu; Michael T. Johnson, Yuming Zhang, Virtual Reality Robot-Assisted Welding Based on Human Intention Recognition, IEEE Transactions on Automation Science and Engineering, Volume 17 No. 2, April 2020.
5. Yi Liu; Liang He; Jia Liu; Michael T. Johnson, Introducing phonetic information to speaker embedding for speaker verification, EURASIP Journal on Audio, Speech, and Music Processing, December 2019.
6. Qiyue Wang, Yongchao Cheng, Wenhua Jiao, Michael T. Johnson, Yuming Zhang, Virtual reality human-robot collaborative welding: A case study of weaving gas tungsten arc welding, Journal of Manufacturing Processes, Volume 49, 2019.
7. Qiyue Wang, Wenhua Jiao, Rui Yu, Michael Johnson, YuMing Zhang, 2019. Modeling of Human Welder Operation in Virtual Reality Human-Robot Interaction Welding System. IEEE Robotics and Automation Letters, 4(3): 2958-2964, July 2019.
8. Ruijie Yan, Liangrui Peng, Shanyu Xiao, Michael T. Johnson, Shengjin Wang, Dynamic temporal residual network for sequence modeling. International Journal on Document Analysis and Recognition, Volume 22, 235–246, July 2019.
9. Liang He, Xianhong Chen, Can Xu, Yi Liu, Jia Liu, Michael T. Johnson, Latent class model with application to speaker diarization, EURASIP Journal on Audio, Speech, and Music Processing, July 2019.
10. Jian Kang, Wei-Qiang Zhang, Weiwei Liu, Jia Liu., Michael T. Johnson, Lattice Based Transcription Loss for End-to-End Speech Recognition, Journal of Signal Processing Systems, Volume 90 Issue 7, pp 1013-1023, July 2018
11. Jian Kang, Wei-Qiang Zhang, WeiWei Liu, Jia Liu, Michael T. Johnson, Advanced recurrent network-based hybrid acoustic models for low resource speech recognition, EURASIP Journal on Audio, Speech, and Music Processing, 2018.
12. Liang He, Xianhong Chen, Can Xu, Jia Liu, Michael T. Johnson, Local Pairwise Linear Discriminant Analysis for Speaker Verification, IEEE Signal Processing Letters Volume 25 No. 10, 1575-1579, 2018.
13. Jeffrey Berry, Andrew Kolb, James Schroeder, Michael T. Johnson, Jaw Rotation in Dysarthria Measured With a Single Electromagnetic Articulography Sensor, American Journal of Speech – Language Pathology, June 2017.
14. Ji, An; Johnson, Michael T; Berry, Jeffrey J., Parallel Reference Speaker Weighting for Kinematic-Independent Acoustic-to-Articulatory Inversion, IEEE/ACM Transactions on Audio, Speech, and Language Processing",24,10, October 2016, 1865-1875.

15. Xu-Kui Yang, Liang He, Dan Qu, Wei-Qiang Zhang, Michael T Johnson, Semi-supervised feature selection for audio classification based on constraint compensated Laplacian score, *EURASIP Journal on Audio, Speech, and Music Processing*, 1, 2016, 1-10.
16. Liu Wei-Wei, Cai Meng, Zhang Wei-Qiang Zhang, Liu Jia, Johnson Michael T., "Discriminative Boosting Algorithm for Diversified Front-End Phonotactic Language Recognition," *Journal of Signal Processing Systems*, February 2016.
17. Wei-Qiang Zhang, Cong Guo, Qiao Zhang, Jian Kang, Liang He, Jia Liu, and Michael T. Johnson, A Speech Enhancement Algorithm Based on Computational Auditory Scene Analysis. *Journal of Tianjin University*, in press, 2015. (Chinese journal. Original language citation: 张卫强, 郭聪, 张乔, 康健, 何亮, 刘加, and Michael T. Johnson, 一种基于计算听觉场景分析的语音增强算法, 天津大学学报.)
18. B.T.W. Bocher, K. Cherukurib, J.S. Makib, M. Johnson, D.H. Zitomer, Relating methanogen community structure and anaerobic digester function, *Water Research*, 70 (1), March 2015, 425-435.
19. PM Scheifele, MT Johnson, M Fry, B Hamel, K Laclede, Vocal classification of vocalizations of a pair of Asian Small-Clawed otters to determine stress, *The Journal of the Acoustical Society of America*, 138 (1), EL105-EL109, 2015.
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21. Arik Kershenbaum, Daniel Blumstein, Marie Roch, Michael T. Johnson, et. al., Acoustic sequences in non-human animals: a tutorial review and prospectus, *Biological Reviews*, 2014.
22. C Yu, KK Wójcicki, PC Loizou, JHL Hansen, MT Johnson, "Evaluation of the importance of time-frequency contributions to speech intelligibility in noise", *The Journal of the Acoustical Society of America*, vol. 135, no. 5, May 2014, 3007-3016.
23. Marek B. Trawicki, Michael T. Johnson, "Speech enhancement using Bayesian estimators of the perceptually-motivated short-time spectral amplitude (STSA) with Chi speech priors", *Speech Communication*, vol. 57, no. 2, February 2014, pp101-103.
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25. Junhong Zhao, Wei-Qiang Zhang, Hua Yuan, Michael T Johnson, Jia Liu, Shanhong Xia, "Exploiting contextual information for prosodic event detection using auto-context", *EURASIP Journal on Audio, Speech, and Music Processing*, vol. 2013, no. 1, December 2013 pp 1-14.
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27. An Ji, Michael T. Johnson, Edward J. Walsh, JoAnn McGee, Doug L. Armstrong, Discrimination of individual tigers (*Panthera tigris*) from long distance roars, *The Journal of the Acoustical Society of America*, vol. 133 no. 3, March 2013, pp1762-1769.
28. Yongzhe Shi, Weiqiang Zhang, Jia Liu, Michael T. Johnson, "RNN language model with word clustering and class-based output layer", *EURASIP Journal on Audio, Speech, and Music Processing* vol. 2013 no. 1, January 2013, pp1-7.

29. Peter M. Scheifele, Michael T. Johnson, David C. Byrne, John G. Clark, Ashley Vandlik, Laura W. Kretschmer, Kristine E. Sonstrom, "Noise impacts from professional dog grooming forced-air dryers", *Noise and Health*, vol. 14 no. 60, October 2012, p224-226.
30. Wen-Lin Zhang, Wei-Qiang Zhang, Bi-Cheng Li, Dan Qu and Michael T. Johnson, "Bayesian Speaker Adaptation Based on a New Hierarchical Probabilistic Model", *IEEE Transactions on Speech and Language Processing*, vol. 20 no. 7, July 2012, pp2002-2015.
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Peer-reviewed Conference Publications and Presentations

53. Mohammad Soleymanpour, Michael T. Johnson, Rahim Soleymanpour, Jeffrey Berry, Synthesizing Dysarthric Speech Using Multi-speaker TTS for Dysarthric Speech Recognition, International Conference on Acoustics, Speech, and Signal Processing (ICASSP), May 2022, Singapore.
54. Narjes Bozorg, Michael T Johnson, Mohammad Soleymanpour, Autoregressive Articulatory WaveNet Flow for Speaker-Independent Acoustic-to-Articulatory Inversion,

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55. Subash Khanal, Michael T Johnson, Mohammad Soleymanpour, Narjes Bozorg, Mispronunciation Detection and Diagnosis for Mandarin Accented English Speech, 2021 international Conference on Speech Technology and Human-Computer Dialog (SpeD), pp. 62-67, October 2021.
 56. Mohammad Soleymanpour, Michael T Johnson, Jeffrey Berry, Dysarthric Speech Augmentation Using Prosodic Transformation and Masking for Subword End-to-end ASR, 2021 international Conference on Speech Technology and Human-Computer Dialog (SpeD), pp. 42-46, October 2021.
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 60. Narjes Bozorg and Michael Johnson, Acoustic-to-Articulatory Inversion with Deep Autoregressive Articulatory-WaveNet, Interspeech 2020, October 2020, Shanghai China.
 61. Jiaqi Xie, Ruijie Yan, Shanyu Xiao, Liangrui Peng, Michael T. Johnson, Weiqiang Zhang, Dynamic Temporal Residual Learning for Speech Recognition, International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2020, May 2020, Barcelona, Spain.
 62. Jeffrey Berry and Michael T. Johnson, Perceived Acoustic Working Space Modulates Sensorimotor Learning, Motor Speech Conference, Santa Barbara, California, February 2020.
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 65. Narjes Bozorg, Michael T. Johnson, Jeffrey Berry, Comparing Articulography Consistency Between L1 and L2 Speakers, IEEE International Symposium on Signal Processing and Information Technology, Ajman, United Arab Emirates, December 2019.
 66. Narjes Bozorg, Michael T. Johnson, MLLR-PRSW for Kinematic-Independent Acoustic-to-Articulatory Inversion, IEEE International Symposium on Signal Processing and Information Technology, Ajman, United Arab Emirates, December 2019.
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84. Berry, Jeffrey, A Kolb, C North, and Michael T. Johnson, Acoustic and Kinematic Characteristics of Vowel Production through a Virtual Vocal Tract in Dysarthria, Interspeech, Singapore, September 2014.
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91. Jeffrey Berry, Christine Belchel, Cassandra North, and Michael T. Johnson, "Learning novel articulatory-acoustic mappings in dysarthria", American Speech Language Hearing Association 2013, Chicago, IL, November 2013.
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93. An Ji, Michael T. Johnson, Jeffrey Berry, "Articulatory space calibration in 3D Electromagnetic Articulography", China SIP, July 2013, Beijing China.
94. Jeffrey Berry, Cassandra North, Benjamin Meyers, Michael T. Johnson, Speech sensorimotor learning through a virtual vocal tract, Proceedings of Spring 2013 Meetings on Acoustics, Montreal, June 2013.
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University Mandarin accented English corpus, Proceedings of Spring 2013 Meetings on Acoustics, Montreal, June 2013.

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100. Jianglin Wang, An Ji, Michael T. Johnson, "Features for phoneme independent speaker identification", 2012 International Conference on Audio, Language and Image Processing (ICALIP), Shanghai, China, July 2012.
101. Michael T. Johnson and Patrick Clemins, "Individual Identification using Hidden Markov Models for Population Monitoring and Assessment", Invited Presentation to Special Session "Topical Meeting on Signal Processing of Subtle and Complex Acoustic Signals in Animal Communication", Acoustical Society of America Spring 2010 Meeting, Baltimore, Maryland, April 2010.
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111. Z. W. Slavens, R. S. Hinks, J. A. Polzin, and M. T. Johnson, "Improved MR Image Magnification by Generalized Interpolation of Complex Data," Proceedings of the 15th Annual Meeting of ISMRM, Berlin, Germany, (Abstract #1887), May 2007.
112. Craig A. Struble, Richard J. Povinelli, Michael T. Johnson, Dina Berchanskiy, Jidong Tao, Marek Trawicki, "Combined Conditional Random Fields and n-Gram Language Models for Gene Mention Recognition, Proceedings of the Second BioCreative Challenge Evaluation Workshop, Madrid, Spain, April 2007.
113. Xi Li, Jidong Tao, Michael T. Johnson, Joseph Soltis, Anne Savage, Kirsten M. Leong,, John D. Newman, Stress and Emotion Classification using Jitter and Shimmer Features, International Conference on Acoustics Speech and Signal Processing 2007 (ICASSP07), Honolulu, Hawaii, April 2007, pp 1081-1084.
114. Mohamed A. Mneimneh, Richard J. Povinelli, Michael T. Johnson " An Integrative Approach for the Measurement of QT interval" Computers in Cardiology, Valencia, Spain, September 2006, pp 329-332.
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116. Richard J. Povinelli, Mohamed A. Mneimneh, Michael T. Johnson "Cardiac Model Based Approach to QT Estimation" Computers in Cardiology, Valencia, Spain, September 2006, pp. 333-336.
117. Patrick Clemens, Marek B. Trawicki, Kuntoro Adi, Jidong Tau, and Michael T. Johnson, "Generalized Perceptual Features for Vocalization Analysis across Multiple Species", International Conference on Acoustics, Speech and Signal Processing (ICASSP) 2006, Toulouse, France, May 2006.
118. Kuntoro Adi, Michael T. Johnson, Cepstral moment normalization for robust individual identification of ortolan bunting (*emberiza hortulana L*), presented at the Spring 2006 meetings of the Acoustical Society of America, May 2006, Providence, RI
119. Anthony D. Ricke, Richard J. Povinelli, Michael T. Johnson. "Segmenting Heart Sound Signals," Computers in Cardiology, Leon, France, September 2005.
120. Marek Trawicki and Michael T. Johnson, "Automatic Song-Type Classification and Speaker Identification of Norwegian Ortolan Bunting (*Emberiza Hortulana*) Vocalizations", IEEE International Conference on Machine Learning in Signal Processing (MLSP), Mystic, Connecticut, September 2005.

121. Patrick J. Clemens and Michael T. Johnson, "Unsupervised Classification of Beluga Whale Vocalizations", Spring 2005 Meetings of the Acoustical Society of America, Vancouver, May 2005.
122. Kevin M. Indrebo, Richard J. Povinelli, Michael T. Johnson. "Third-Order Moments of Filtered Speech Signals For Robust Speech Recognition," International Conference on Non-Linear Speech Processing (NOLISP) 2005, Barcelona, Spain, 151-157.
123. Jidong Tao and Michael T. Johnson, "Comparison and Evaluation of Animal Vocalization Enhancement Techniques", Fall 2004 Meetings of the Acoustical Society of America, San Diego, November 2004.
124. Kuntoro Adi and Michael T. Johnson, "Automatic Song-type classification and individual identification of Ortolan Bunting (*Emberiza Hortulana* L) vocalizations", Fall 2004 Meetings of the Acoustical Society of America, San Diego, November 2004.
125. Heather E. Ewalt and Michael T. Johnson, "Combining Multi-source Wiener Filtering with Parallel Beamformers to Reduce Noise from Interfering Talkers", International Conference on Signal Processing (ICSP) 2004, Beijing, August 2004.
126. Kevin M. Indrebo, Richard J. Povinelli, and Michael T. Johnson, "A comparison of reconstructed phase spaces and cepstral coefficients for multi-band phoneme classification", ICSP 2004, Beijing, August 2004.
127. Patrick Clemens and Michael T. Johnson, "Generalized perceptual features for animal vocalization classification", Spring 2004 Meetings of the Acoustical Society of America, New York, May 2004.
128. Andrew Lindgren, Michael T. Johnson, and Richard J. Povinelli, Joint Frequency Domain and Reconstructed Phase Space Features for Speech Recognition", International Conference on Acoustics, Speech and Signal Processing (ICASSP) 2004, Montreal, May 2004.
129. Mike Zimmerman, Richard J. Povinelli, Michael T. Johnson, and Kris Ropella, "A Reconstructed Phase Space Approach for Distinguishing Ischemic from Non-Ischemic ST Changes using Holter ECG Data", Computers in Cardiology, Thessalonica, Greece, September, 2003.
130. Patrick Clemens and Michael T. Johnson, "Automatic Classification of African Elephant (*Loxodonta africana*) Follicular and Luteal Rumbles", 1st International Conference on Acoustic Communication by Animals, Baltimore, Maryland, July 2003, pp 81-82.
131. Patrick Clemens and Michael T. Johnson, "Automatic Type Classification and Speaker Identification of African Elephant (*Loxodonta Africana*) Vocalizations", Spring 2003 Meetings of the Acoustical Society of America, Nashville, May 2003.
132. Heather E. Ewalt and Michael T. Johnson, "Multiple Speech Signal Enhancement using a Microphone Array", Spring 2003 Meetings of the Acoustical Society of America, Nashville, May 2003.
133. Kevin M. Indrebo, Richard J. Povinelli, and Michael T. Johnson, "A Combined Sub-band and Reconstructed Phase Space Approach to Phoneme Classification", Non-Linear Speech Processing (NOLISP) 2003, Le Croisic, France, May 2003.

134. Jinjin Ye, Michael T. Johnson, and Richard J. Povinelli, "Study of attractor variation in the reconstructed phase space of speech signals, NOLISP 2003, Le Croisic, France, May 2003.
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137. Andrew C. Lindgren, Michael T. Johnson, and Richard J. Povinelli, "Speech Recognition using Reconstructed Phase Space Features", Proceedings of the International Conference on Acoustics, Speech and Signal Processing (ICASSP) 2003, Hong Kong, April 2003, pp I-60 – I-63.
138. Patrick Clemins and Michael T. Johnson, "Application of Speech Recognition to African Elephant (*Loxodonta Africana*) Vocalizations", Proceedings of ICASSP 2003, Hong Kong, April 2003, pp I-484 – I-487.
139. Michael T. Johnson, Andrew C. Lindgren, Richard J. Povinelli, and Xiaolong Yuan, "Performance of Nonlinear Speech Enhancement using Phase Space Reconstruction", Proceedings of ICASSP 2003, Hong Kong, April 2003, pp I-872 – I-875.
140. Xiaolin Liu, Richard Povinelli, and Michael T. Johnson, "Detecting Determinism in Speech Phonemes", IEEE Digital Signal Processing Workshop 2002, October 2002.
141. Jinjin Ye, Richard J. Povinelli, and Michael T. Johnson, "Phoneme Classification Using naïve Bayes Classifier in Reconstructed Phase Space", IEEE Digital Signal Processing Workshop 2002, October 2002.
142. Richard J. Povinelli, Michael T. Johnson, Nabeel A.O. Demerdash, and John F. Bangura, "A Comparison of Phase Space Reconstruction and Spectral Coherence Approaches for Diagnostics of Bar and End-Ring Connector Breakage and Eccentricity Faults in Polyphase Induction Motors using Motor Design Particulars", IEEE Industry Applications Society meetings 2002, October 2002.
143. Richard J. Povinelli, Felice M. Roberts, Kristina M. Ropella, and Michael T. Johnson, "Are Nonlinear Ventricular Arrhythmia Characteristics Lost, As Signal Duration Decreases?", Computers in Cardiology 2002, September 2002.
144. Patrick Clemins and Michael T. Johnson, "Automatic Speech Recognition and Speaker Identification of Animal Vocalizations", Measuring Behavior 2002, August 2002, pp 41-46.
145. Patrick Clemins, Heather Ewalt and Michael Johnson, "Time-aligned SVD Analysis for Speaker Identification", International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2002, May 2002, p 4160.
146. Michael T. Johnson and Leah H. Jamieson, "Temporal Features for Broadcast News Segmentation", ISCA Workshop on Prosody in Speech Understanding and Recognition, October 2001.
147. Michael T. Johnson and Mary P. Harper, "Near Minimal Weighted Word Graphs for Post-processing Speech", International Workshop on Automatic speech Recognition and Understanding (ASRU), December 1999.

148. M.P. Harper, M.T. Johnson, L.H. Jamieson, S.A. Hockema, and C.M. White, "Interfacing a CDG Parser with an HMM Word Recognizer Using Word Graphs", Proceedings of the International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 1999, March 1999, pp 733-736.
149. Michael T. Johnson, Leah H. Jamieson, and Mary P. Harper "Interfacing Acoustic Models with Natural Language Processing Systems", Proceedings of the International Conference on Spoken Language Processing (ICSLP) 1998, December 1998, pp 2419-2422.
150. A.M. Supranant, S.L. Hura, M.P. Harper, L.H. Jamieson, G. Long, S.M. Thede, A. Rout, T.H. Hsueh, S.A. Hockema, M.T. Johnson, J.B. Laflen, P. Srinivasa, and C.M. White, "Familiarity and Pronouncibility of Nouns and Names: The Purdue Proper Name Database", Spring 1998 Meetings of the Acoustical Society of America, June 1998.
151. Michael Johnson and Mita Desai, "An Adaptive Approach for Texture Modelling", Proceedings of the International Conference on Image Processing (ICIP) 1994, November 1994.
152. Harold Longbotham, Michael Johnson, Jack Harris, and Redouan Rouzky, "The FatBear: a nonarithmetic pico filter", Proceedings of the SPIE, Image Algebra and Morphological Image Processing IV, July 1993, pp 128-139.

Selected Invited Presentations

- Keynote Speaker, Weld Penetration Identification Based on Convolutional Neural Network, The 2019 International Workshop on Intelligentized Welding Manufacturing, University of Kentucky, Lexington, Kentucky, November 2019.
- Invited Colloquium Speaker, Acoustic to Articulatory Inversion for Computer-Aided Pronunciation Training, University of Texas at San Antonio Department of Electrical and Computer Engineering Colloquium Series, April 2019.
- Keynote Speaker, IEEE Joint Louisville-Lexington Chapters Holiday Dinner, Louisville, Kentucky, December 2018.
- Invited speaker, Shanghai Jiaotong University Colloquium Series, "Electromagnetic Articulography for Speaker Independent Acoustic-to-Articulatory Inversion, April 2015.
- Invited speaker, University of Memphis Institute of Intelligent Systems Colloquium Series, "Applying Speech Technology to Animal Vocalizations", April 2014.
- Invited speaker, NIMBioS Investigative Workshop on Analyzing Animal Vocal Sequences, "Using HMMs to model vocal structure and sequence for classification and identification", October 2013.
- Featured Keynote Speaker, GE Healthcare Technology symposium, "Innovations and Applications of Artificial Intelligence and Bayesian Modeling", August 2011.
- "Individual Identification using Hidden Markov Models for Population Monitoring and Assessment", Invited Presentation to Special Session "Topical Meeting on Signal Processing of Subtle and Complex Acoustic Signals in Animal Communication", at Acoustical Society of America Spring 2010 Meeting, Baltimore, Maryland, April 2010.
- Featured speaker at ShARE World Seminar 2008, "Innovation in Computation", Tsinghua University, Beijing China, December 2008.

- Keynote speaker for annual Tau Beta Pi banquet, “The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations,” November 2007.
- Presentation to Beijing Institute of Technology annual colloquia, “The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations,” Beijing, China, May 2007.
- Presentation to Tsinghua University Center for Speech Technology, “The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations,” Beijing, China, May 2007.
- Presentation to the Chinese Academy of Sciences, “The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations,” Institute of Acoustics, Beijing, China, May 2007.
- Michael T. Johnson, “The Dr. Dolittle Project: Applying speech technology to animal vocalizations”, Sigma Xi annual banquet, April 27, 2007.
- Michael T. Johnson, “Speech Research at Marquette – The Dr. Dolittle Project”, Marquette Trustees Committee on Academic Affairs and Planning, March 3, 2004.
- Michael T. Johnson, “Speech Recognition and Chaos Theory: A new approach to signal analysis”, Lynde Bradley Science Club, Rockwell Automation, November 11, 2003.
- Michael T. Johnson, “The Dr. Dolittle Project: Applying human speech processing algorithms to other species”, UW Milwaukee Computer Science Department Colloquium, November 7, 2003.
- Michael T. Johnson, “Talking with the Animals: Automatic Classification and Identification of Animal Vocalizations using Speech Technology”, Keynote Speaker for Physics Club of Milwaukee Annual Banquet, June 5, 2002.
- Michael T. Johnson, “Speech Processing Research at Marquette (Elephants and Cocktail Parties)”, COE National Research Council meeting, April 26, 2002.
- Michael T. Johnson, “Speech and Signal Processing Research at Marquette”, Invited speaker at Sigma Xi annual meeting, February 20, 2002.

PhD Dissertation and MS Thesis

- Michael T. Johnson, "Incorporating Prosodic Information and Language Structure into Speech Recognition Systems", PhD Dissertation, Purdue University School of Electrical and Computer Engineering, August 2000.
- Michael T. Johnson, "An Adaptive Texture Model Based on Markov Random Fields", Masters Thesis, University of Texas at San Antonio, August 1994.

Student PhD Dissertations (primary advisor)

- Soleymanpour, Mohammad, “Synthesizing Dysarthric Speech Using Multi-speaker TTS for Dysarthric Speech Recognition”, PhD Dissertation, University of Kentucky, August 2022.
- Narjes Bozorg, “Articulatory Wavenet: Deep Autoregressive Model for Acoustic-to-Articulatory Inversion”, PhD Dissertation, University of Kentucky, December 2020.

- An Ji, “Speaker Independent Acoustic-to-Articulatory Inversion”, PhD Dissertation, Marquette University, December 2014.
- Jianglin Wang, "Physiologically-motivated feature extraction methods for speaker recognition", PhD Dissertation, Marquette University, December 2013.
- Jidong Tao, “Acoustic Model Adaptation for Automatic Speech Recognition and Animal Vocalization Classification”, PhD Dissertation, Marquette University, May 2009.
- Marek Trawicki, “Distributed Multichannel Processing for Signal Enhancement”, PhD Dissertation, Marquette University, May 2009.
- Kuntoro Adi, “Hidden Markov model based animal acoustic censusing: learning from speech processing technology”, Ph.D. Dissertation, Marquette University, May 2008.
- Patrick J. Clemens, “Automatic Classification of Animal Vocalizations”, PhD Thesis, Marquette University, May 2005.

Student MS Theses (primary advisor)

- Subash Khanal, “Mispronunciation Detection and Diagnosis in Mandarin Accented English speech, Masters Thesis, University of Kentucky, May 2020.
- Jose, Neenu, “Speaker and Gender Identification using Bioacoustic Data Sets”, Masters Thesis, University of Kentucky, May 2018.
- Vonderhaar, Joe, “Speaker-specific Adaptation of Maeda Synthesis Parameters for Auditory Feedback”, Masters Thesis, Marquette University, May 2017.
- Jones, Deriq, “Development of Kinematic Templates for Automatic Pronunciation Assessment using Acoustic-to-Articulatory Inversion”, Masters Thesis, Marquette University, May 2017.
- Kelly Vonderhaar, Analysis of Interference between Electromagnetic Articulography and Electroglottograph Systems, Masters Thesis, Marquette University, August 2016.
- Xiangyu Zhou, Least-squares Mapping from Kinematic data to Acoustic Synthesis Parameters for Rehabilitative Acoustic Learning, Masters Thesis, Marquette University, May 2016.
- Andrew Kolb, Development of Software Tools and Analysis methods for the Use of Electromagnetic Articulography Data in Speech Research”, Masters Thesis, Marquette University, May 2015.
- Zac Slavens, “Generalized interpolation applied to MR image magnification and gradient nonlinearity correction”, Masters Thesis, Marquette University, May 2008.
- Xi Li, “SPEech Feature Toolbox (SPEFT) Design, and Emotional Speech Feature Extraction”, M.S. Thesis, Marquette University, August 2007.
- Anthony D. Ricke, “Automatic Frame Length, Frame Overlap, and Hidden Markov Model Topology for Automatic Speech Recognition of Animal Vocalizations”, M.S. Thesis, Marquette University, December 2006.
- Jinjin Ye, “Speech Recognition Using Time Domain Features from Phase Space Reconstructions”, M.S. Thesis, Marquette University, May 2004.
- Franck Hounkpevi, “Triphone Creation Through Rule-based Trajectory Interpolation for Continuous Speech Recognition”, M.S. Thesis, Marquette University, May 2003.

- Xiaolong Yuan, “Auditory Model-Based Bionic Wavelet Transform for Speech Enhancement”, M.S. Thesis, Marquette University, May 2003.
- Andrew Lindgren, “Speech Recognition using Features extracted from Phase Space Reconstructions”, M.S. Thesis, Marquette University, May 2003.
- Heather Ewalt, “Speech Signal Enhancement Using a Microphone Array”, M.S. Thesis, Marquette University, December 2002.

Selected News and Magazine Articles

- “Listening to the Animal Kingdom”, Discover Magazine, Marquette University, 2008, pp 14-15.
- “Decoding Calls of the Wild”, by Mark Johnson, feature article in Milwaukee Journal Sentinel, part of “Brainpower: Groundbreaking Thinkers in Wisconsin”, October 2008.
- “Say What?”, cover article on Dr. Dolittle Project, Weekly Reader Science, Spring 2007.
- “Animal Talk”, by Elena Cabral, Scholastic News Edition 5/6, December 18, 2006.
- “Animal “Speech” Project Aims to Decode Critter Communication”, by Maryann Mott, National Geographic Online, <http://news.nationalgeographic.com/news/2006/09/060926-dolittle-project.html>, September 2006.
- “Parsing the Puffin’s Patois”, article on Dr. Dolittle Project by Rachel Metz, Wired News online, July 2006.
- “Quacking the Code”, by Katie Pelech, article in Milwaukee Magazine, March 2006, p. 30.
- “What Jumbo tells Dumbo” by Alex Antunes, feature article in Computers in Science and Engineering (CISE) magazine, published by IEEE Computer Society, September-October 2005, pp 3-6.
- “Discoveries with Disney”, article about collaborative project with Disney’s Animal Kingdom published in MU Magazine, Fall 2003, p. 17.

GRANTS

Major Grants Funded

1. Michael T. Johnson (PI), Aaron Cramer (co-PI), Dan Ionel (co-PI), and Janet Lumpp (co-PI), GAANN Fellowship Program in Electrical and Computer Engineering at the University of Kentucky, Department of Education, 10/1/2019 – 9/30/2022. Total Budget \$600,702 over 3 years. Supplement of \$51,000 in 2020 for 1 additional fellow.
2. Jeffrey Berry (PI) and Michael T. Johnson (co-PI), “Speech Sensorimotor Control in Non-progressive Dysarthria”, National Institutes of Health R-15, 7/1/2018 – 6/30/21. Total Budget \$446,956 over 3 years (\$166,428 to University of Kentucky as subcontract to Marquette University).
3. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), “RI: Small: Speaker Independent Acoustic-Articulator Inversion for Pronunciation Assessment”, National Science Foundation CISE Directorate, 2013. Total Budget \$449,643 over 3 years.
4. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, National Science Foundation CISE Directorate, 2011. Total Budget: \$149,896.00 over 1 year.
5. Michael T. Johnson (PI), Elizabeth Mugganthalder (Fauna Communications Research Institute), Peter Scheifele (National Undersea Research Center, University of Connecticut), Mike Darre (Animal Sciences, University of Connecticut), Anne Savage (Disney Animal Kingdom), “The Dr. Dolittle Project: A Framework for Classification and Understanding of Animal Vocalizations”, NSF IT-R program under CISE Directorate, 2003. Total Budget: \$1,200,000 over 4 years.
6. Michael T. Johnson (PI) and Richard J. Povinelli, “Integration of Stochastic and Dynamical Methods for Speech Technology”, NSF IT-R program under CISE directorate, 2001. Total Budget: \$360,000 over 3 years.
7. Richard J. Povinelli (PI), Nabeel A. O. Demerdash, Edwin Yaz, and Michael T. Johnson, “A Novel Approach to Fault Modeling, Diagnostics, and Prediction in Motor Drive Systems”, NSF Division of Controls, Networks, and Computational Intelligence, 2003. Total Budget: \$340,000 over 3 years.

Additional Grants Funded

1. Peter Scheifele (PI), Leesa Scheifele (co-PI), and Michael T. Johnson (co-PI), Measuring Hearing Acuity and Vocalization Signals of the Asian Small-Clawed Otter (*Amblonyx cinereus*) in Response to Anthropogenic Noise, Indianapolis Zoo, March 2017, Funded \$14,300.
2. Kevin McGowan (PI) and Michael T. Johnson (co-PI), Apparatus and Procedure for Ultrasound Investigation of Velopharyngeal port aperture, University of Kentucky Igniting Research Collaborations project, Funded \$27,726 in 2018-2019.
3. Jeffrey Berry (PI) and Michael T. Johnson (co-PI), “Speech Rehabilitation Using a Virtual Vocal Tract”, Marquette Regular Research Grant (RRG). Funded \$6,000 in 2014-15.
4. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), Research Experiences for Undergraduates (REU) supplement to “RI: Small: Speaker Independent Acoustic-Articulator Inversion for Pronunciation Assessment”, NSF REU program, CISE directorate. Funded \$6,000 in 2013-2014, \$6000 in 2014-2015, \$6000 in 2015-2016.
5. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), OISE International supplement to “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, co-funded by CISE and OISE directorates. Funded \$19,840 in 2012-2013.

6. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), Research Experiences for Undergraduates (REU) supplement to “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, NSF REU program, CISE directorate. Funded \$8,000 in 2011-2012.
7. Richard J. Povinelli (PI) and Michael T. Johnson (PI), “An examination of phoneme confusability in spoken English”, Marquette Regular Research Grant (RRG). Funded \$15,000 in 2011-2012.
8. Michael T. Johnson (PI) et. al., Research Experiences for Undergraduates (REU) supplement to “The Dr. Dolittle Project”, NSF REU program, CISE directorate. Funded at \$6,000 in 2003-04, \$15,000 in 2004-2005, \$12,000 in 2005-2006, \$15,000 in 2006-2007 Total Budget: \$48,000.
9. Michael T. Johnson (PI) and Richard J. Povinelli, REU supplement to “Integration of Stochastic and Dynamical Methods for Speech Technology”, NSF REU program, CISE directorate. Funded at \$15,000 in 2001-02, 2002-03, and 2003-04, Total Budget: \$45,000.
10. Richard J. Povinelli (PI), Nabeel A. O. Demerdash, Edwin Yaz, and Michael T. Johnson, REU supplement to “A Novel Approach to Fault Modeling, Diagnostics, and Prediction in Motor Drive Systems”, NSF REU Program, CNCI Division. Funded at \$6,000 in 2003-04 and 2004-05. Total Budget to-date: \$6,000.

PROFESSIONAL ACTIVITIES

Technical Committees and Publication editing

- Area Editor, Speech Enhancement, Speech Communication Journal, 2014-present
- Member, IEEE Signal Processing Society Speech and Language Technical Committee (SLTC), 2014-2017

Selected Publication Review Participation

- Speech Communication Journal
- International Conference on Acoustics, Speech and Signal Processing
- Interspeech IEEE Automatic Speech and Recognition Workshop
- Workshop on Spoken Language Technology
- IEEE Transactions on Speech and Audio Processing
- Journal of the Acoustical Society of America
- IEEE Sensors Journal
- IEEE Transactions on VLSI
- IEEE Transactions on Signal Processing
- IEEE Signal Processing Letters
- IEEE Transactions on Neural Systems
- Journal of Applications of Signal Processing
- Animal Behavior Journal
- EURASIP Journal on Audio, Speech and Music Processing
- International Journal on Adaptive Control and Signal Processing
- Biosystems Engineering Journal
- Acoustics Research Letters Online
- International Conference on Decision and Control
- Midwest Symposium on Circuits and Systems
- Iasted Circuits and Systems Conference
- Workshop on Signal Processing and Applications
- American Controls Conference
- Journal of Zhejiang University Science C (Computers & Electronics)
- Journal of Medical Engineering & Physics
- Open Signal Processing Journal
- IEEE Spoken Language Technology workshop
- International Conference on Signal Processing (ICSP)
- Journal of Computers
- International Journal of Electronics and Communications
- Franklin Institute journal

Selected Funding Review Participation

- National Science Foundation
Have reviewed for CCLI, CAREER, CONICYT, and ITR programs, as well as various regular programs in the CISE and Biological Sciences directorate
- International: Austrian Science Fund, Icelandic Research Fund
- Medical College of Wisconsin Clinical and Translational Science Institute (CTSI)

Professional and honorary society memberships

- ECE Department Heads Association (ECEDHA), 2016-present
ECEDHA Awards Committee, 2018-present
- Southeast ECEDHA (SECEDHA), 2016-present
Secretary 2019-2020, Vice President 2020-2022, President 2022-2023
- Southeastern Center for EE Education (SCEEE) Board of Directors, 2020-2023
- Senior Member IEEE, membership 1994-present
- IEEE Signal Processing Society, 1998-present
- Acoustical Society of America (ASA), 2001-present
- Eta Kappa Nu, 1996-present
- Upsilon Pi Epsilon, 2001-present
- Senior Member Sigma Xi, 2001-2012
- Association of Computer Machinery (ACM), 1998-2010
- IEEE Computer Society, 1998-2010
- IEEE Circuits and Systems Society, 2001-2009
- Association of Computational Linguistics (ACL), 1998-2002
- International Speech Communication Association (ISCA), 2001-2006

Other professional activities and awards

- Selected for SEC Academic Leadership Development Program, 2019-2020
- Featured in “Groundbreaking Thinkers in Wisconsin” article series 2007
- Researcher of the Year, Marquette College of Engineering, 2006-2007
- Young Engineer of the Year, Engineers and Scientists of Milwaukee (ESM), 2006
- Registered Professional Engineer, State of Wisconsin

TEACHING AND EDUCATIONAL ACTIVITIES

Courses Taught

Undergraduate

EECE Freshman Seminar
Computer Hardware
Digital Logic Design
Linear Systems Analysis
Embedded Systems Design
Circuits 1
Signals and Systems Lab
Introduction to Engineering
Understanding Leadership

Graduate and Joint Undergraduate/Graduate

Digital Signal Processing
Speech Processing
Optimal/Adaptive Signal Processing
Pattern Recognition
Analysis of Algorithms
Information Theory
Professional Research Writing

Teaching awards

Marquette Eta Kappa Nu (HKN) honor society EECE Teacher of the Year, 2014, 2005

Selected Teaching, Education, and Pedagogy activities

UK Global Engagement Academy, Spring 2022
Center for Enhancement of Learning and Teaching (CELT) workshop, Spring 2021
UK Counseling Center, ECE faculty meeting guest on Student Mental Health, Fall 2021
UK Unconscious Bias Seminar, Fall 2019
UK Experienced Leadership Academy, Fall 2018
UK CoE Teaching Community of Practice, Fall 2017
UK Chairs Academy, Fall 2016
Chair, CoE Teaching Best Practices Committee, Spring 2017
Coordinator for Marquette EECE Freshman Seminar, 2011-2015
Participant in KEEN/Lafferty Teaching Development Workshop, Fall 2015, Spring 2016
Mentored numerous Research Experience for Undergraduates (REU) through NSF grants
International Teaching Experience, Tsinghua University, 2008-2009, 2011, 2013, 2014-2015
Signals and Systems Concept Inventory site coordinator for NSF study, 2003-2006
Developed and gave numerous DSP/Audio Workshops for middle and high school outreach
Participated with IEEE Signal Processing Education Technical Committee activities
Developed introductory DSP module for General Engineering freshman course
Participated in MU Assessment seminar by Barbara Wolvoord, May 2005
Participated in NSF Engineering Education Scholars (EES) program for new faculty
Participated in Prentice Hall Symposium on Education, Chicago, IL, 2001
Participated in NSF/ASEE Visiting Scholars Teaching Workshop, Fall 2000 and Spring 2001
Member, Advisory committee MU Center on Teaching and Learning (2005-2010)

Student Mentoring Activities (2000-present)

MS and PhD Committees
Undergraduate Research Projects
Senior Design Projects
Individual Student Mentoring

ACADEMIC SERVICE

Academic Service at the University of Kentucky

University:

- Steering Committee for Wildcat Foundations first-year experience program (2017-2019)
- Represented UK in SEC Academic Leadership Development Program (ALDP), 2019-2020

College:

- Chair, Scholarship Processes Committee, Engineering (2023-2024)
- Member, Funkhouser Renovation Planning Committee (2023-2024)
- Advocate, Advocates and Allies Working Group (2023-2024)
- Chair, BME Department Chair Search Committee (2022-2023)
- Chair, Engineering Technology Faculty Committee (2020 – present)
ET Chair search, LST Faculty search (2020-2021), CPT Faculty search (2021-2022)
- CoE Funkhouser (2019, 2021, 2023) and Grehan (2018-19) Building Committees
- CoE Recruitment Advisory Committee (2019-present)
- CoE Search Committee for Senior Director of Philanthropy (2018)
- Director, CoE Scholars in Engineering Leadership Program (2017-present)
- CoE Search Committee for Director of Marketing and Communications (2017)
- Chair, CoE Teaching Best Practices committee (Spring 2017)
- CoE Chair's Committee (2016-present, Chair 2018-2019)

Departmental:

- ECE Department Chair (2016-2023)

Academic Service at Marquette University

University:

- University Board of Graduate Studies (2009-2016; chair, 2012-2016)
- University Search Committee, Vice Provost for Enrollment Management (2015-2016)
- University Steering committee on academic integrity (2012-2014)
- University Subcommittee on academic integrity (2011)
- COF, Subcommittee on Nominations and Elections (chair, 2004-2007).
- Faculty mentor for Dissertation Boot Camps (2008, 2010)
- Advisory Committee for Center on Teaching and Learning (2005-2010)

College and Departmental:

- COE Research Activity Committee (2011)
- COE Dean of Engineering Search Committees (2003, 2009)
- COE Freshman Programs Committee (2006-2007)
- COE Discovery Towers Building Committee (2004-2005)
- BME Biocomputer Program Advisory Board (2001-2016)
- EECE Graduate Committee (2000-2003, 2005-2016)
- EECE Undergraduate Committee (2003-2008, 2012-2015)
- EECE Director of Graduate Studies (2009-2012)
- EECE Department Grad Program Assessment Coordinator (2009-2012)
- Numerous EECE faculty search and other subcommittees

Selected public service and outreach activities

- UK CoE E-day, every spring annually since 2016
- UK CoE and ECE prospective student tours, several times per month since 2016
- Guest speaker, Triangle UK Leadership Academy students, July 2023
- Keynote Speaker, International Workshop on Intelligentized Welding Manufacturing, November 2019
- Keynote Speaker, IEEE Joint Lexington/Louisville Section Meeting, December 2018
- UK ECE Middle School STEM program, November 2018
- UK ECE STEM Outreach camp for girls, June 2016, June 2017
- Guest speaker, Marquette Girls Who Code outreach program, March 2016.
- Coordinator for Marquette EECE open house activities, Fall 2015.
- Faculty advisor for Marquette's HKN chapter and Sigma Phi Delta engineering fraternity
- Guest speaker, MOTIF English corner, Beijing, China. "Diversity", September 2014, "Humor", March 2015, "Languages of the World", June 2015.
- Senior Visiting Scholar, Tsinghua University, China
2014-15, 2008-09, Summer 2011, Summer 2013
- COE iPalm2 outreach program to high school students, August 2011, 2012, 2013
- Represented Marquette COE on visits to Harbin Institute of Technology, China, 2011
- Panelist for Marquette University Advancement workshop, 2011
- Marquette ORSP Brown Bag panelist "One thing led to another" research series, 2010
- Spoke to students at Rosenwald Dunbar elementary school in Nicholasville, KY, about the Dolittle project and careers in engineering, 2008
- Spoke to students at Cardinal Valley elementary school in Lexington, KY, about the Dolittle project and careers in engineering, 2006
- Participated in Marquette University High School Junior Career Day, 2005
- Spoke to "Future Problem Solvers" gifted program at Templeton Middle School about Artificial Intelligence, 2005
- COE representative for "We are Marquette" World Café Conversation, 2004
- EECE representative and presenter for GEMS Advanced Education Fair, 2004
- Spoke at MSOE about graduate school opportunities in EECE, 2002-2003
- Presented at Lynde Bradley Science Club, Rockwell Automation, 2003
- Presented at UW Milwaukee Computer Science Department Colloquium, 2003
- Marquette ORSP Brown Bag panelist on Pre-proposal Contact with Sponsors, 2003
- Keynote on Global Research, Marquette COE National Advisory Council, 2002
- Keynote Speaker at Physics Club of Milwaukee annual banquet, 2002
- Spoke at UWM Elementary Education class on careers in science and technology, 2002
- Participated in Nathan Hale H.S. Career Day regarding careers in engineering, 2002
- Active participant in Marquette EECE and COE open houses, 2000-2015
- Blackjack dealer for Marquette engineering student council casino nights, 2000-15
- Regular participant in Marquette parent lunch program, 2000-2015
- Member, Eastbrook Church & worship choir, 2001-2016
- Numerous student and student group activities, both UK and Marquette, 2000-present
- Numerous research talks, colloquia, and seminars, 2000-present
- Numerous other volunteer and community activities, 2000-present

LITIGATION EXPERIENCE

US District Court

Snap-On Technologies, Inc. vs. Hunter Engr. Company, 98-C-369, 98-C-837, 00-C-929, 00-C-1511 (2002)

Jurisdiction: USDC Eastern District of Texas
Client: Hunger Engineering Company (Defendant)
Law Firm: Kirkland and Ellis LLP
Activities: Declaration, Deposition

EVS Codec Tech LLC and Saint Lawrence Comm. LLC vs. ZTE and ZTE USA Inc., 3:19-cv-00385 (2019)

Jurisdiction: USDC Northern District of Texas
Client: ZTE Inc. (Defendant)
Law Firm: K&L Gates LLP
Activities: Source Code Review, Technical consulting services

Nuance Communications, Inc. vs. MModal LLC., 1:17-cv-01484-MN (2019)

Jurisdiction: USDC District of Delaware
Client: MModal LLC (Defendant)
Law Firm: Duane Morris, LLP and Latham & Watkins, LLP
Activities: Reports (3), Source Code Review, Deposition

VoiceAge EVS LLC vs. Apple Inc, 1:20-cv-01061-CFC (2020)

Jurisdiction: USDC District of Delaware
Client: Apple Inc. (Defendant)
Law Firm: WilmerHale LLP
Activities: Source Code Review, Technical consulting services

Freshhub vs. Amazon Inc., 1:19-cv-00885-ADA (2020)

Jurisdiction: USDC Western District of Texas
Client: Amazon (Defendant)
Law Firm: Fenwick and West LLP
Activities: Reports (3), Source Code Review, Depositions (2), Trial Testimony

Vocalife LLC vs. Sonos, Inc., 2:21-cv-00129 (2021)

Jurisdiction: USDC Eastern District of Texas
Client: Sonos (Defendant)
Law Firm: Lee, Sullivan, Shea, and Smith LLP
Activities: Declaration

VB Assets, LLC vs. Amazon.com, Inc., 1:19-cv-01410-MN (2021)

Jurisdiction: USDC District of Delaware
Type of Litigation: Patent Infringement
Client: Amazon (Defendant)
Law Firm: Fenwick and West LLP
Activities: Reports (3), Depositions (2), Trial Testimony

Dialect, LLC v. Amazon.com et al., No. 1:23-cv-581 (2024)

Jurisdiction: USDC Eastern District of Virginia
Type of Litigation: Patent Infringement
Client: Amazon (Defendant)
Law Firm: Fenwick and West LLP
Activities: Technical Consulting (currently active)

Patent Trial and Appeal Board (PTAB) Inter Partes Review (IPR)

LG Electronics vs. Saint Lawrence Communication, IPR2015-01874, IPR2015-01875 (2015)

Client: LG Electronics (Petitioner)

Law Firm: Mayer Brown LLP

Activities: Declarations (2)

ZTE Corp. vs. Saint Lawrence Communications, IPR2016-00704, IPR2016-00705 (2016)

Client: ZTE Inc. (Petitioner)

Law Firm: Finnegan, Henderson, Farabow, Garrett & Dunner LLP

Activities: Declarations (2), Deposition

Google LLC vs. Koninklijke Philips, IPR2017-00437 (2017)

Client: Koninklijke Philips (Patent Owner)

Law Firm: Fitzpatrick, Cella, Harper, & Scinto

Activities: Declaration, Deposition

Google LLP vs. Sonos, Inc., IPR2021-00475 (2022)

Client: Sonos (Patent Owner)

Law Firm: Ackerman LLP and Lee, Sullivan, Shea, and Smith LLP

Activities: Declaration

Sonos, Inc. vs. Google LLP, IPR2022-01592, 2022-01594, 2023-0118, 2023-0119 (2022)

Client: Sonos (Petitioner)

Law Firm: Lee, Sullivan, Shea, and Smith LLP

Activities: Declarations (4) (currently active, all 4 IPRs instituted)

Sonos, Inc. vs. Google LLP, IPR2023-01288, IPR2023-01284 (2023)

Client: Sonos (Petitioner)

Law Firm: Lee, Sullivan, Shea, and Smith LLP

Activities: IPR Declarations (2)

Jawbone Innovations, LLC vs. Meta Platforms, Inc., IPR2024-00349, IPR2024-00350, (2024)

Jurisdiction: USDC Western District of Texas

Client: Meta Platforms (Petitioner)

Law Firm: Allen & Overy LLP

Activities: IPR Declarations (2)

International Trade Commission (ITC)

Google LLP vs. Sonos, Inc. 337-TA-1329, 337-TA-1330 (2022)

Client: Sonos (Respondent)

Law Firm: Lee, Sullivan, Shea, and Smith LLP

Activities: Technical Consulting, 4 reports, 2 depositions, ITC hearing testimony

Other Jurisdictions

Google LLP vs. Sonos, Inc. Docket T-952-20 (2021)

Jurisdiction: Canadian Federal Court

Type of Litigation: Patent Infringement

Client: Sonos (Defendant)

Law Firm: ROBIC, LLP and Lee, Sullivan, Shea, and Smith LLP

Activities: Reports (2), Source Code Review, Trial Testimony

APPENDIX B

Description
U.S. Patent No. 8,150,044 (“’044”)
File History of U.S. Application No. 8,150,044 (“’044FH”)
U.S. Patent Application Publication No. 2001/0046304 (“Rast”)
European Patent Application No. 1,519,625 A2 (“Victorian”)
U.S. Patent Application Publication No. 2007/0038395 (“Green”)
U.S. Patent Application Publication No. 2009/0085873 (“Betts”)
U.S. Patent Application Publication No. 2006/0293882 (“Giesbrecht”)
U.S. Patent 4,325,142 (“Nakazawa”)
International Publication No. 2006/070229 A2 (“Jaakkola”)
Spoken language Processing (“SLP”), Huang, 2001, pages 500- 501
File History of Provisional Patent Application No. 60/883,013